Session 1aAAa

Architectural Acoustics: Towards a Benchmark in Computational Room Acoustics

Alexander C. Bockman, Cochair
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Chair's Introduction—7:30

Invited Papers

7:35

1aAAa1. Round robins in room acoustics. Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

In 1995, an intercomparison on room acoustics computer simulations was launched, the so-called round robin I. The object of interest was a mid-sized auditorium. That project also included a small measurement project, where sessions of impulse response measurements were carried out with several teams and instrumentation. Since that time, the variations in the results obtained by computer simulations and those obtained by measurements were studied more deeply, and this was the starting point for two more round robins focusing on aspects other room sizes such as a multi-purpose hall and a rather small recording studio. Also, investigations on the uncertainty in ISO 3382 measurements were initiated. In this short presentation the historic round robins I to III are re-visited and plans for a round robins IV focusing on auralization on room acoustics are discussed.

7:45

1aAAa2. Benchmarks in computational room acoustics. U. Peter Svensson (Dept. of Electron. and Telecomm., Norwegian Univ. of Sci. and Tech., NO-7491 Trondheim, Norway, svensson@iet.ntnu.no)

A benchmark project called "A Benchmarking Framework for Wave-Based Computational Methods" was run in Japan, 2003–2005, coordinated by Tetsuya Sakuma, University of Tokyo, and with a few European participants. The project defined a number of interior and exterior (scattering) cases, ranging from very simple shapes to more realistic cases. A web site was set up where results, timing data, and methods could be submitted in great detail. A recent benchmark initiative, by Dirk Schroder and Michael Vorländer from RWTH, Aachen, is to create an "open measurement" web site where measurement results, software, etc., can be gathered and openly available. Brief presentations of those benchmark efforts will be given.

7:55

1aAAa3. Reverberation modeling workshops. John S. Perkins (Naval Res. Lab., Washington, DC 20375, john.perkins@nrl.navy.mil) and Eric I. Thorsos (Univ. of Washington, Seattle, WA 98105-6698)

To evaluate progress made in basic and applied underwater acoustic reverberation modeling and to make recommendations for transitions to operational systems, a series of two reverberation modeling workshops (RMWs) was held (the last in May 2008). A basic goal of the RMWs was to provide well-defined problems and consensus solutions to support verification and validation for new models, upgrades to Navy Standard models, and geoacoustic inversion techniques based on reverberation data. The basic problem in designing the workshop was that even the simplest reverberation problems of interest to the Navy do not have closed form solutions and are still (essentially) beyond our computational capabilities to solve using standard "exact" numerical techniques. All current, practical underwater reverberation models replace the physical problem by employing scattering and loss functions or tables. We discuss the development of a sequence of well-defined problems (physics-based), with the equivalent loss/scattering input, which increases in complexity. We also discuss the lessons learned in this process and point out some of the unexpected results from the workshops, and make recommendations for future benchmarking workshops. [Work supported by the Office of Naval Research.]

8:05—8:35 Panel Discussion

Session 1aAAb

Architectural Acoustics and Underwater Acoustics: Computational Methods for Auralization in Air and Water I

Jason E. Summers, Cochair
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Michael Vorländer, Cochair
Inst. fur Technische Akustik, RWTH Aachen Univ., D-52056 Aachen, Germany

Chair's Introduction—8:50

Invited Papers

8:55

1aAAb1. Computational methods in architectural acoustics. Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

Sound field modeling and auralization have been used in architectural acoustics for many years. Today, the technique of auralization can also be implemented with real-time performance. Then it is part of the technology of virtual reality. Apparent simple scenarios of interaction, however, for instance, when a person is leaving a room and closes a door, require complex models of room acoustics and sound transmission. Otherwise the coloration, the loudness, and timbre of sound within and between rooms are not represented adequately. Still numerous approximations must be made to reach this goal. In the end, the resulting sound is not intended to be physically correct, but perceptively plausible. Knowledge about human sound perception is, therefore, a very important prerequisite to evaluate auralized sounds. In this paper the algorithms for sound field rendering and spatial reproduction are reviewed and discussed with regard to future research.

9:15

1aAAb2. Time series simulation of underwater sound: Key issues and techniques. Robert P. Goddard (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, robert_goddard@apl.washington.edu)

Computer systems for generating simulated underwater sound have been in use for more than 30 years. Such systems, including the author's sonar simulation toolset (SST), enable users to build an artificial ocean that sounds like a real ocean, as heard by a user-specified listening system. Such signals are useful for designing new acoustic systems for undersea sensing or communication, testing existing systems, predicting performance, developing tactics, training operators, planning experiments, and interpreting measurements. Signals of interest include reverberation, target echoes, discrete sound sources, and background noise. The author will describe, in broad outline, techniques for generating simulated underwater sound and criteria for choosing among them, focusing on target echoes and reverberation for active sonar systems. The characteristics of the ocean as an acoustic medium, the characteristics of the sensors, the processing and display systems behind the sensors, the purpose of the simulation, and the computational resources available for the simulation all strongly influence the choice of algorithms to achieve useful, cost-effective, simulated underwater sound. The characteristics of the human auditory system are of minor importance.

9:35

1aAAb3. Sound propagation at micro-scale in urban areas. Jian Kang (School of Architecture, Univ. of Sheffield, Sheffield S10 2TN, United Kingdom)

While large scale noise-mapping techniques have been applied extensively in practice, as required by the EU Directive on environmental noise, they are often not applicable for micro-scale urban areas, such as a street or a square. This talk will discuss a series of simulation techniques as well as related acoustic theories for accurately calculating the sound field for micro-scale urban areas. This includes energy-based image source methods for street canyons and urban squares with geometrically (specularly) reflecting boundaries, image source method considering interference, ray-tracing, radiosity model for diffusely reflecting boundaries, transport theory, equivalent source method, and some other models. Techniques for urban acoustic animation will also be briefly discussed.

9:55—10:05 Break

10:05

1aAAb4. Computational techniques for underwater acoustic time-series simulations. Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201)

In the underwater environment, electromagnetic signals are highly attenuated and therefore acoustic systems are primarily used for remote sensing, communications, and imaging. Underwater acoustic propagation modeling is used to understand performance of these systems and the subject is considered fairly mature. A variety of approaches have been developed to simulate acoustic time-series in underwater environments (i.e., the ocean). Usually the biggest challenge is simply knowing the propagation environment in enough detail for accurate simulations. Setting that aside, there are several other complicating factors that limit the quality of simulations even in known environments. Among these are the irregular boundaries at the sea-surface and seabed boundaries which scatter the acoustic field. Solutions to these scattering problems are often formulated in the frequency domain and time-series formed using Fourier synthesis. However, Doppler effects due to sea-surface motion as well as source and receiver motion add additional modeling challenges particularly for frequency domain approaches. In these cases, ray-based techniques have been used successfully to model acoustic time-series. In this presentation the various approaches to simulating underwater acoustic time-series will be described for static and dynamic environments. Applications of these simulations for predicting reverberation and multipath in the underwater environment will also be presented.

10:25

1aAAb5. The inclusion of diffraction effects in room acoustical modeling. U. Peter Svensson (Dept. of Electron. and Telecomm., Norwegian Univ. of Sci. and Tech., NO-7491 Trondheim, Norway, svensson@iet.ntnu.no)

This paper will give a brief overview of some alternatives for including diffraction effects in computational room acoustics. Geometrical acoustics is the basis for most computer modeling methods in room acoustics, and both deterministic and stochastic approaches are used. The image source method and the beam tracing method dominate for the former, and ray tracing for the latter. Deterministic methods are used for sequences of specular reflections, and diffraction effects are straightforward to include in such methods. Paths that involve at least one diffuse reflection are usually handled by ray tracing and recent work has suggested how to implement diffraction in ray tracing. The inclusion of diffraction effects might offer more accurate modeling, most notably for free-hanging reflectors, orchestra pits, balcony edges, etc. The boundary condition issue will be discussed since diffraction solutions that are useful in room acoustics exist only for ideally rigid surfaces. Underlying approximations in some formulations will be mentioned, and computational aspects will be described, including the huge number of diffraction paths that are generated in a room, with a corresponding huge range of amplitudes. Special attention will be given to singularity issues and an attempt at an outlook for the near future will be offered.

10:45

1aAAb6. Efficient acoustic radiance transfer method with time-dependent reflections. Samuel Siltanen, Tapio Lokki, and Lauri Savioja (Dept. of Media Technol., Aalto Univ. School of Sci. and Technol., P.O. Box 15400, FI00076 Aalto)

Modern desktop computers are equipped with graphics cards that provide massive parallel computation power that was previously available only in supercomputers. On the other hand, there are several room acoustics modeling methods, but only some of them scale well to hundreds or thousands of parallel processors. The scalability of the acoustic radiance transfer method is examined. It is shown that it can almost fully utilize the available computing power. In simple cases, this technique achieves real-time performance. While taking into account the limitations of the energy-based acoustic modeling approach, the presented system can model arbitrary reflections . The reflections are presented as bi-directional reflectance distribution functions, which depend on the incoming and outgoing directions of acoustic energy. It is also possible to add time-dimension to such a reflection model. Some measurements are presented to show that spreading in time dimension occurs at reflections. Most of the previous room acoustics modeling techniques have ignored that phenomenon, but the acoustic radiance transfer technique can be easily modified to take this spreading effect into account.

11:05

1aAAb7. Efficient simulation of active sonar using concepts from room acoustics and auralization. Jason E. Summers (Appl. Res. in Acoust., LLC, Washington, DC 20024-2302)

Many systems for real-time simulation of active sonar reflect the historical division between propagation and reverberation algorithms. Though sonar returns comprise continua of echoes, systems treat scatterers as discrete targetlike entities or as distributed entities described by scattering strengths associated with regions of the ocean boundaries or volume. This approach limits the development of computational algorithms. In the fields of room acoustics and virtual reality there has been significant research devoted to development of efficient algorithms that enable computational simulation and real-time rendering (auralization) of sound fields resulting from complex scenarios. Such work exploits knowledge of the physical processes and limitations of the human auditory system to enhance the relevant aspects of fidelity while reducing computational load. Similar conditions exist for the simulation of active sonar. This presentation describes how concepts from room acoustics and auralization can be applied toward an active-sonar simulation that is scalable and implicitly incorporates tradeoffs between speed and accuracy. Rather than develop new propagation or reverberation algorithms, gains can be achieved by creating a system that treats all echos within a unified framework and explicitly accounts for properties of source, path, and receiver in order to optimize use of existing algorithms.

Contributed Papers

11:25

1aAAb8. Real-time auralization of wave simulation in complex three-dimensional acoustic spaces. Nikunj Raghuvanshi, John Snyder (Microsoft Res., Redmond, WA 98052, nikunjr@microsoft.com), Ravish Mehra, Ming C. Lin (Dept. of Comput. Sci. Sitterson Hall, Univ. of North Carolina, Chapel Hill, NC), and Naga K. Govindaraju (Microsoft Corp., Redmond, WA 98052)

A technique has been developed for modeling real-time sound propagation in static acoustic spaces that relies on pre-computed wave simulation. The associated system can auralize propagation that includes diffraction, interference, scattering and late reverberation, while supporting tens of moving point sources and moving listener in highly complex three-dimensional scenes. Since direct storage of simulated impulse responses for runtime use is infeasible, a novel technique was developed to extract and compactly encode the perceptually salient information in the simulated band-limited impulse responses. The response is automatically broken into early reflections (ERs) and late reverberation (LR), via a threshold on the temporal density of arriving wave-fronts. The LR is simulated and stored once per room. Detailed spatial variation in ER is simulated, and encoded by a set of peak delays/amplitudes in the time domain and a residual frequency response sampled in octave bands, at each source/receiver point pair on a five-dimensional grid. An efficient run-time uses this pre-computed representation to perform binaural sound rendering based on frequency-domain convolution. The system demonstrates audible wave-based effects in real time—diffraction low-pass filtering behind obstructions, sound focusing (caustics), hollow reverberation in empty rooms, sound diffusion in fully-furnished rooms, and late reverberation with non-exponential decay.

11:40

1aAAb9. Comparison of room-acoustical parameters predicted using different surface-reaction models. Behrooz Yousefzadeh and Murray Hodgson (Acoust. and Noise Res. Group, Univ. of BC, 2206 East Mall, 3rd Fl., Vancouver, BC, V6T 1Z3, Canada)

This paper presents the development of a beam-tracing model for calculating the transient responses of rooms. The model is wave-based (i.e., includes phase changes due to distance traveled and wall reflections), and can

be applied to rooms with extended-reaction surfaces. Room surfaces can be modeled as multiple layers of solid, fluid, and poroelastic materials; their acoustical properties are calculated using a transfer-matrix approach. The beam-tracing model calculates the complex transfer function of a room. Pressure impulse responses are then computed via Fourier transformation, and the room-acoustical parameters derived. Since pressure impulse responses are calculated, the model can also be used for auralization. The model has been applied to different room configurations in order to study the effects of different surface-reaction models on the predicted steady-state characteristics and temporal variations of sound-pressure fields in various room configurations. In particular, the audibility of using different boundary conditions (local versus extended reaction, wave-based versus energy based modeling, and phase changes on reflection) on the room-acoustical parameters has been investigated: In each configuration, room parameters have been calculated using different boundary conditions, and audible variations of the parameters have been studied and explained.

11:55

1aAAb10. Customized room acoustics simulations using scripting interfaces. Arthur W. van der Harten (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

Geometrical computer modeling is commonly used in consulting practice to make predictions of room acoustical quality. A large number of new parameters have been created by academicians and acousticians in recent years—so much so that it has been difficult for room acoustics simulation providers to include them all. This presentation will demonstrate how customized scripted simulations in Rhinoceros (a CAD program), using Pachyderm (an open source geometrical acoustics simulation engine), can allow the use of new parameters, or parameter simulation types, without reliance on software vendors to explicitly release software versions implementing new code.

MONDAY MORNING, 23 MAY 2011 — ISSAQUAH, 8:00 TO 11:50 A.M.

Session 1aABa

Animal Bioacoustics, Noise, and Underwater Acoustics: Ambient Noise and Marine Mammals

Michael B. Porter, Cochair
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Christian P. de Moustier, Cochair
Heat Light and Sound Research, Inc., 3366 N. Torrey Pines Ct., La Jolla, CA 92037

Chair's Introduction—8:00

Invited Papers

8:05

1aABa1. Merchant ship-radiated noise source levels. Stephen C. Wales (Naval Res. Lab., Code 7120, Washington, DC 20375, steve.wales@nrl.navy.mil) and Richard M. Heitmeyer (Global Strategies Inc., Crofton, MD 21114)

Ship-radiated noise is the principal source of noise in the 20–300 Hz frequency regime. This presentation provides a review of the issues relating to measuring ship-radiated noise source levels and predicting the source levels from ship parameters. The process of measuring and calculating the source levels is covered, including a discussion of propagation effects and source representation. Based on measurement results it is shown that, contrary to the classical model of shipping source levels, there is a negligible correlation between the source levels of an ensemble of ships and the transiting speeds and lengths of those ships. Issues concerning using the source level model in a modeling environment are discussed, including some effects of changing speed and source depth. Additionally, evidence is presented that predictions of increases in the source levels of the world's ships based on increases in their speeds is not justified, while increases due to size (length) are not trivial. [Work supported by the ONR through NRL-base funding.]

8:35

1aABa2. Mid-basin deep-water low-frequency ambient noise estimation. William M. Carey and Richard B. Evans (Mech. Eng., College of Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Estimating the increase in noise due to commercial shipping is of interest because of naval operations and its environmental consequences to marine life. Recent low-frequency noise calculations [Evans and Carey, Proceedings of the 9th ICTCA, Univ. Bundeswehr, DE] for the mid-Philippine sea illustrate the major uncertainties due to the basin's slope reflectivity, the number density, and the source level radiation characteristics of commercial ships. Canonical source level characteristics are based on naval studies on measurements some 3 decades ago. In the early seventies, Ross [J.O.E. **30**(2), (2005)] estimated the rate of increase of noise levels based on an empirical relationship between the radiated source level and tonnage to be order 0.5 dB/year based on the commercial ships of that era. Currently there are as many as seven classes of ships with an order of magnitude increase in tonnage, hull size, drafts, and propeller size. A plausible consequence is a radiation characteristic with a different directionality and effective efficiency. Deep ocean calculations are presented illustrating the uncertainties of slope enhancement and shipping noise levels. The current commercial ships are reviewed, qualitative estimates of the radiation characteristics are presented, and ambient noise implications discussed. [Work sponsored by ONR OA.]

9:05

1aABa3. Ocean traffic noise in the context of natural ambient noise in impacts on marine mammals. Douglas H. Cato (Defence Sci. & Tech. Org. and Univ. of Sydney, P.O. Box 44, Pyrmont, New South Wales, 2009, Australia, doug.cato@sydney.edu.au)

Traffic noise, the background noise from distance shipping, is the most widespread anthropogenic component of ambient noise. There has been concern for decades that traffic noise may be limiting the ability of marine mammals to communicate, but there is little direct evidence of actual impacts or the significance of these on the well being of populations, possibly because of the difficulties in obtaining such evidence. This paper compares traffic noise with natural ambient noise in the same frequency band. Marine mammals have evolved to cope with the range of levels in natural ambient noise. It draws on studies from areas near Australia where there is a wide range in traffic noise levels. In some areas, traffic noise is so low that it is possible to determine the range of natural ambient noise in the frequency band where traffic noise usually dominates. In other areas, traffic noise reaches levels similar to some of the high levels observed near North America. Humpback whales that migrate along the east coast of Australia are subject to high levels of traffic noise and noise from passing ships, but there seems to be little impact on the whales at the population level.

9:35

1aABa4. Using auditory models to study masking due to anthropogenic sound. David C. Mountain (Dept. of Biomedical Eng., Boston Univ., 44 Cummington St., Boston, MA 02215, dcm@bu.edu)

The mammalian auditory system is a highly evolved acoustic signal processing system that performs well even in highly reverberant and cluttered acoustic environments. In cetaceans, the auditory system is even more highly evolved than vision and is extremely important for navigation, foraging, and social communication. As humans inject more and more acoustic energies into the marine environment, these important acoustic functions may become compromised. Unfortunately little is known about the impact of anthropogenic sounds that could mask biologically significant signals. In this study a desktop simulation environment was used to study masking effects in a variety of conditions. The acoustic scenarios were created by mixing cetacean vocalizations recorded under relatively quiet conditions with scaled recordings of shipping noise. Biophysical computer models (http://earlab.bu.edu) based on physiological and behavioral experiments performed on humans were extrapolated to represent several different cetacean species. Model parameters for species of interest were estimated from behavioral audiograms and from other available data. These models were then used to predict how different types of biologically significant sounds are represented in neural firing patterns and how the neural representation degrades in the presence of anthropogenic noise.

10:05—10:20 Break

Contributed Papers

10:20

1aABa5. Oceanic shipping soundscapes. Christian de Moustier and Michael Porter (HLS Res., Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, cpm@hlsresearch.com)

Shipping and wind are key sources in the oceanic soundscape that affects marine mammal habitats. A new method of forming such soundscapes is presented. Frequency and range dependent transmission losses are precomputed from a grid of virtual sources using fast ray computations (BELLHOP) on a specified number of radial lines. Each radial line samples the bathymetry along its bearing out to a given maximum range. A shipping soundscape is then estimated by assigning a source spectral density level (dB *re* 1 μ Pa^2/Hz) and a shipping density (number of ships per unit area per unit time) to the various grid nodes. Such density values are obtained directly from ships carrying an automatic identification system (AIS) that transmit information such as ship type, position, heading, and speed. They can be obtained also from compiled statistics of AIS data (e.g., number of transits per year in an area). The same gridding approach is used to predict wind-generated sound levels based on maps of average wind speeds in an area for a given epoch, or on maps of forecast wind speeds.

10:35

1aABa6. Application of automatic identification system information to ocean soundscape modeling. John E. Joseph and Christopher Miller (Dept. of Oceanogr., Naval Postgrad. School, 1 University Cir., Monterey, CA 93943)

The impact of anthropogenic noise on marine life is an important issue to both the scientific community and public policy makers. Human-generated noise has potential to disrupt critical marine mammal biological functions such as foraging, communication, and navigation. Commercial shipping contributes significantly to the ocean soundscape, typically dominating the noise field at frequencies less than 500 Hz. Market conditions, trends in vessel design and propulsion, use of more economical ship routes, operational efficiency, and environmental factors are all important variables that help shape the changing soundscape. To reliably model the temporal and spatial variability of a regional soundscape, accurate characterization of the sources of noise is needed. Acoustic recordings taken at the Point Sur Ocean Acoustic Observatory (OAO) and Automatic Identification System (AIS) reports broadcast by ships passing the OAO site have been used to determine ship source levels over the 25–600 Hz band, categorized by ship class and speed. Source levels are then applied to a model used to evaluate temporal variability of the noise field at several sites along the central California coast based on AIS-reported shipping traffic transiting the region. Results of our calculations are presented and discussed. [Research supported by US Navy CNO(N45).]

10:50

1aABa7. Passive acoustic monitoring near Pt. Sur, California, in 2008–2009. Tetyana Margolina, Christopher Miller, John E. Joseph, Ching-Sang Chiu, and Curtis A. Collins (Dept. of Oceanogr., Naval Postgrad. School, 1 University Cir., Monterey, CA 93943)

How sounds from human activities affect behavior of marine mammals is in focus of many ongoing research projects. An important step in this direction is to develop regional databases of marine mammal vocalizations and sounds generated by human activities and natural sources. Analysis of passive acoustic recordings collected on top of Sur Ridge, near the Point Sur Ocean Acoustic Observatory (OAO), in 2008–2009 is presented. The data have been acquired with high-frequency acoustic recording package in the 10 Hz–100 kHz frequency band at a 200 kHz sampling frequency. These recordings have been scanned to detect and identify signals from various underwater acoustic sources. Pressure spectrum level has been analyzed in time/frequency space to reveal interannual, seasonal, and diel variabilities, as well as possible correlations of different sounds, with primary focus on anthropogenic sounds and marine mammal vocalizations. [Research supported by US Navy CNO(N45).]

11:05

1aABa8. Shipping noise signatures. Val Veirs, Scott Veirs (Beam Reach Marine Sci. and Sustainability School, 7044 17th Ave. NE, Seattle, WA, 98115, val@beamreach.org), and Jason Wood (SMRU and The Whale Museum, Friday Harbor, WA, 98250)

Throughout 2010, underwater recordings have been made of each ship passing two separate Haro Strait nodes of the OrcaSound.net hydrophone network. About 20 ships pass each day. Each ship has been identified in real time [automatic identification system (AIS)]. Measurements of received underwater noise levels and AIS variables are recorded as each ship passes the listening stations. Individual ships are observed multiple times moving in either northerly or southerly directions at times separated by a day or two and also by intervals of months. A database has been developed that contains the spectrum level of each ship (bandwidth 96 kHz at one location and 22 kHz at the other) and the source level both in terms of intensity and angular distribution. Ship signatures in terms of frequency quantiles and angular distributions of emissions are quite reproducible. This database can be used to predict limitations on echolocating and vocalizing marine mammals' active space due to specific ship noise emissions. In particular, predictions of marine mammal noise exposures in specific frequency bands can be made prior to specific vessels' entry into an area opening the possibility of planning field observations to investigate correlations between behaviors and specific predicted noise exposures.

11:20

1aABa9. Measurements of radiated underwater noise from modern merchant ships relevant to noise impacts on marine mammals. Megan F. McKenna, Donald Ross, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0205, jhildebrand@ucsd.edu)

There is mounting concern over the effect of ship noise on marine mammals; however, limited empirical data quantifying this noise impede our ability to evaluate impacts. An opportunistic approach for measuring radiated ship noise (20–1000 Hz) was used in this study. Calibrated acoustic data were combined with archived information on seven types of modern merchant ships transiting the coast of southern California. Three metrics for describing ship noise were applied: received sound levels (RLs) during 1 h passages, estimated source levels (SLs), and sound exposure levels (SELs). 1 h passages provided an estimate of the spatial extent of ship noise. At 40 Hz, container ships elevated noise above background up to 7 km forward of the ship and 19 km aft; bulk carriers elevated noise above background up to 5 km at bow and stern aspects. These ship-types had similar broad band estimated SL, 186 dB *re* 1 μPa at 1 m. The cumulative exposure to ship noise varied by ship type; we presented equations for estimating SELs for specific ship types. In concert, these metrics create a tool for quantifying ship noise within coastal marine environments, and can be used to assess the impact of ship noise on marine mammals. [This work was supported by the U.S. Navy CNO N45 and additional funds from the ONR, the NOAA, and the NSF.]

11:35

1aABa10. Behavioral response of harbor porpoises to vessel noise in a tidal strait. Brian Polagye (Dept. of Mech. Eng., Univ. of Washington, Box 352600, Seattle, WA 98195, bpolagye@uw.edu), Jason Wood (Sea Mammal Res. Unit Ltd., Vancouver, BC, V6R 1J6, Canada), Chris Bassett (Univ. of Washington, Seattle, WA 98195), Dom Tollit (Sea Mammal Res. Unit Ltd., Vancouver, BC V6R 1J6, Canada), and Jim Thomson (Univ. of Washington, Seattle WA 98105)

Admiralty Inlet, a narrow channel in Puget Sound, WA, is the proposed location of a pilot tidal energy project. A pair of hydrokinetic turbines would be deployed, for evaluation, on the seabed in approximately 60 m of water. When extracting power from strong tidal currents, these turbines will also generate broadband noise. Harbor porpoises are known to exhibit strong avoidance behavior to loud noise and occur frequently in this area. Consequently, there is a concern that the project could cause local displacement of this species. Because Admiralty Inlet is a major shipping lane and traversed by a passenger ferry, there is already periodic, high intensity anthropogenic noise in the vicinity of the proposed tidal energy project. The behavioral response of harbor porpoises to these existing noise sources is evaluated to provide context for the potential impact from tidal turbine noise. This study combines data on shipping and ferry traffic from an automatic identification system receiver, received noise levels from broadband autonomous hydrophones, and current velocity from Doppler profilers. These are correlated with porpoise presence, as assessed by echolocations detected by Chelonia C-Pods. Information collected over a full year provides insight into behavior at several time scales.

MONDAY MORNING, 23 MAY 2011 ASPEN, 8:25 TO 11:15 A.M.

Session 1aABb

Animal Bioacoustics and Acoustical Oceanography: Active Acoustic Applications to Bioacoustics Research

Elizabeth Kusel, Chair
Oregon State Univ., NOAA/PMEL, 2030 SE Marine Science Dr., Newport, OR 97365

Chair's Introduction—8:25

Invited Papers

8:30

1aABb1. Using active acoustics to reveal patchiness in the coastal ocean and its ecological consequences for plankton, fish, birds, and marine mammals. Kelly Benoit-Bird (College of Oceanic and Atmospheric Sci., Oregon State Univ., 104 COAS Admin Bldg., Corvallis, OR 97331, kbenoit@coas.oregonstate.edu)

In the ocean, most resources are heterogeneously distributed and highly dynamic. This patchiness in time and space has significant consequences for population dynamics, trophic interactions, community organization and stability, and the cycling of elements. However, this heterogeneity also presents a significant sampling challenge. A combination of active acoustic tools was used to quantify the relationships between predators and their prey in a variety of marine systems. The results show that patches of prey can have ecosystem impact disproportionate to their biomass. We found that the number and intensity of aggregations at each trophic level rather than the biomass in each step of the food chain involved were the most significant predictors of variation in adjacent trophic levels in the pelagic sub-arctic, temperate, and tropical systems examined. The importance of spatial pattern in ecosystems has long been recognized and its effects on predator-prey pairs has been examined in a number of previous studies; however, patchiness as the dominant force regulating an entire system has not been previously demonstrated, primarily because of the technical challenges of measuring the spatial and temporal scales of biological variability in the ocean.

8:50

1aABb2. Imaging and localizing fish and marine mammals with acoustics. Purnima Ratilal, Duong Tran, Roger Gong, David Reed, Hari Chauhan (Dept. of Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115), and Nicholas Makris (MIT, Cambridge, MA 02139)

During the 2006 ocean acoustic waveguide remote sensing (OAWRS) experiment in the Gulf of Maine, large shoals of Atlantic herring were instantaneously imaged in the frequency range from 300 to 1200 Hz. Simultaneously, several thousand instances of marine mammal vocalizations were passively recorded on a high resolution towed horizontal receiving array. A vast majority of the vocalizations were from humback whales in the 300–600 Hz frequency range. Vocalizations from other marine mammals species ranging from 40 Hz to over 3 kHz were also recorded. Various approaches are employed to localize and track the whales both passively and actively. The bearing of calling whales can be found by beamforming their vocalizations measured on the receiving array. An efficient and robust matched filter is designed to enhance the signal-to-noise ratio of the whale vocalizations. The array invariant method [Lee and Makris, J. Acoust. Soc. Am. (2006)] is then applied for instantaneous whale range estimation. The array invariant approach has been verified theoretically with modeled complex nonlinear whale vocalizations propagated long ranges exceeding 50 km in a range-dependent ocean waveguide. The whale range estimates obtained with the array invariance technique are verified by hyperbolic localization of the vocalization signals measured by the moving receiver array along a given track.

9:10

1aABb3. Active acoustic monitoring systems for detecting, localizing, tracking, and classifying marine mammals and fish. Peter J. Stein (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pstein@scisol.com)

Detection, localization, tracking, and classification (DLTC) of marine mammals and fish is necessary for a wide range of bioacoustic studies. This includes those related to understanding anthropogenic effects and to the development of methods for mitigating harm. Active acoustic monitoring (AAM) is a robust method for monitoring marine life as it can detect and accurately localize a silent target, enabling full DLTC. With the growth of the offshore renewable energy industry and the need to mitigate harm from pile driving, seismic surveys, and military sonar operations, there is strong interest in developing AAM systems and integrating them with current mitigation techniques. There are a host of significant issues including the standard sonar problems of reverberation and propagation in high-clutter shallow water environments, false alarms, classification, methods of deployment, and cost. Furthermore, AAM systems transmit acoustic energy that has the potential to disturb marine life. Much work lies ahead to develop systems that balance the risks, benefits, performance, and costs. This paper will review the status and issues of AAM systems. This includes a discussion of implemented near-field (imaging) and far-field (tracking) systems, experimental results, and plans for further development, testing, integration, and permitting.

Contributed Papers

9:30

1aABb4. Acoustic characterization of pingers on Queensland Shark Control nets. Christine Erbe, Craig McPherson, and Andrea Craven (JASCO Appl. Sci., Brisbane Technol. Park, P.O. Box 4037, Eight Mile Plains, QLD 4113, Australia)

Active acoustic applications in marine bioacoustics include the use of pingers to dissuade marine mammals. Pingers are most frequently used by the fishing industry to prevent depredation and entanglement. Pingers are also used by other marine industries to dissuade animals from potentially dangerous sites, e.g., underwater turbines. The Queensland Shark Control uses pingers to prevent marine mammal entanglement in shark control nets along public beaches. We have recorded and characterized some of the most common pingers off Queensland. Sound propagation and the potential detection of pinger sounds by marine mammals were modeled. Ambient noise was recorded in the vicinity of shark nets to estimate the contribution of pingers to ambient sound budgets. [Work supported by the Australian Department of the Environment, Water, Heritage and the Arts; Australian Antarctic Division.]

9:45

1aABb5. Application of active acoustic techniques to studies of krill aggregation and interaction with higher predators. Gareth L. Lawson, Andone C. Lavery, Peter H. Wiebe, and Nancy J. Copley (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, glawson@whoi.edu)

Large aggregations of euphausiids are often observed in regions of abrupt topography such as continental shelf breaks and submarine canyons. Understanding the biological and physical factors that lead to such aggregations is an important problem as these regions often constitute key habitat for top predators. A series of three cruises was conducted to the margins of Georges Bank, sampling zooplankton with a broadband active acoustic system, a multi-frequency acoustic system, a video plankton recorder, and depth-stratified nets, and sampling seabirds and marine mammals via visual observations. A combination of coarse-scale acoustic mapping and fine-scale adaptive surveys were used to identify and track individual euphausiid aggregations and to observe how their structure and vertical position varied with changing conditions. Distinct spatial and temporal variabilities were observed in euphausiid abundance, patch structure, and community composition, along with changes in the abundance and distribution of higher predators. These cruises represent among the first uses of the broadband technology for the study of zooplankton ecology and allow for an assessment of the relative advantages and disadvantages of broadband vs multifrequency approaches and of the importance of independent ground-truthing information.

10:00—10:30 Break

10:30

1aABb6. Frequency shifts of echolocation pulses in a cluttered environment: Comparison between *Pipistrellus abramus* and *Miniopterus fuliginosus*. Toshiya Takenaga, Shizuko Hiryu (Doshisha Univ., Kyotanabe 610-0321, Japan), James A. Simmons (Brown Univ., Providence, RI 02912), Hiroshi Riquimaroux, and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe 610-0321, Japan)

Changes in frequency range (ΔF) of FM echolocation pulses were recorded with an onboard wireless microphone (Telemike) for *Pipistrellus abramus* and *Miniopterus fuliginosus* in a cluttered environment created by dense chain-row obstacles. The duration of echo streams (ESD) from the chains reached ~40 ms when the bats were flying toward the chain-row

obstacles. *P. abramus* emitted pulses in pairs (strobe group) with 20–40 ms interpulse intervals (IPIs), which occasionally created pulse-echo ambiguity (IPI was shorter than ESD). When successive echo streams did not overlap, ΔF within strobe groups was less than 2–3 kHz. When overlap occurred, ΔF increased to 5–6 kHz (F-test, $P < 0.001$). By shifting frequency range of successive FM pulses, *P. abramus* appeared to assign echoes to the corresponding pulses and avoid pulse-echo ambiguity. Similar finding was previously reported in *Eptesicus fuscus*. On the other hand, *M. fuliginosus* never emitted pulses in a strobe group, and IPIs were always adjusted to be longer than ESDs. Thus, the amount of ΔF was less than 2 kHz, which corresponded to that in *P. abramus* without pulse-echo ambiguity. These comparative results suggest that echolocation strategy in the cluttered environment is different between two FM bat species. [Work supported by JSPS and ONR.]

10:45

1aABb7. Spatial and temporal patch dynamics of mysid swarms using combined stationary and shipboard active acoustics. Amanda M. Kaltenberg, Kelly J. Benoit-Bird, and Chad M. Waluk (College of Oceanic and Atmospheric Sci., Oregon State Univ., 104 COAS Admin. Bldg., Corvallis, OR 97331, akaltenb@coas.oregonstate.edu)

Mysid populations form into dense swarms and provide an important prey resource for large predators on the central Oregon coast including resident gray whales and a variety of fish species. The spatial and temporal characteristics of mysid patches were investigated in July and August, 2010 using combined stationary and shipboard active multifrequency acoustics. Mysid patches were analyzed to characterize the dynamics relevant to their availability as prey to larger predators. Concurrent net and video sampling provided validation of mysid species and size-class identification. Dense patches containing distinct size classes and mixed-size assemblages were observed from shipboard spatial surveys inshore of the 15 m isobath, where gray whales were actively foraging throughout the study. Bottom-mounted acoustic moorings revealed a distinct diel pattern of swarm density and vertical distribution, as mysid patches often separated into layers during night while reforming very near bottom during the day. This study demonstrates the usefulness of combining information from stationary and moving platforms of active acoustic methods while providing important information on the patch characteristics of a critical prey group.

11:00

1aABb8. Assessing juvenile Atlantic bluefin tuna schools in the Northwest Atlantic using sonar data and aerial imagery. Madeline L. Schroth-Miller and Tom C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 15 Mill Rd., Durham, NH 03824, m_schroth05 @yahoo.com)

Over the past 2 years, a feasibility study has been conducted in order to establish a methodology for assessing the biomass of juvenile Atlantic bluefin tuna (*Thunnus thynnus*). Over several days in August 2009, a 400 kHz Reson 7125 multibeam sonar installed on a commercial fishing vessel was used to collect acoustic backscatter from tuna schools. The multibeam sonar was oriented on the starboard side of the vessel to image a vertical slice of the water column. Because the fishing vessel was led to the tuna schools by a spotter plane, we were restricted to examining only near-surface tuna schools that were visible from the air. The same spotter plane collected aerial images of the same schools that were examined with the multibeam sonar. The multibeam sonar data allowed us to estimate attributes such as the maximum depth, cross sectional area, and morphology of the fish schools in a vertical plane, while metrics such as the nearest neighbor distance and number of fish were estimated from the aerial photographs. Taken together, the sonar data and aerial imagery provide a viable methodology for assessing juvenile Atlantic bluefin tuna.

MONDAY MORNING, 23 MAY 2011 — METROPOLITAN A, 8:00 TO 11:55 A.M.

Session 1aAO

Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Ocean Observing Systems: Acoustical Observations and Applications I

Timothy D. Duda, Cochair
Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543-1053

Brian D. Dushaw, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Contributed Paper

8:00

1aAO1. A brief personal introduction to the history and status of the ocean observing system and acoustical measurements. Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105, dushaw@apl.washington.edu) and Timothy F. Duda (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Integrated ocean observing systems (IOOSs) originated during the early 1990s as a way to provide valuable information to society as return on decades of investment in oceanographic research. The archetypical Argo program (www.argo.ucsd.edu) was initiated following the landmark OceanObs’99 conference and reassessed during OceanObs’09. With the possibility of significant sustained funding for comprehensive oceanic observations, one hallmark of planning conferences has been their political overtones. Regional ocean observing systems (www.ioos.gov, www.usnfra.org, and www .nanoos.org), purposely distinct from NOAA, began to develop in the early 2000s. The OOSes are not research programs, but sustained monitoring and information-providing services. The ocean observing initiative (OOI) developed out of the deep earth observing system (DEOS) idea of the early 2000s to deploy platforms over the world’s oceans for geophysical research. With the advent of significant funding for such platforms, the geophysical thrust was thrust aside in favor of two components: the undersea cabled networks and a small set of “Pioneer” platforms for oceanographic research. There has been little representation of acoustical techniques in these programs. Successful implementation of acoustical components for the IOOS requires the sustained cooperation and encouragement of three factions: the oceanographic community, the national and state funding agencies, and the acoustics community.

Invited Papers

8:15

1aAO2. The Northwest Association of Networked Ocean Observing Systems and opportunities for acoustical applications. Jan Newton, Matthew Alford, John Mickett (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, newton@apl.washington.edu), John Payne (POST (Pacific Ocean Shelf Tracking Project), Seattle, WA), and Fritz Stahr (Univ. of Washington, Seattle, WA)

NANOOS, the Pacific Northwest Regional Association of the U.S. Integrated Ocean Observing System (IOOS), aims to provide coastal ocean observations improving understanding and enabling decisions to improve safety, enhance the economy, and protect the environment. Biological aspects of ocean observing lag behind physical because of the difficulty of observing animal behavior beneath the surface; yet information about the behavior and survival of marine species in the ocean is identified as a critical need for fisheries management and marine spatial planning. Acoustics are poised to contribute to this need; however, limitations of acoustic arrays must be addressed. NANOOS recognizes opportunities for acoustical applications and has begun exploring these. NANOOS and POST deployed an acoustic receiver on a seaglider to test feasibility of using gliders to extend the scope of acoustic tracking arrays into deeper water and as a rapid deployment technology to test fixed arrays and investigate oceanographic features. During a 3-month test deployment, July–October 2010, a receiver mounted on the seaglider worked flawlessly and made two detections: one from a salmon and one from a Humboldt squid. NANOOS has proposed to develop methods for integrating instruments for tracking migrations of fish and marine mammals into ocean observing networks.

8:35

1aAO3. Acoustic challenges in aquatic ecosystem assessment. John K. Horne (School of Aquatic and Fishery Sci., Univ. of Washington, Box 355020, Seattle, WA 98195), Charles E. Schmid (Acoust. Society of America, Ste. 1NO1, 2 Huntington Quadrangle, Melville, NY 11747), Robert McClure (BioSonics Inc., 4027 Leary Way NW, Seattle, WA 98107), and Martin Siderius (Portland State Univ., P.O. Box 751, Portland, OR 97207)

In an effort to increase communication and collaboration between members of the Acoustical Society of America and the American Fisheries Society, a joint workshop focusing on the integration of biological and physical elements in ecosystem research, current policy and regulations, and applications of acoustic technologies developed field use will be held Thursday and Friday during the conference. Topics of interest in the workshop include the following: ecosystem based resource management, cabled observatories, monitoring renewable energy sites, acoustic species discrimination, statistical analysis for fisheries assessment, progress in fish tagging, acoustical models for fisheries assessment habitat inventory, detecting, and monitoring episodic events: natural and anthropogenic, active and passive assessment, and quantitative assessments in noisy environments. Ocean observatories is an explicit topic of interest in the workshop, but it can serve as an integrating theme that potentially expands the scope and capabilities of aquatic ecosystem assessment.

8:55

1aAO4. Listening to marine mammals at basin to local scales. Sue E. Moore (NOAA/Fisheries ST7, 7600 Sand Point Way NE, Seattle, WA 98115), Sofie M. Van Parijs (NOAA/NEFSC, Woods Hole, MA 02543), Brandon L. Southall (SEA Inc., Santa Cruz, CA 95060), and Kathleen M. Stafford (APL-UW, Seattle, WA 98105)

The successful use of SOSUS to track broad-scale occurrence patterns in whale calls during the second half of the 20th century fostered the development of autonomous recorders that can be deployed virtually anywhere in the world ocean. Over the past decade, data from these recorders have provided dramatic insights to marine mammal ecology. Patterns of call reception have demonstrated the near year-round occurrence of some baleen whale species in Arctic and Antarctic waters, a discovery that challenges long-held assumptions about the phenology of seasonal migrations. Integration of year-long calling records with physical oceanographic measures at mooring-based ocean observatories provides a means to include large whales in ecosystem-based models. The reception of anthropogenic sounds on nearly all recorders, whether deployed in coastal or remote areas, emphasizes the need to develop regional "soundscapes" based upon integrative sampling and analytical protocols. Examples from several long-term research programs will be provided as the basis for the strong assertion that passive acoustic observation of marine mammals is a vital component of any ocean observing system. Opportunities for future collaborations and the challenges of data management and access will be discussed.

Contributed Papers

9:15

1aAO5. Acoustic monitoring of beluga whales (*Delphinapterus leucas*) in Cook Inlet, Alaska. Manuel Castellote (Natl. Marine Mammal Lab., AFSC /NOAA, Seattle, WA), Robert Small (Alaska Dept. of Fish & Game, Juneau, AK), Shannon Atkinson (Univ. of Alaska Fairbanks, Juneau, AK), Marc O. Lammers (Hawaii Inst. of Marine Biology, Kaneohe, HI), Justin Jenniges (Alaska Dept. of Fish & Game, Juneau, AK), Anne Rosinski (Hawaii Inst. of Marine Biology, Kaneohe, HI), Christopher Garner (Joint Base Elmendorf-Richardson, Anchorage, AK), Sue Moore (NOAA Fisheries, Seattle, WA), and Whitlow L. L. Au (Hawaii Inst. of Marine Biology, Kanehoe, HI)

Cook Inlet belugas (CIBs) are listed as endangered under the U.S. Endangered Species Act. Their current seasonal distribution is essentially unknown and the factors impeding their recovery over the past decade are unknown, yet could include anthropogenic activities that impact their acoustic ecology, including coastal development, oil and gas exploration, shipping and military activities. Beginning in 2008, a cooperative research project has acquired new information on background noise levels and the seasonal presence of CIBs throughout Cook Inlet using passive acoustic monitoring. Mooring packages containing ecological acoustic recorders (EARs) and echolocation loggers (C-PODs) have been deployed at ten sites to continuously monitor the presence of CIBs. Cook Inlet is a challenging environment for acoustic monitoring because of extreme tides and currents, sediment dynamics, debris from rivers and seasonal ice that characterize the area. Noise from both natural and anthropogenic sources often make beluga call detection challenging. However, the effort to date has met with success and is providing valuable insights into beluga movement patterns and the

acoustic environment they face. This methodology also allows monitoring other odontocetes such as killer whales (Orcinus orca), detected mostly in the lower inlet, and harbor porpoise (Phocoena phocoena) detected throughout the inlet.

9:30

1aAO6. Seasonal and spatial patterns of cetacean occurrence off Oahu, HI observed acoustically. Marc O. Lammers, Michael Richlen, Whitlow W. L. Au, Anne E. Rosinski, and Gadea Perez Andujar (Hawaii Inst. of Marine Biology, P.O. Box 1346, Kaneohe, HI 96744)

The seasonal occurrence and distribution of cetaceans around island archipelagos is often poorly understood. This is generally due to the variability in sea surface conditions associated with islands, which make visual surveys prohibitive along exposed areas and/or during seasonal periods of rough seas. Historically, this has been the case for many cetacean species found in Hawaiian waters. Recently, however, concerns about the impacts of anthropogenic activities have created a need to better understand the long-term, spatial and temporal distributions of cetaceans in the archipelago. To meet this need, an effort was begun in February 2009 to study cetacean occurrence around the island of Oahu using a network of passive acoustic recorders. Five ecological acoustic recorders (EARs) were deployed in waters 115–575 m deep along the perimeter of the island and refurbished approximately every 4 months. Data from these deployments are providing an unprecedented perspective on the occurrence of both odontocete and mysticete cetaceans around the island. The southeastern corner of Oahu, in particular, has emerged as a hotspot of odontocetes; diversity and abundance. This and two other sites being monitored have historically received little or no visual survey attention, but are clearly important habitats for a variety of species.

9:45

1aAO7. Using acoustic cue statistics in matrix population models to study short-term and long-term marine mammal population dynamics in the northern Gulf of Mexico. Natalia Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, UL Box 44210, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy Ackleh, Nabendu Pal (Univ. of Louisiana at Lafayette, Lafayette, LA 70504), Juliette W. Ioup, George E. Ioup (Univ. of New Orleans, New Orleans, LA 70148), and Christopher O. Tiemann (Univ. of Texas at Austin, Austin, TX)

Acoustics is emerging as a viable tool for determining population trends of deep diving marine mammals in addition to conventional transect-line visual observations which are often limited by weather conditions, short-time surface presence, costs, etc. In September 2010, following the recent oil spill in the Gulf of Mexico, the Littoral Acoustic Demonstration Center (LADC) conducted a passive acoustic experiment as part of a study of short-term and long-term effects of the oil spill on the resident population of marine mammals. Environmental Acoustic Recording System (EARS) buoys were redeployed in three locations: 9 mi, 23 mi, and 50 mi away from the Deepwater Horizon-oil spill site. LADC previously collected data at these locations in 2001, 2002, 2007. The pre-spill data are used as a baseline for estimating long-term population trends. Densities of acoustic phonations of sperm whales, beaked whales, and dolphins are extracted from collected data and used for point estimates of the resident population density. LADC extensive work on an individual acoustic identification is reviewed in the context of mark-recapture statistics used in matrix population models. Trends in population density after the spill are presented and discussed. [Research is supported by ONR and NSF. Ship services donated by Greenpeace.]

10:00—10:30 Break

Invited Papers

10:30

1aAO8. Next generation science, engineering, and education in the ocean basins: sensor-robotic networks communicating near the speed of light. John R. Delaney (School of Oceanogr., RSN Program, Univ of Washington, Box 357940, Seattle, WA 98195-7940, jdelaney@uw.edu), Kendra L. Daly (Univ. of South Florida, St. Petersburg, FL 33701), Deborah S. Kelley (Univ of Washington., Seattle, WA 98195-7940), and Douglas L. Luther (Univ. of Hawaii, Honolulu, HI 96822)

Complex processes driven by solar and internal geothermal energy within the ocean basins constitute the “flywheel” of our planetary life-support system. Ocean-atmosphere dynamics determine the weather patterns and short-term climatic variations that continually impact the continents. New approaches to understanding the complexity and uncertainties of this “oceanic modulator” arise from the rapid implementation of submarine cabled networks providing unprecedented electrical power and communications bandwidth to thousands of sophisticated robot-sensor systems distributed throughout a full ocean environment. Empowered by cabled systems, oceanographers will benefit from emergent technologies driven by major investment from communities external to ocean sciences. Developments include robotics, biotechnology, cloud computing, *in situ* chemical and genomic sensors, digital imaging, nanotechnology, new visualization technologies, computational modeling, seismic tomography, passive and active acoustic sensors, and the internet. More powerful than any one of these emerging technologies is the convergence of the ensemble in pursuit of ocean science. As rapidly evolving capabilities are integrated into sophisticated, remote, interactive operations, a pervasive human tele-presence throughout our once ‘inaccessible’ global ocean will be realized. Such capabilities are required to meet environmental-societal challenges in the coming decades, which can only be addressed through optimally informed national and international collaborations.

10:50

1aAO9. Acoustic monitoring of marine life with a fiber-optic, ocean-observing network. Brandon L. Southall (SEA, Inc., 9099 Soquel Dr. Ste. 8, Aptos, CA 95003), Christopher Clark (Cornell Univ.), Kendra Daley (Univ. of South Florida), Sue Moore (Natl. Oceanic and Atmospheric Administration), John Payne (Univ. of Washington), Roger Payne (Ocean Alliance), Kate Stafford, Mark Stoermer (Univ. of Washington), Peter Tyack (Woods Hole Oceanograph. Inst.), William Wilcock, and John Delaney (Univ. of Washington)

The application of fiber-optic, high-bandwidth transmission technology is revolutionizing ocean observing by enabling the synoptic acquisition of high-density data streams, including acoustic measurements. A multi-disciplinary collaboration of geophysicists, acousticians, and biologists is developing acoustic observation systems within a cabled observing network being deployed off Washington and Oregon for the next 25 years. This system will include various sensors, including echosounders to detect zooplankton and fish, broadband hydrophone clusters for detecting various marine animals in biologically relevant areas, and low-frequency line arrays to locate and track vocalizing baleen whales across the Juan de Fuca plate region. These capabilities will enable monitoring of acoustically active individuals engaged in feeding, migrating, socializing, and other aspects of natural history. In combination with other tools (e.g., animal tags and remote sensors), these rich data streams will be integrated to monitor ecosystems and the physical and biological forces driving

their composition. The use of this powerful cabled monitoring network to synoptically observe and acoustically monitor marine life provides an unprecedented opportunity to systematically study this important area and the influences of climate variability and human activities on marine life. It also demonstrates the immense benefits for understanding the ocean with emerging technologies and cross-disciplinary collaboration.

Contributed Papers

11:10

1aAO10. Broadband acoustics on the VENUS observatory in Saanich Inlet. Tetjana Ross (Dept. of Oceanogr., Dalhousie Univ., Halifax, NS B3H 4J1, Canada, tetjana@dal.ca), Wu-Jung Lee (WHOI/MIT Joint Program, Woods Hole, MA 02543), Julie Keister (Univ. of Washington, Seattle, WA 98195), Ana Lara Lopez (Scripps Inst. of Oceanogr., La Jolla, CA 92093), and Charles Greene (Earth & Atmospheric Sci., Cornell Univ., Ithaca, NY 14853)

01
5/13/11
by: Scott Ve.

High-frequency sonar is by far the most cost-effective way of "profiling" the water column from an ocean observing system. From a biological oceanographic perspective, long-term acoustic observations are rich with information on the depths and abundances of fish and zooplankton. The drawback is that it is difficult to conclusively identify which species (or even functional groups) are present at any given time. This can be done, but only with plenty of supporting data, generally acquired non-autonomously. Broadband acoustics may be the key to making acoustic observations of fish and zooplankton less qualitative. Here we explore this idea. We present nearly two years (Apr. 2008–Feb. 2010) of broadband (85–155 kHz) echosounder data collected on the VENUS observatory in Saanich Inlet. Using historical and contemporaneous (July 30, 2009) zooplankton net-tow data, we attempt to automate and interpret the resulting classification of scattering layers throughout the long-term record.

11:25

1aAO11. Cabled observatory vent imaging sonar. Russell Light (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98195, russ@apl.washington.edu), Vernon Miller, Darrell R. Jackson (Univ. of Washington, Seattle, WA 98195), Peter A. Rona, and Karen G. Bemis (Rutgers Univ., New Brunswick, NJ 08901)

A cabled observatory vent imaging sonar (COVIS) has been developed to provide plume and Doppler imaging of hydrothermal vents and surrounding diffuse flow. The system was designed to be compatible with the power and data interface standards of the Neptune Canada cabled observatory. COVIS is a 4 m tall, titanium tripod employing a Reson 7125 multibeam sonar. The sonar transducers are positioned by a motor-driven three degree of freedom rotation system (pitch, roll, and yaw). A 400 kHz, 1 × 128 deg fan-beam projector is used with a receiver array that forms 256 beams having horizontal width 0.5 deg and covering a 128 deg azimuthal sector. Volumetric imaging of plumes is generated as the transducer array is scanned in 1 deg pitch steps. Doppler measurements of flow velocity over a 3-D grid are also derived. A 200 kHz, 28 × 128° broad beam projector is used to image the diffuse areas near the base of the hydrothermal vent edifices. Software allows for the creation of complex, arbitrary, autonomously executed experiments that control all aspects of the sonar and rotation system. COVIS was successfully deployed in September 2010. The design of COVIS provides insights relevant to future cabled acoustic systems. [Work supported by NSF.]

11:40

1aAO12. Multibeam sonar observations of hydrothermal flows at the Main Endeavour Field. Peter A. Rona, Karen G. Benis (Inst. of Marine and Coastal Sci., Rutgers Univ., New Brunswick, NJ 08901, rona@marine.rutgers.edu), Christopher D. Jones, and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98195)

The Cabled Observatory Vent Imaging Sonar has been deployed at the Main Endeavour Node of the Canadian Neptune cabled observatory and has acquired data on plume and diffuse hydrothermal flows. Based on the Reson 7125 multibeam sonar and operating at 200 and 400 kHz, two-dimensional and three-dimensional time series are produced using plume backscattering, Doppler shift, and acoustic scintillation. Hydrothermal plumes and diffuse flow are important as agents of transfer of heat, chemicals, and biological material from the mantle and crust into the ocean in quantitatively significant amounts. High-frequency sonar measurements offer the possibility of inversion to obtain fluxes of central importance in these processes. Long-term time series, obtainable in cabled systems, allow observations of hydrothermal response to tidal, tectonic, and volcanic forcing. Examples will be given of plume bending due to currents, determination of entrainment of ambient water, time variation of diffuse flows, and Doppler determination of volume flux. [Work supported by NSF Grants Nos. OCE-0824612 and OCE-0825088.]

Session 1aBA

Biomedical Acoustics: Medical Acoustics in Urology

Michael R. Bailey, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Robin O. Cleveland, Cochair
Boston Univ., Dept. of Mechanical Engineering, Boston, MA 02215

Contributed Paper

7:45

1aBA1. Real-time tissue change monitoring during the treatment of prostate cancer using Sonablate 500 with high intensity focused ultrasound. Wo-Hsing Chen, Narendra T. Sanghvi, Roy F. Carlson (Focus Surgery, Inc., 3940 Pendleton Way, Indianapolis, IN 46226, wchen@focus-surgery.com), Georg Schatzl, and Michael Marberger (Medical Univ. of Vienna, Waehringer Guertel 18-20, A-1090 Vienna)

Tissue change monitoring (TCM) during HIFU is an essential required feedback during the HIFU treatment. The Sonablate 500 (SB500) HIFU is enhanced with quantitative, real-time TCM software that estimate changes in tissue properties due to HIFU treatment of prostate cancer. TCM generates energy reading based on spectral analysis of the two-dimensional rf backscattered ultrasound signals acquired during HIFU. These energy changes are correlated to tissue temperature. TCM results are overlaid on the real-time ultrasound image in green, yellow, and orange to represent low, medium, and high degree of change in backscattered energy levels. To validate the TCM process five patients with histologically confirmed, organ confined prostate cancer were enrolled for the study. Needles containing three thermocouples were placed transperineally under TRUS guidance in the prostate to monitor temperatures from focal zone, posterior to the focal zone and on the lateral gland where no HIFU was applied. The measured temperatures in the HIFU treatment zones were from 70–114 °C (average 84 °C). The TCM results, estimated tissue temperatures were from 75 to 100 °C for 83% of treatment sites with an average tissue temperature of 91 °C.

Invited Paper

8:00

1aBA2. Design of dual mode high intensity focused ultrasound transducers for prostate therapy. Chapelon Jean-Yves, Bouchoux Guillaume, and Lafon Cyril (INSERM, LabTAU U556, Univ. of Lyon, 151 cours Albert Thomas, 69424 Lyon Cedex 03, France, jean-yves.chapelon@inserm.fr)

In HIFU applications, either MRI or ultrasound is used for treatment guidance and monitoring. When ultrasound imaging is used, a separate imaging transducer can be confocally aligned with the therapy probe itself. Yet, in endocavitary applications, such as transrectal prostate HIFU treatment, there are significant constraints on the size and geometry of the final probe, and the integratation of the imaging transducer with the therapy probe is a significant challenge. One solution to this problem is to use the same transducer elements for both imaging and therapy. Dual-mode transducers meet the specifications for therapy and imaging by taking into account the constraints linked to the features of the therapy transducer different from those of the imaging probe. This presentation discusses the results of simulations undertaken to determine an optimal geometry for a dual-mode probe that can be used both for thermal ablation of the prostate as well as for obtaining ultrasound images of sufficient quality for treatment guidance. A new dual-mode design is presented that allows for a smaller overall device than with separate imaging/therapy transducer designs, and Field II simulated images of prostate show that image quality equivalent to or better than those obtained with dedicated imaging probes can be obtained. [This research is supported in France by ANR Program Tecsan 2010.]

Contributed Paper

8:15

1aBA3. High speed imaging of shockwave-induced dynamics of cavitation bubbles and vessel wall. Hong Chen, Camilo Perez, Andrew A. Brayman, and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

High speed optical imaging under a microscope (high speed photomicrography) was used to observe shockwave-induced bubble dynamics and bubble-induced vascular dynamics. Ultrasound contrast agent microbubbles, serving as cavitation nuclei, were injected into the vessels of *ex vivo* rat mesentery. The bubbles were then insonated by focused shock wave pulses with peak positive pressures of 42 MPa and peak negative pressures of 10 MPa, generated by an electromagnetic shockwave source (Storz Duolith). The recorded images were analyzed to obtain bubble radius-time curves, vessel wall displacement, as well as their corresponding velocities. In general, bubble dynamics induces vessel distention (outward displacement of vessel wall) and invagination (displacement of vessel wall into the lumen). Comparisons of shockwave-induced dynamics with HIFU-induced dynamics will also be presented. [Work supported by NIH EB000350 and AR053652.]

Invited Paper

8:30

1aBA4. Numerical simulation of bubble dynamics in deformable vessels. Vedran Coralic and Tim Colonius (Div. of Eng. and Appl. Sci., California Inst. of Technol., 1200 E. California Blvd., MC 104-44, Pasadena, CA 91125, vcoralic@caltech.edu)

The growth and collapse of cavitation bubbles has been implicated as a potential damage mechanism leading to the rupture of blood vessels in shock wave lithotripsy (SWL) [Bailey *et al.*, in The Fifth International Symposium on Cavitation, Osaka, Japan (2003)]. While this phenomenon has been investigated numerically, the resulting simulations have often assumed some degree of symmetry and have often failed to include a large number of influential physics, such as viscosity, compressibility, surface tension, phase change, and fluid-structure interactions (FSI). We present here our efforts to explore the role that cavitation bubbles play in the rupture of blood vessels in SWL and to improve upon the current state of the numerical approach. We have developed a 3-D, high-order accurate, shock- and interface-capturing, multicomponent flow algorithm that accounts for the effects of surface tension and FSI. The preliminary results for the case of a bubble collapse, induced by a shock wave lithotripter pulse and occurring inside a deformable vessel, are presented. [This research was supported by the NIH (Grant No. 2PO1DK043881.]

Contributed Papers

8:45

1aBA5. Interface capturing simulations of acoustically driven bubble dynamics in and near tissue. Jonathan Freund, Arpit Tiwari (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61854), Ratnesh Shukla (Indian Inst. of Sci.), and Carlos Pantano (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61854)

Tissue injury during therapeutic ultrasound or lithotripsy is thought, in cases, to be due to the action of cavitation bubbles. Assessing this and mitigating it is challenging since bubble dynamics in the complex confinement of tissues or in small blood vessels are challenging to predict. Simulations tools require specialized algorithms to simultaneously represent strong acoustic waves and shocks, topologically complex liquid-vapor phase boundaries, and the complex viscoelastic material dynamics of tissue. We discuss advances in a simulation tool for such situations. A single-mesh Eulerian solver is used to solve the governing equations. Special sharpening terms maintain the liquid-vapor interface in face of the finite numerical dissipation included in the scheme to accurately capture shocks. A recent enhancement to this formulation has significantly improved this interface capturing procedure, which is demonstrated for simulation of the Rayleigh collapse of a bubble. The solver also transports elastic stresses and can thus be used to assess the effects of elastic properties on bubble dynamics. A shock-induced bubble collapse adjacent to a model elastic tissue is used to demonstrate this and draw some conclusions regarding the injury suppressing role that tissue elasticity might play.

9:00

1aBA6. Stone comminution using histotripsy. Timothy L. Hall, Alex P. Duryea, Charles A. Cain, and William W. Roberts (Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109, hallt@umich.edu)

This work explores the use of histotripsy to enhance urinary stone comminution ordinarily accomplished by lithotripsy. Histotripsy is a method of tissue ablation or erosion using high-energy pulsed therapeutic ultrasound to induce and control a cavitation bubble cloud. We have shown previously that histotripsy applied to model urinary stones causes surface erosion producing fine debris, which can be more readily eliminated. In this work, we show that optimal acoustic parameters can yield an effective comminution rate (96 mg/min) similar to a standard lithotripter (111 mg/min). As histotripsy is a surface erosion effect within the cloud of cavitation bubbles and lithotripsy has been shown to be most efficient at initial coarse fragmentation of stones, we propose a combined approach where the lithotripter coarsely fragments the stone, while histotripsy is used to erode these fragments with their greatly increased surface area.

Invited Paper

9:15

1aBA7. Holmium: YAG lithotripsy varies with power settings. Joel M. H. Teichman, Jason Sea, Lee Jonat, Ben Chew (Dept. of Urological Sci., Univ. of British Columbia, Vancouver, Canada), Jinze Qiu, and Thomas Milner (Biomedical Optics Program, Univ. of Texas, Austin, TX)

The holmium:YAG laser fragments stones by photothermal mechanism. Increased pulse energy (PE) produces larger ablation craters, implying faster lithotripsy. However, increased PE increases retropulsion, implying slower lithotripsy. Optimal power settings were studied. Uniform stone phantoms were ablated in water (500 J total energy). Six power settings were tested: ranging from 0.2 to 2.0 at 10–40 Hz. Two conditions were tested: no stabilization vs stabilization devices placed behind the stone. Total fragmentation (TF) and fragment sizes were quantified. In the no stabilization cohorts, retropulsion was measured. Pressure transients were measured by needle hydrophone. Stone crater volumes were quantified by optical computed tomography. With or without stabilization, TF increased as PE increased, $p<0.0001$; and fragment size increased as PE increased, $p<0.05$. Without stabilization, retropulsion increased as PE increased, $p<0.0001$. TF was greater with vs without stabilization, $p<0.01$. Pressure transients were <30 bars even at 2.0 J. Crater volumes increased as PE increased, $p<0.01$ but remained symmetric. Increased PE produces more lithotripsy but also larger fragments. Even at high PE (2.0 J) Ho:YAG lithotripsy is photothermal. Low PE produces small fragments but less lithotripsy. Modest PE (0.2–0.5 J) at high-repetition rate produces more fragmentation, small fragments, and less retropulsion. [Work supported by Percsys and Boston Scientific.]

9:30

1aBA8. Prototype for expulsion of kidney stones with focused ultrasound. Anup Shah, Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, 1959 NE Pacific St., Seattle, WA 98195), Bryan W. Cunitz, John C. Kucewicz, Yak-Nam Wang, Julianna C. Simon, Wei Lu, Peter J. Kaczkowski, and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105)

Residual fragments remain in over 50% of treatments for lower pole kidney stones. A second-generation device based on a diagnostic ultrasound system and scanhead has been developed with a unique algorithm for stone detection and the capability to focus ultrasound to expel residual fragments. Focused ultrasound was applied to a bead on string in a water tank as well as to human stones (<5 mm) implanted in the lower pole of a live porcine model via retrograde ureteroscopy. Histological samples were collected and scored in a blinded fashion for therapeutic exposures and for supertherapeutic levels. The *in-vitro* bead was visually observed to move under focused ultrasound. Even with progressive manual displacement of the bead, the system continuously tracked and caused bead movement in real time. In the live porcine model, stones were expelled from the lower pole to the ureteropelvic junction in seconds to minutes using pulses at a duty factor of 0.02 and 8 W total acoustic power. Injury was observed no more frequently than in controls. Occurrence of injury rose slightly above control at a duty factor of 0.02 and 80 W and at a duty factor of 1 and 8 W. [Work supported by NIH DK48331, NIH DK086371, and NSBRI through NASA NCC 9-58.]

9:45—10:00 Break

10:00

1aBA9. Assessing the effects of respiratory motion on stone comminution *in vitro*. Nathan Smith, W. Neal Simmons, and Pei Zhong (Dept. of Mech. Eng. and Mat. Sci., Duke Univ., 101 Sci. Dr., Durham, NC 27708)

The effects of respiratory motion on stone comminution produced by an electromagnetic (EM) shock wave lithotripter were evaluated *in vitro*. Individual spherical BegoStone phantoms (D = 10 mm) were placed in a flat-base tube holder and treated at various radial distances in the lithotripter focal plane. To assess the effects of respiratory motion on stone comminution, the holder was set into translational motions in another series of experiments with various excursion distances (1.5–3.0 cm), breath rates (12–24 bpm), and drift factors for motion randomization. Stone comminution tests were performed using either a newly designed acoustic lens with a wide focal width and a low peak pressure or the original lens under equivalent acoustic energies. The results shows that the new lens produces statistically higher stone comminution ($p < 0.01$) than the original lens. Moreover, stone comminution at various radial distances will be compared, together with cavitation potential calculated by the Gilmore model based on the pressure waveforms measured by an FOPH.

10:15

1aBA10. Clinical assessment of shockwave lithotripsy accuracy. Anup Shah, Jonathan D. Harper, Jonathan L. Wright, Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, 1959 NE Pacific St., Seattle, WA 98195), Marla Paun, and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105)

Kidney stone movement primarily due to patient respiration compromises shock wave lithotripsy (SWL) targeting and efficacy. The objective of this study is to describe the use of B-mode ultrasound to evaluate the accuracy of targeting during SWL. Patients undergoing electrohydraulic SWL were enrolled into this institutionally approved research study. A commercial diagnostic ultrasound imaging system, either Philips HDI 5000 or iU-22, was used to intermittently visualize and detect any shockwave-induced motion of the stone during 1–3 min periods. Four patients (mean age 52.7) underwent treatment of seven renal stones with mean individual stone size of 10.41 ± 4.5 mm. A mean of 2937 shocks (range 2750–3000) were delivered at a rate of 1–2 Hz and charging voltage of 14–26 kV. Stone oscillation or jumping at the exact time of individual shock delivery was visualized with ultrasound: no stones completely failed to move. Accurate alignment, as interpreted by positive stone motion, occurred in a mean of 50 ± 20.4% of shockwaves. Ultrasound imaging represents a method of real-time assessment of accuracy in SWL and may provide the basis for devices to control targeting so that shockwaves are only delivered when the stone is in focus. [Work supported by NIH DK43881 and DK086371.]

10:30

1aBA11. Interrogating and imaging renal stones using vibro-acoustography. Paul R. Illian, Jr., Dan Gross, Wei Lu, Neil R. Owen, Michael R. Bailey, and Pierre D. Mourad (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, rillian@apl.washington.edu)

Vibro-acoustography (VA) is an ultrasound interrogation and imaging technique with a variety of applications. Here it was used to identify optimal parameters for detecting and imaging kidney stones in phantoms. The parameters varied included the difference frequency and the position in time of the analysis window used for image construction. Experiments in a water tank were conducted using a focused PVDF membrane hydrophone (receiver) placed in a central opening of an annular, dual element transducer (source), itself mounted on a translation stage. Our source consisted of 90-ms pulses with a center frequency of 2.0 MHz and difference frequencies between 50 and 350 kHz, applied both on and off stone. Variations in the amplitude of the measured ultrasound backscatter and acoustic emissions as a function of difference frequency, between signals from stone and phantom, guided the choice of imaging parameters. The results were detailed images of renal stones measuring 10 dB above the background tissue. These findings suggest that spectral information from the scattering and reverberation of VA induced ultrasound can be used to guide the interrogation and imaging of kidney stones. [Work supported by NIH DK43881, NIH DK086371, and NSBRI through NASA NCC 9-58].

10:45

1aBA12. Investigation of the effect of signal amplitude on twinkling artifact. Wei Lu, Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Oleg A. Sapozhnikov, Peter J. Kaczkowski, John C. Kucewicz, Neil R. Owen, Michael R. Bailey, and Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105)

Twinkling artifact on color Doppler ultrasound is the color labeling of hard objects, such as kidney stones, in the image. The origin of the artifact is unknown, but clinical studies have shown that twinkling artifact can improve the sensitivity of detection of stones by ultrasound. Although Doppler detection normally correlates changes in phase with moving blood, here the effect of amplitude on the artifact is investigated. Radio-frequency and in-phase and quadrature (IQ) data were recorded by pulse-echo ensembles using a software-programmable ultrasound system. Various hard targets in water and in tissue were insonified with a linear probe, and rectilinear pixel-based imaging was used to minimize beam-forming complexity. In addition, synthesized radio-frequency signals were sent directly into the ultrasound system to separate acoustic and signal processing effects. Artifact was observed both in onscreen and post-processed images, and as high statistical variance within the ensemble IQ data. Results showed that twinkling artifact could be obtained from most solid objects by changing the Doppler gain, yet signal amplitude did not have to be sufficiently high to saturate the receive circuits. In addition, low signal but high time gain compensation created the largest variance. [Work supported by NIH DK43881, DK086371, and NSBRI through NASA NCC 9-58.]

11:00

1aBA13. Autoregressive ultrasound imaging method to enhance kidney stone twinkling and suppress blood flow. John C. Kucewicz, Bryan W. Cunitz, Barbrina Dunmire, Michael R. Bailey, and Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ of Washington, 1013 NE 40th St., Seattle, WA 98105)

"Twinkling" is a widely reported ultrasound artifact whereby kidney stones and other similar calcified, strongly reflective objects appear as turbulent, flowing blood in color and power Doppler. The twinkling artifact has

been shown to improve kidney stone detection over *B*-mode imaging alone, but its use has several limitations. Principally, twinkling can be confused with blood flow, potentially leading to an incorrect diagnosis. Here a new method is reported for explicitly suppressing the display of color from blood flow to enhance and/or isolate the twinkle signal. The method applies an autoregressive model to standard Doppler pulses in order to differentiate tissue, blood flow, and twinkling. The algorithm was implemented on a software-based, open architecture ultrasound system and tested by a sonographer on phantoms and on stones implanted in a live porcine kidney. Stones of 3–10 mm were detected reproducibly while suppressing blood flow in the image. In conclusion, a new algorithm designed to specifically detect stones has been tested and has potential clinical utility especially as efforts are made to reduce radiation exposure on diagnosis and monitoring. [This work was supported by the National Institutes of Health (NIH Grant No. DK43881) and the National Space Biomedical Research Institute through Grant No. NASA NCC 9-58.]

11:15

1aBA14. Modeling of radiation force imparted to an elastic sphere from an ultrasound beam of arbitrary structure. Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119991, Russia, olegs@apl.washington.edu) and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105)

The radiation force created by an acoustic wave incident on an elastic sphere is studied theoretically. Elastic spheres with properties similar to kidney stones are considered. An acoustic wave is taken in the form of a high-intensity focused ultrasound beam of megahertz frequency, which is typical for transducers proposed for stone therapy. To study radiation force of beams with arbitrary structure, the source excitation is modeled as a sum of plane waves of various inclinations (angular spectrum representation). First, a plane acoustic wave scattering at the stone is modeled using the known solution in the form of a spherical harmonics series. Then superposition of such solutions is used to calculate the scattered field from a focused beam. Once the acoustic field is known, the radiation stress tensor is calculated on a surface surrounding the sphere. Finally, the net force acting on the sphere is calculated by integrating the radiation stress along the surface. Numerical calculations show that the direction and value of the radiation force acting on the sphere depend on the pressure field structure in the region where the scatterer is positioned. [Work supported by NIH DK43881 and DK086371, RFBR, and NSBRI through NASA NCC 9-58.]

11:30

1aBA15. Real-time tracking of renal calculi displaced by the radiation force of focused ultrasound. Paul R. Illian, Jr., Bryan W. Cunitz, John C. Kucewicz, Michael R. Bailey, and Peter J. Kaczkowski (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab.-Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, rillian@apl.washington.edu)

An area of active research involves using the radiation force of ultrasound to expel small kidney stones or fragments from the kidney. The goal of this work is real-time motion tracking for visual feedback to the user and automated adaptive pushing as the stone moves. Algorithms have been designed to track stone movement during patient respiration but the challenge here is to track the stone motion relative to tissue. A new algorithm was written in MATLAB and implemented on an open-architecture, software-based ultrasound system. The algorithm was first trained then implemented in real-time on B-mode IQ data recorded from phantom experiments and animal studies. The tracking algorithm uses an ensemble of image processing techniques (2-D cross-correlation, phase correlation, and feature-edge detection) to overlay color on the stone in the real-time images and to assign a color to indicate the confidence in the identification of the stone. Camera images as well as ultrasound images showed that the system was able to locate a moving stone, re-target, and apply a new focused push pulse at that location. [Work supported by NIH DK43881, NIH DK086371, and NSBRI through NASA NCC 9-58].

11:45

1aBA16. Stress waves in kidney stones. Haibiao Luo and Robin O. Cleveland (Dept. Mech. Eng., Boston Univ., Boston, MA 02215, robinc@bu.edu)

A 3-D time-domain finite-difference solution to the linear elastic equations was applied to investigate the stress and velocity fields of kidney stones subject to lithotripsy shock waves. The kidney stone models were scanned from micro-computed tomography and had diameters from 2 to 5 mm. It was found for these shapes that shear waves induced by interference of the shock wave with stone boundaries dominated the high stress in the stones. The traditional belief of stone comminution mechanism by spall mechanism does not play important role due to the irregular proximal and distal stone surfaces. It was found for natural stones that stone orientation had an impact on the generation of high stress with a smooth convex surface producing the highest internal stresses. The results indicate that lithotripters with a focal width larger than a stone should be able to break a stone more efficiently since the large focal width shock waves result in stronger interaction with stone circumferences and produce larger shear waves.

12:00

1aBA17. Location of coupling defects influences stone breakage in shock wave lithotripsy. Guangyan Li, James A. McAteer, and James C. Williams, Jr. (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, 635 Barnhill Dr., Indianapolis, IN 46202-5120)

SWL is the most common treatment for kidney stones. However, compared to ureteroscopy and nephrostolithotomy, SWL is least effective—failing in ~50% of cases. Since stone breakage is highly effective under controlled conditions, acoustic coupling between the lithotripter and patient may be the weak link. Previous *in vitro* studies determined that air-pockets created in routine coupling reduce SW-amplitude by ~20% and defects occupying only 2% of coupling area reduced breakage 20%–40%. As a step toward determining if the position of defects influences SW delivery to the target we used styrofoam to selectively block portions of the coupling interface between a DoLi-50 and the test tank. Stone breakage was ~three times greater when the entire 13 cm diameter coupling interface was unblocked than when all but the center 6 cm was blocked, consistent with the reduction in surface area. However, the transition was abrupt, with ~70% of the loss in efficacy occurring upon reduction in the aperture from 7 to 6 cm. Reducing the aperture had a greater effect on $P-$ than $P+$ ($P+/P-$ no aperture ~41/−4.3 MPa; 7 cm ~42/−3.5 MPa; 6 cm ~37/−2.5 MPa). These initial findings begin to identify a region of the overall coupling interface where defects are likely to be problematic. [Work supported by Grant No. NIH-DK43881.]

Session 1aNS

Noise: Indoor Psychoacoustic Response to Outdoor Noise Sources

Alexandra Loubeau, Cochair
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Jonathan Rathsam, Cochair
NASA Langley Research Center, Structural Acoustics Branch, Hampton, VA 23681

Chair's Introduction—8:00

Invited Papers

8:05

1aNS1. Indoor human response to blast noise measured *in situ*. S. Hales Swift, Dan Valente, Edward T. Nykaza (U.S. Army Corps of Engineers, ERDC-CERL, P.O. Box 9005, Champaign, IL 61826, stephen.h.swift@usace.army.mil), and Kathleen Hodgdon (The Penn State Univ., State College, PA 16804)

As a result of suburban sprawl, the number of people living near military installations is drastically increasing. Coupled with an escalation of military activities and preparedness, the potential for noise generated by an installation to impact the surrounding communities has grown, especially for large amplitude impulsive events such as those generated during artillery training exercises. To assess the effect of blast noise on individuals living near installations, a large scale *in situ* study has been performed. The homes of study participants were instrumented and outdoor/indoor blast signature pairs of routine installation activities were captured over the course of 1 year. Participants filled out short questionnaires whenever they heard blast noise events. Measurements of single events at subjects' homes along with their responses present unique data with which to investigate the human response to blast noise on an event-by-event basis. In this presentation, the characteristics of the noise typically experienced by residents in their own homes will be examined and used to create dose-response relationships. Comparison will be made of dose-response curves based on annoyance, interference, and loudness as a function of level-based metrics, and as a function of a variety of psychoacoustic metrics. [Work supported by the Strategic Environmental Research and Development Program.]

8:25

1aNS2. A theory-based model of the prevalence of transportation noise annoyance. Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St., Woodland Hills, CA), Paul Schomer (Schomer and Assoc., Inc., Champaign, IL 61821, schomer@schomerandassociates.com), and Vincent Mestre (Landrum and Brown, Laguna Niguel, CA 92677)

Dosage-response relationships between cumulative noise exposure and the prevalence of annoyance in communities are generally developed by statistical curve fitting methods. Generic methods of this sort, such as regression, can characterize the central tendency of findings of social surveys, but provide no explanation for the great variability of these data. Further, confidence intervals around a regression curve do not yield prediction intervals appropriate for use in environmental disclosure documents. An alternative approach under consideration by ISO Working Group 45 for inclusion in an updated draft international standard is based on the hypothesis that the rate of growth of the prevalence of annoyance closely resembles that of the rate of growth of effective (that is, duration-corrected) loudness. A comprehensive database of the findings of 43 aircraft noise annoyance studies provides strong empirical support for the hypothesis.

8:45

1aNS3. A simple method for comparing social survey findings on the annoyance of transportation noise. Vincent Mestre (Landrum & Brown, 27812 El Lazo Rd., Laguna Niguel, CA, 92677, vmestre@landrum-brown.com), Paul Schomer (Schomer and Assoc., Champaign, IL 61821), and Sanford Fidell (Fidell Assoc., Woodland Hills, CA 91367)

Findings of social surveys about the annoyance of transportation noise vary greatly from one community to the next. Popular speculations about the sources of this variability have included situational differences in exposure, methodological differences in interviewing, and errors of measurement. The model described in the prior presentation provides a simple method for summarizing the community-specific variability in a single-valued metric, the community tolerance Index (CTI). Findings of several CTI-based analyzes are discussed in this presentation, along with some of their policy and regulatory implications.

9:05

1aNS4. Comparison of subjective test methodologies for assessing human annoyance to low-amplitude sonic booms. Denise M. Miller and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., University Park, PA 16802)

In a subjective test assessing human annoyance to low-amplitude sonic booms, different subjective testing methods were utilized. These included a hybrid categorical line scaling method as well as magnitude estimation with two different reference sounds. The test methods will be discussed, as well as their advantages and disadvantages. Other topics of discussion will include experimental design, data analysis, as well as subject participant preference. [Work sponsored by the NSF and the FAA through the PARTNER Center of Excellence.]

9:25

1aNS5. Overview of an indoor sonic boom simulator at NASA Langley Research Center. Jacob Klos, Alexandra Loubeau, and Jonathan Rathsam (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681)

A facility has been constructed at NASA Langley Research Center to simulate the soundscape inside residential houses that are ensonified by environmental noise from aircraft. The purpose of this facility, the interior effect room, is to examine parameters that affect psychoacoustic response in a controllable indoor listening environment. The single room facility, built using typical residential construction methods and materials, is surrounded on two sides by arrays of loudspeakers. These exterior arrays are used to simulate aircraft noise sources that transmit into a room of a typical house. The exterior sound reproduction system, which consists of 52 subwoofers and 52 mid-ranges in close proximity to the walls of the room, has been designed to enable study of sonic booms transmitted into residential structures and has a usable bandwidth of 3 Hz–6 kHz. In addition to these exterior arrays, satellite speakers placed inside the room are used to simulate rattle and other audible contact-induced noise that can result from low frequency excitation of a residential house. The layout of the facility, operational characteristics, acoustical characteristics, and equalization approaches are summarized. Current research efforts utilizing the facility are described in two companion papers.

9:45

1aNS6. Evaluation of new indoor sonic boom subjective test facility at NASA Langley Research Center. Alexandra Loubeau, Jonathan Rathsam, and Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681)

A sonic boom simulator at NASA Langley Research Center has been constructed for research on human response to low-amplitude sonic booms heard indoors. Research in this facility will ultimately lead to a psychoacoustic model for single indoor booms that will be validated by future community studies. The first subjective test was designed to explore indoor human response to variations in sonic boom rise time and amplitude. Another goal was to identify variability across listener locations within the facility. Finally, the test also served to evaluate the facility as a laboratory research tool for studying indoor human response to sonic booms. Subjects listened to test sounds and were asked to judge the annoyance relative to a reference boom. Measurements of test signals were conducted for objective analysis and correlation with subjective responses. Results confirm the functionality of the facility and effectiveness of the test methods and indicate that calculated loudness does not fully describe the indoor annoyance.

10:05—10:25 Break

10:25

1aNS7. Sonic-boom loudness test using new indoor sonic-boom subjective-test facility at NASA Langley Research Center. Jonathan Rathsam, Alexandra Loubeau, and Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681)

A sonic-boom simulator at NASA Langley Research Center has been constructed to research the human response to low-amplitude sonic booms heard indoors. The facility's initial goal is the development of a psychoacoustic model for individual sonic booms to be validated by future community studies. The current test assesses the suitability of existing loudness metrics for predicting indoor human annoyance to sonic-boom waveforms. The test signals consist of synthesized and recorded sonic-boom waveforms chosen to systematically vary the low-frequency content. Some waveforms are presented with and without high-pass filtering to examine the effect of low-frequency content on annoyance. Equally annoying presentation levels are determined among the test signals by paired comparison with a reference sonic-boom waveform. A second reference waveform is also used for some signals to examine if results change with the reference sound. Loudness metrics are then calculated for each measured test signal at the subjective-equality level. Loudness metrics are thus evaluated based on their ability to predict annoyance for a wide range of sonic-boom waveforms.

Contributed Papers

10:45

1aNS8. Sound fields inside street canyons with inclined flanking building facades. S. K. Tang and K. E. Piippo (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hong Kong, China)

Street canyons are common in modern cities. It is well known that the multiple sound reflections within the canyons tend to increase the noise levels inside the canyons. A scaled down model experiment was conducted in the present investigation to study the effect of the inclination of building facade on the sound field. A line source consisted of 100 2-in. aperture loudspeakers was used to simulate the road traffic source. The whole experiment was carried out inside an anechoic chamber. The canyon was 4 m long, 2 m high, and 1 m wide (1:4 scale down ratio). The case of a single facade was used acted as the reference. The reverberation inside the model canyon was strong when the two model facades are vertical (inclination 90 deg) and parallel to each other. However, it was found that such reverberation deteriorated very rapidly as the inclination of one of the model facade was reduced to 80 deg. The sound strength inside the model canyon was also reduced. The sound levels at the top region of the canyon decreased more rapidly. It was also found that the effect of the opposite facade was basically unchanged once its inclination was less than 60 deg.

11:00

1aNS9. A noise mapping study for heterogeneous road traffic conditions considering horn sounds. Kalaiselvi Ramasamy and Ramachandraiah Alur (Dept. of Civil Eng., Indian Inst. of Technol.-Madras, kalaiarchi @gmail.com)

In recent years noise mapping has become an increasingly useful tool for environmental noise assessment. Noise mapping is the process of determining and visualizing noise impact on the environment. The current noise mapping softwares are typically designed for the express and freeways of other countries where the roads are widely open with the homogeneous traffic conditions and speeds of vehicles touching 100 km/h. This situation is not suitable for Indian conditions where heterogeneity of traffic and honking of vehicles are the characteristics in an urban environment. A pilot study in typical areas of a metropolitan city (Chennai city) using soundplan software shows a difference of up to 7 to 10 dB(A) in Lden values between obtained and measured noise levels. A multiple regression model has been developed taking into account the horn aspects and heterogeneity. The statistical parameters such as L10, L50, L90, and Leq are measured along with the vehicle speed. The vehicle count and number of horn events are also observed during the field measurements with the video camera. The developed multiple regression models can be used as a plug-in in any open source GIS softwares such as QGIS, GRASS, etc., for noise mapping purposes. With this background this paper explains a methodology to build 3-D noise models for urban areas to analyze and visualize the 3-D distribution.

11:15

1aNS10. A new metric for the objective determination of steady state noise source measurement validity as affected by extraneous noise. Noel W. Hart, Robert D. Bruce (CSTI acoustics, 16155 Park Row, Ste. 150, Houston, TX 77084-5100, noel@cstiacoustics.com), Ralph R. Galetti (Boeing), and Mark Rubino (Industrial Noise Control, North Aurora, IL 60542)

The measured difference between the maximum sound level (L_{max}) and the equivalent sound level (L_{eq}) of a steady state source can be used to objectively demonstrate inconsistencies in measurement. This method has particular applications to field measurements made in an uncontrolled environment where extraneous, time-variant noise from wind, animals, people, or machinery often interferes. The method is also straight forward to implement with modern sound level meters, as the necessary data are easily recorded.

11:30

1aNS11. A study of noise mitigation techniques for explosive training scenarios. Michelle E. Swearingen (US Army ERDC-CERL, P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Donald G. Albert (US Army ERDC-CRREL, Hanover, NH 03755), and Dan Valente (US Army ERDC-CERL, Champaign, IL 61826)

Utilizing explosives to destroy questionable munitions is a standard procedure in the military. It is critical to have teams trained to perform these activities safely. Often the training involves practice in setting up and detonating relatively small charges of explosives, but these activities can cause annoyance in surrounding civilian populations. A study was performed at one military installation to determine best practices for managing the noise generated by these activities. Three methods were investigated: burying charges with sandbags, covering charges with a rubber blast mat, and spraying water over the charges during detonation. Acoustic and seismic measurements were performed at several distances between 4 and 2500 m in two directions to investigate the relative effectiveness of each method. The study found that covering the charges with sandbags provided a reduction in noise levels of as much as 15 dB in the far field with minimal impact on the training. Use of sandbags was therefore superior to the other methods investigated in maximum reduction, ease of use, and cost. This presentation will provide an overview of the study results.

11:45

1aNS12. Directivity and variability characterization of a propane cannon. Tom W. Noble and Dan P. Valente (Construction Eng. Res. Lab., US Army Engineer Res. and Development Ctr., P.O. Box 9005, Champaign, IL 61826, thomas.w.noble@usace.army.mil)

When studying the influence of an environment on blast noise propagation, it is often unrealistic to use typical blast noise sources such as plastic explosives or artillery fire. A propane cannon, designed as a bird scare-away device, is a reasonable surrogate source which produces a loud impulsive sound rich in low frequency energy. In order to confirm the cannon's utility as a blast surrogate, we examined the directivity of the propane cannon, the variability in received level as a function of distance, and the variability in one-third octave band SEL as a function of distance and direction. The results indicated that the propane cannon has low shot-to-shot variability and is nearly omni-directional within 60 deg of the forward facing direction.

MONDAY MORNING, 23 MAY 2011 WILLOW A, 8:45 TO 11:30 A.M.

Session 1aPA

Physical Acoustics: Interaction of Sound with Sound

Joel Mobley, Chair

Univ. of Mississippi, National Ctr. for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Contributed Papers

8:45

1aPA1. The acoustic levitation and simultaneous contactless transportation of matter in the third resonance mode (H3-mode) of a line-focused system. Daniele Foresti, Majid Nabavi, and Dimos Poulikakos (Dept. of Mech. and Process Eng., Inst. of Energy Technol., Lab. of Thermodynamics in Emerging Technologies, ETH Zurich, CH-8092, Zurich, Switzerland, dforesti@ethz.ch)

We investigate theoretically herein higher modes of resonance for transportation of matter (particles or droplets) in line-focused acoustic levitation. Contactless transportation was achieved by varying the height between the radiating plate and the reflector. Transportation and levitation of volatile liquids, in particular, involves two limits of the acoustic forces. The lower limit corresponds to the minimum force required to levitate a droplet, i.e., to overcome the gravitational force. The higher limit corresponds to the maximum acoustic pressure before atomization of the droplet occurs. By increasing the size of the droplet, the lower limit increases and the higher limit decreases. Therefore, in order to have large droplets levitated in the device, a relatively flat radiation pressure over the translation distance is needed. In this study, using a finite element model, the Gor'kov potential was calculated for different heights between the reflector and the radiating plate. It was found that the best levitation configuration is the H3-mode, which represents a good compromise between high levitation power and smooth pattern transition. The H3-mode also allows three translation lines in parallel.

9:00

1aPA2. Manipulating 5 nanometer diamond nanoparticles in user-defined patterns using bulk acoustic waves. Bart Raeymaekers (Dept. of Mech. Eng., Univ. of Utah, 50 S. Central Campus Dr., Salt Lake City, UT 84112), Cristian Pantea, and Dipen N. Sinha (Los Alamos Natl. Lab., Los Alamos, NM 87545)

We investigate manipulating 5 nm diamond nano-particles in a user-defined pattern on a substrate using the acoustic radiation force associated with a bulk acoustic standing wave. Both concentric and rectangular patterns are studied and the experimental results are compared with theoretical predictions. The effect of drag force acting on a nanoparticle is evaluated and limits for particle speed and particle size that can be moved by acoustic radiation force are determined. The importance of Brownian motion when manipulating nanoparticles is discussed. We found good agreement between our experimental results and existing theoretical models and demonstrate that nano-sized particles can be manipulated effectively by means of bulk wave acoustic radiation force.

9:15

1aPA3. Metamaterial synthesis using the acoustic radiation force and characterization with x-ray microcomputed tomography. F.G. Mitri and D.N. Sinha (Los Alamos Natl. Lab., Los Alamos, NM 87545)

In this research, we demonstrate the manipulation of clusters of ~5 nm-diameter carbon nano-particles using the acoustic radiation force at 1 and 2 MHz. 1-D and 2-D patterns are synthesized and experimental results show the successful synthesis of a nano-composite 3D metamaterial structure. Furthermore, x-ray micro-computed tomography (CT) is used as a tool to charaterize each metamaterial. Though not investigated here, the aim is to create finite element models based on the x-ray micro-CTs to study the multiple functionalities of the acoustically-assembled metamaterials. [Work supported by LANL-LDRD X9N9.]

9:30

1aPA4. Arbitrary acoustic scattering of a high-order Bessel vortex beam by a sphere. F. G. Mitri (Los Alamos Natl. Lab., Sensors and Electrochemical Devices, MS D429, Los Alamos, NM 87545)

The arbitrary acoustic scattering of a high-order Bessel vortex beam by a sphere is investigated. It is shown here that shifting the sphere off of the axis of wave propagation induces a dependence of the scattering on the azimuthal angle. Theoretical expressions for the incident and scattered fields from a rigid immovable sphere are derived. The near- and far-field acoustic scattering fields are expressed using partial wave series involving the *m*th-order spherical harmonics, the scattering coefficients of the sphere, the half-conical angle of the wave number components of the beam, its order, and the beam-shape coefficients. The scattering coefficients of the sphere and the three-dimensional (3-D) scattering directivity plots in the near- and far-field regions are evaluated using a numerical integration procedure. The calculations indicate that the scattering directivity patterns near the sphere and in the far-field are strongly dependent on the position of the sphere facing the incident high-order Bessel vortex beam. In addition to providing physical insight into the off-axial scattering of acoustic Bessel vortex beams, this investigation would potentially assist in the development of the transverse acoustic radiation force and could provide a useful test of finite element codes for the evaluation of the scattering.

9:45

1aPA5. Radiation torque on solid spheres and drops centered on an acoustic helicoidal Bessel beam. Likun Zhang and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA 99164-2814)

Somewhat analogous to circularly polarized electromagnetic waves carrying axial angular momentum and generating a corresponding torque, a helicoidal acoustic (vortex) beam also carries axial angular momentum and absorption of such a beam should also produce an axial radiation torque [B. T. Hefner and P. L. Marston, J. Acoust Soc. Am. **106**, 3313–3316 (1999)]. Here the acoustic radiation torque on solid spheres and spherical liquid drops centered on acoustic helicoidal Bessel beams is analyzed. Using a relation between the scattering and the partial wave coefficients for a sphere in a helicoidal Bessel beam, the torque is predicted to be proportional to the ratio of the absorbed power to the acoustic frequency. Calculations suggest that beams with a low topological charge tend to be more efficient for generating torques on solid spheres for all frequencies and on liquid drops in the low frequency regime. Balanced by drag torque, a steady rotation is generated. Calculations of the steady-state angular velocity suggest that the rotation in this type of traveling wave beam could be significant using appropriate frequencies. [Work supported in part by NASA.]

1a MON. AM

10:00—10:15 Break

10:15

1aPA6. Modal exchange between an elastic mode and an acoustic mode of a fluid-filled spherical shell resonator at an avoided crossing. Joel B. Lonzaga, Joel Mobley (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jblonzag@olemiss.edu), and D. Felipe Gaitan (Impulse Devices, Inc., Grass Valley, CA 95945)

High pressure liquid-filled spherical resonators are promising devices for increasing the energy density of a cavitating liquid. Since increasing the static pressure in the liquid also increases the sound speed, it is important to investigate the dependence of the resonances on the sound speed. In this paper, we report on an avoided crossing or repulsion between an elastic mode and an acoustic mode of a fluid-filled spherical shell resonator as the sound speed is varied. Such an avoided crossing is attributed to the fluid-shell interaction and leads to a modal exchange between the acoustic and the elastic modes. Furthermore, the eigenfrequency curves of both the acoustic and elastic modes are discontinuous in the frequency-sound speed phase space in the neighborhood of the avoided crossing. Additional analysis reveals that the elastic mode also forms an avoided crossing with other acoustic mode orders and that this result can be extended to other pairs of modes with different symmetry. The acoustic pressure field in the fluid and the stress field in the shell are shown to take extreme values at this region. [Work supported and funded by Impulse Devices, Inc. ACPT Contract No. W9113M-07-C-0178.]

10:30

1aPA7. Quasispherical acoustic resonators: The shell-gas interaction. James B. Mehl (P.O. Box 307, Orcas, WA 98280, jmehl@rockisland.com) and Michael R. Moldover (Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8360)

Spherical acoustic cavity resonators are useful tools for high-precision measurements of the speed of sound in gases. Conventionally, the gas-acoustic resonances are corrected for coupling to the shell using analytic expressions valid for isotropic, unsupported, shells of uniform thickness. Practical resonators have nearly spherical inner surfaces but exterior surfaces with non-spherical features and support structures. Finite-element methods have been used to calculate the response of realistic resonator shells that are coupled to support structures. Improved corrections of the gas-mode eigenfrequencies can be obtained in a form that is similar to the usual analytic expression, except that the shell response function must be calculated numerically. It is also shown that the coupling of gas modes through the shell is normally a weak effect proportional to the square of the static pressure and is totally negligible except in very unusual circumstances. The evolution of gas-shell coupling is followed when the speed of sound in the gas is varied so that a radial gas mode eigenfrequency f_g approaches a shell mode eigenfrequency f_s. The gas mode becomes unobservable as f_g nears f_s due to an avoided-crossing phenomenon similar to simple coupled oscillators.

10:45

1aPA8. Measurement of the penetration depth for an ultrasonic diffraction grating in contact with a slurry. Margaret S. Greenwood (Appl. Phys., Pacific Northwest Natl. Lab., P.O. Box 999, Richland, WA 99352)

The initial measurements of the penetration depth were presented at the Portland ASA meeting [Proceedings of Meetings in Acoustics, Vol. 6 P.045001]. This report describes additional measurements using several gratings with transducer frequencies of 1, 3.5, and 7 MHz. The grating unit (with the grating surface in the horizontal plane) is placed atop a vessel containing a slurry of polystyrene spheres in water. The data acquisition system was set up to take up to 20 measurements of the FFT amplitude per second. When the magnetic stirrer is turned off, the increasing FFT amplitude shows the transition from a slurry to water as the particles fall to the bottom of the vessel. As the slurry descends it forms a layer which is videotaped to yield the descent velocity. The penetration depth is obtained from the transition time and the descent velocity. The penetration depth varies with frequency and is less than about 4 mm. These measurements were not taken at the critical frequency for which the penetration depth is expected to be very large.

11:00

1aPA9. Attenuation mechanism for slurry in contact with ultrasonic diffraction grating. Margaret S. Greenwood (Appl. Phys., Pacific Northwest Natl. Lab., P.O. Box 999, Richland, WA 99352)

Attenuation measurements for ultrasound penetrating a slurry in contact with an ultrasonic diffraction grating were presented at the Paris ASA meeting (Proceedings of Meetings on Acoustics, **4**, 045017). Also, the viscous-inertial model was suggested as the mechanism, in which the attenuation is due to the dissipation in the thin boundary layer of liquid surrounding the particle. Further, it was proposed that, in order for the dissipation to occur, the circumference of the particle must be equal to or greater than 1 ultrasonic wavelength. Measurements will be reported for the following slurries in contact with a grating using a 1 MHz transducer: (1) 275-μm diameter polystyrene spheres, (2) 671-μm polystyrene spheres, and (3) 1588-μm acrylic spheres using a diffraction grating with a 1 MHz transducer. Attenuation was observed for slurry cases 2 and 3. However, no attenuation could be observed for the slurry of 275-μm diameter polystyrene spheres in water even at a volume fraction of 0.4; the signal was the same as that for water. Since the circumference for the 275-μm diameter particles was less than wavelength in water, these data provide confirmation of the proposed mechanism. Also, such measurements provide a method of exploring the viscous-inertial model.

11:15

1aPA10. Numerical simulation of acoustic wave interaction with inhomogeneous elastic bodies containing homogeneous inclusions. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Res., 739 Calle Sequoia, Thousand Oaks, CA 91360)

An extended version of the previously developed volumetric integral equation solver is presented, which is based on solving a coupled system of volumetric and surface equations describing, respectively, the inhomogeneous elastic object and the embedded piecewise homogeneous inclusions. The considered method is capable of accurate large-scale numerical simulations involving anatomically realistic models of a human head characterized by complex geometrical details and large density contrasts. The main application of the approach consists in an accurate analysis of interaction of acoustic waves with human hearing system; in this case, precise modeling of the highly intricate structure of middle and inner ears is essential for reliable numerical simulations capable of discerning between different mechanisms of energy transfer to the human ear. The method allows us to treat the inner ear region as a piecewise homogeneous inclusion described by surface integral equations with displacement and traction fields as unknowns. The results of representative calculations will be presented, which were carried out with a realistic detailed geometry model of the middle and inner ear including essential geometry elements needed in simulating energy transfer processes. [This work is supported by the AFOSR.]

MONDAY MORNING, 23 MAY 2011 GRAND BALLROOM C, 8:15 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: Cochlear Models and Psychoacoustics of Speech

Douglas S. Brungart, Chair
Walter Reed Army Medical Center, Washington, DC 20307

Contributed Papers

8:15

1aPP1. Four-chamber cochlea box model: Establishing acoustic comfort, illustrating injury and toward therapy. Luis Ma. T. Bo-ot, Henry V. Lee, Jr., Henry J. Ramos (Natl. Inst. of Phys., Univ. of the Philippines, Diliman, Quezon City 1101, Philippines), and Che-Ming Chiang (Natl. Cheng Kung Univ., Tainan 701, Taiwan)

The unrolled cochlea is modeled using the finite-element software ANSYS with four inner chambers representing the Scala Vestibuli contiguous with the Scala Tympani thru a rounded helicotrema, the Scala Media, the inner and the outer hair cells. The tectorial membrane is represented as a plate in contact with the hair cells. An improvement from previously presented results is the inclusion of a tapered helicotrema. Various geometries are compared, i.e. with straight sides and with tapered sides, and differences in the frequency response of the models are seen. Applying real values for material properties and the human hearing range and using characteristic frequency at certain nodes inside the Scala Media, the tapered model is calibrated to establish the reference comfort sound level. Hearing injury is regarded by subjecting nodes to increased sound pressure levels until the frequency response disappears. This is done at the same time monitoring the change in electrical potential in the inner and outer hair cell regions. The potential change between the normal and the injured conditions is inputted to the Gibbs energy equation for the ATP-ADPase glycolysis to identify a possible route to remedy.

8:30

1aPP2. Auditory filter with minimum-uncertainty product between frequency and scale. Douglas H. Keefe (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, douglas.keefe@boystown.org)

A scale representation of a signal describes its compression or dilation as a function of time or frequency. Because the frequency scaling properties of cochlear mechanics vary slowly at frequencies above 1–1.5 kHz, a frequency-scale representation is of particular interest for auditory filters. An auditory filter is constructed based on the analytic signal with the minimum uncertainty product between frequency and scale. Its spectral and temporal properties are analyzed. The complex spectrum of the filter is completely specified as a function of the frequency relative to the center frequency of the filter, the filter $Q,$ and the dimensionless group delay normalized by the number of periods at center frequency. In approximate agreement with basilar-membrane mechanics and human psychophysics, the frequency of peak spectral amplitude decreases by a half octave as tuning broadens, the spectral amplitude gain increases with sharper tuning, and the dimensionless group delay of the filter is independent of frequency. The peak envelope of the filter impulse response occurs at a time inversely proportional to the peak spectral frequency. Results are obtained on auditory phase perception for tones embedded in chirp maskers. A nonlinear filterbank composed of frequency-scale filters has potential applications in auditory research and audio engineering.

8:45

1aPP3. Synchrony-capture filterbank: A novel cochlear signal processing model. Ramdas Kumaresan, Vijay Peddinti (Kelley Hall, 4 East Alumni Ave., Univ. of Rhode Island, Kingston, RI 02881), and Peter Cariani (Harvard Med. School, Boston, MA)

Examination of the representation of low harmonics of complex sounds in the auditory nerve shows a striking feature known as "synchrony capture." Fibers of an entire cochlear region are driven almost exclusively by one local, dominant harmonic component [Delgutte and Kiang, 1984]. Sharp boundaries characteristic of such synchrony capture are also seen between the different CF regions driven by different dominant, formant-region harmonics for multiformant vowels. Based on this observation, we propose a model for peripheral processing, which is not just a filter bank but behaves more like signal adaptive receivers. We call this model synchrony capture filterbank (SCFB). SCFB consists of a traditional gammatone filterbank, the individual filters of which are then cascaded with a bandpass filter (BPF) triplet. The BPF triplet is an adaptive tone follower and consists of three overlapping bandpass filters whose center frequencies can be changed using feedback. The amplitudes at the output of the the bandpass filters are feedback to tune the BPF triplet such that it centers itself on top of the dominant tone in the input signal. The SCFB exhibits the synchrony capture behavior that is observed in real auditory nerve fibers.

9:00

1aPP4. Subjective evaluation of across-frequency delays. Magdalena Wojtczak, Jordan A. Beim, and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 74 East River Rd., Minneapolis, MN 55455)

Basilar-membrane filtering introduces frequency-dependent cochlear delays. Cochlear models predict that the delay between the peaks in responses to 100-Hz versus 10-kHz tones may be as long as 10 ms. This study investigated the role of across-frequency delays for asynchrony perception. A method of constant stimuli was used to evaluate the perception of delays between two tones, a 250-Hz tone paired with a higher-frequency tone at 1, 2, 4, or 6 kHz for a range of delays between the high- and low-frequency tones. Bandpass noise located between the two tones was used to avoid within-channel cues. Listeners judged whether the tone pairs appeared synchronous or asynchronous. For each pair, the proportion of "synchronous" responses was plotted as a function of the delay between the tones. Since cochlear-filter bandwidths depend on level, the perception of across-frequency delays was evaluated for two levels of the tones, 20 dB SL and 85 dB SPL. Overall, the data support the hypothesis of a higher-level mechanism compensating for cochlear delays, producing veridical perception. However, the results do suggest a perceptual asymmetry, with synchronous judgments occurring more often for leading high-frequency tones than vice versa. [Work supported by NIH Grant No. R01DC006804.]

9:15

1aPP5. Cochlea-scaled entropy predicts speech intelligibility when intensity is scaled as linear amplitude or as loudness, but not when scaled logarithmically. Christian E. Stilp and Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706, cestilp@wisc.edu)

Stilp and Kluender [Proc. Natl. Acad. Sci. **107**, 12387–12392 (2010)] reported sentence intelligibility to be well-predicted by measures of sensory change over time [cochlea-scaled spectral entropy (CSE)] and not consonants, vowels, or duration of signal replaced. As greater amounts of CSE were replaced by noise, intelligibility worsened. They calculated CSE as Euclidean distances between spectra defined by linear amplitude as a function of equal rectangular bandwidth (ERB). The present experiments compare measures of speech intelligibility as a function of CSE when amplitude is measured on a linear scale as before and when log-scaled (dB) or loudness-scaled (sone). Listeners typed all words they understood after hearing sentences with 80- and 112-ms segments replaced by speech-shaped noise. Log-scaled (dB) CSE was a poor predictor of performance. Both linear-scaled and loudness-scaled measures of CSE predict intelligibility very well, replicating Stilp and Kluender. Linear and loudness scales will be discussed in the context of upward spread of masking and two-tone suppression as they relate to speech signals. These perceptual findings suggest that conventional use of spectrograms (linear frequency $\times$ dB intensity) could systematically misinform researchers concerning properties of speech signals that are most useful to listeners. [Work supported by NIDCD.]

9:30

1aPP6. Cortical neural coding of speech in simple and complex auditory scenes. Nai Ding and Jonathan Z. Simon (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD 20740)

The neural representation of continuous speech in human auditory cortex was obtained noninvasively via magnetoencephalography. The neural response was recorded from human subjects listening to a spoken narrative, either in clean or in the presence of interfering speech. The cortical neural response to clean speech is demonstrated to precisely track the slow temporal modulations ($<$10 Hz) of speech in a broad spectral region between 400 Hz and 2 kHz. The neural code is sufficiently faithful to decode acoustic features of speech. To examine the robustness of, and the role of attention in, this neural code, another spoken narrative was presented simultaneously, either to a different ear (dichotically) or to the same ear (diotically), instructing the subjects to focus on only one of the two speech signals. The cortical representation of the attended speech is found to be substantially stronger than that of the unattended speech. This attentional effect is significant during the subjects' first exposure to the spoken narratives. These results demonstrate that auditory cortex precisely represents the slow temporal modulations of speech and maintains separate neural representations for concurrent speech signals that can be individually and strongly modulated by attention. [Work supported by the NIDCD Grant No. R01-DC-008342.]

9:45

1aPP7. Relationships linking age, selective attention, and frequency following in the brainstem. Dorea R. Ruggles and Barbara G. Shinn-Cunningham (Dept. of Biomedical Eng., Boston Univ., 44 Cummington St., Boston, MA 02215, ruggles@bu.edu)

Several studies have demonstrated that increasing age leads to a decrease in the ability of the brainstem to phase-lock to periodic stimuli. This frequency following response has additionally been related to several other metrics, including musical training, experience with a tonal language, and ability to understand speech in noise. The current study explored a potential relationship between brainstem responses and large, observed individual differences in the ability of normal-hearing listeners to deploy selective spatial auditory attention. Previously, we reported that the ability to selectively attend varies enormously (from near chance to 90% correct) in normal-hearing listeners, but is uncorrelated with age (ranging from 20–55 years). Here, we recruited a subset of the original listeners and measured the brainstem's frequency-following response (FFR). Although selective-attention performance was correlated with the strength of the FFR, there was an interaction between the FFR strength and age. For younger listeners, the FFR closely predicted selective attention ability, but the FFR was weak in all older listeners, even those who performed well behaviorally. These results show that the FFR strength decreases with age without necessarily leading to degraded perceptual abilities; however, young listeners with weak FFR cannot effectively deploy selective spatial attention.

10:00—10:15 Break

10:15

1aPP8. Across-ear integration of complex speech stimuli. Douglas S. Brungart (Walter Reed Army Medical Ctr., Washington, DC), Nandini Iyer, and Brian D Simpson (Air Force Res. Lab., WPAFB, OH)

In listening environments with spatially separated sources, listeners can often obtain a substantial improvement in performance by implementing a "better ear" listening strategy where they selectively attend to the ear with the more advantageous signal to noise ratio (SNR). However, relatively little is known about the ability of listeners to implement the better ear strategy on a band-by-band basis by focusing attention on a different ear within each frequency band. This experiment examined how efficiently listeners are able to implement a band-by-band better-ear listening strategy by dividing a modified rhyme test speech stimulus into 20 frequency bands of approximately equal intelligibility and presenting a subset of those bands either monaurally, with all the bands in one ear, or dichotically, with half the bands assigned to one ear and half assigned to the other ear. The results showed that the listeners were able to integrate information across the two ears on a band-by-band basis, and that this ability was robust in a wide range of within-band and across-band masking conditions. However, there was a slight penalty for dichotic presentation, which was estimated to be roughly equivalent to a 1.2 dB decrease in the SNR of a broadband speech signal in noise.

10:30

1aPP9. The effect of disrupting the continuity of target sentences on the perceptual segregation of competing voices. Nandini Iyer (Air Force Res. Lab., WPAFB, OH 45433), Douglas S. Brungart (Walter Reed Army Medical Ctr.), and Brian D. Simpson (Air Force Res. Lab.)

Relative differences in fundamental frequency and prosodic features have been studied as possible cues that enable listeners to segregate a target talker in multitalker listening tasks. However, little is known about the effects of disrupting the prosody characteristics of a talker within a single phrase. The current experiment measured color-number keyword identification scores with the coordinate response measure (CRM) sentences under two conditions: a "normal prosody" condition, where the target and masker phrases were spoken contiguously by a single talker, and a "disrupted prosody" condition, where the target and masking phrases were generated by concatenating together individual words that were originally spoken in different sentences. The results show that the elimination of normal prosodic cues had the greatest impact on performance when the target sentence was masked by a single CRM masking phrase that was presented at the same level as the target talker. The elimination of prosodic information had only a very slight impact on performance when the target talker was masked by more than one simultaneous talker, suggesting that the use of prosodic cues for segregation may be impaired when more than one interfering talker is present in the mixture.

10:45

1aPP10. Cues above 4 kilohertz can improve spatially separated speech recognition. Sunil Puria, Suzanne Carr Levy, Daniel J. Freed, and Michael Nilsson (Earlens Corp., 200 Chesapeake Dr., Redwood City, CA 94063, spuria@earlenscorp.com)

There are multiple acoustic cues above 4 kHz that may be used to improve speech recognition in noisy environments. This study tests this hypothesis with spatially separated speech maskers. The reception threshold of speech (RTS) was measured in two conditions: (1) asymmetric condition with target speech at $-45°$ and maskers at $+45°$, and (2) diffuse condition with speech at 0° and one masker at each of four quadrants around the listener ($\pm 45°$, $\pm 135°$). HINT sentences were rerecorded using a male talker and a 22.05 kHz bandwidth while different males were recorded speaking the television and rainbow as maskers. SRTs were measured with all materials low-pass filtered at 4, 6, 8, and 10 kHz for the asymmetric and diffuse conditions. 12 normal hearing subjects (on the way to 24) were tested using free-field loudspeakers in a sound booth. The mean RTSs for the asymmetric condition were -17.43, -19.07, -19.34, and -20.65 dB for 4, 6, 8, and 10 kHz, respectively. For the diffuse condition the mean RTSs were -7.77, -9.00, -9.88, and -10.05 dB for the same frequencies. The results suggest that high-frequency acoustic cues can enhance the ability to segregate spatially separated speech.

11:00

1aPP11. Room reverberation and constancy in sparse noise-vocoded speech. Anthony Watkins, Andrew Raimond, and Simon Makin (Dept. of Psych., Reading Univ., Reading RG6 6AL, United Kingdom)

Speech played several metres from the listener in a room is usually heard to have much the same phonetic content as it does when played nearby, although the different amounts of reverberation at these distances make the temporal envelopes of these signals very different. To study this "constancy" effect, listeners heard natural-speech messages and noise-excited vocoder versions of them. The vocoder used eight auditory-filter shaped noise-bands, each with the temporal envelope arising in that filter when the speech message is played. The center frequencies of the equally log-spaced bands ranged from 250 Hz to 4.24 kHz. An 11-step "sir-to-stir" continuum was formed by amplitude modulation and each step was played in the context, "next you'll get … to click on." Listeners identified test words appropriately, even in the eight-band conditions where the speech had a robotic quality. Constancy was assessed by comparing the influence of reverberation on the test word across conditions where the context had either the same level of reverberation (i.e., from the same, far distance), or where it had a much lower level (i.e., from nearby). Constancy effects were obtained with both the natural- and the eight-band speech, with the higher-frequency bands having more importance.

11:15

1aPP12. Predicting speech intelligibility based on the envelope power signal-to-noise ratio after modulation-frequency selective processing. Torsten Dau and Søren Jørgensen (Dept. of Elec. Eng., Tech. Univ. of Denmark, Lyngby, Denmark)

A model for predicting the intelligibility of processed noisy speech is proposed. The model represents a speech-based version of the envelope power spectrum model [Ewert and Dau, 2000] originally developed to account for modulation detection and masking data. The model estimates the ratio of speech-to-noise envelope power, SNR_{env}, at the output of a modulation filterbank and relates this metric to speech intelligibility using the concept of an ideal observer. Model predictions were compared to literature data, obtained with speech mixed with stationary speech-shaped noise. Furthermore, the model was tested in conditions with noisy speech subjected to reverberation and spectral subtraction. Consistent with new experimental data, the model predicted an increase in SRT as a function of the reverberation time as well as in conditions of spectral subtraction, the latter in contrast to the STI. An analysis of the model's internal representation of the stimuli processed by spectral subtraction revealed that the decrease of the predicted speech intelligibility was caused by an elevation of the noise envelope power, which exceeded that of the speech envelope power. The results strongly suggest that the signal-to-noise ratio at the output of modulation frequency selective processing represents a critical measure of speech intelligibility.

11:30

1aPP13. Temporal integration of interrupted speech: Effects of time compression and expansion on the intelligibility of words and sentences. Valeriy Shafiro, Stanley Sheft, and Robert Risley (Dept. Comm. Disord. Sci., Rush Univ. Medical Ctr., 1015 AAC, 600 S. Paulina Str., Chicago, IL 60612, valeriy_shafiro@rush.edu)

Temporal integration of interrupted speech was investigated by contrasting the performance on gated to time-compressed/expanded words and sentences. In the control condition, HINT sentences and CNC words were gated at 0.5, 1, 2, 4, 8, or 16 Hz using a 50% duty cycle. In the two experimental conditions, stimuli previously gated at each rate were either time-compressed by concatenating the consecutive speech segments or time-expanded by doubling the silent intervals between consecutive speech segments. Across rates, these manipulations thus varied both the size of intact speech intervals and the duration of silence between the intervals. No differences were observed in the rate-intelligibility functions of gated versus time-expanded words and sentences with the overlapping functions monotonically rising with rate. Though intelligibility was similar to the gated control condition at the lower and higher rates, the time-compressed rate-intelligibility function for sentences was nonmonotonic with a minimum at 2-Hz. In contrast, the time-compressed function for words remained similar to the corresponding gated and time-expanded functions. These findings indicate that the temporal integration of interrupted speech takes places on different perceptual time scales for words and sentences and is affected by the size of speech segments being integrated. [Work supported by NIH /NIDCD.]

11:45

1aPP14. Discrimination of acoustic differences improves with greater dissimilarity to experienced covariance. Christian E. Stilp and Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706, cestilp@wisc.edu)

Stilp *et al.* [Proc. Natl. Acad. Sci. (in press)] demonstrated efficient coding of correlations between complex acoustic attributes (attack/decay and spectral shape) in novel sounds. Discrimination of differences between sounds that violate correlations (orthogonal) is initially inferior and later comparable to that for sound pairs that respect the correlation (consistent). Subsequent findings [Stilp *et al.*, J. Acoust. Soc. Am. **128**, 2455–2456 (2010)] suggest that this effect may depend more on acoustic similarity between consistent and orthogonal sounds than upon probability of orthogonal test trials. Across the present experiments, listeners discriminated (AXB) the same 15 consistent sound pairs and a single orthogonal sound pair of varying similarity to the consistent sounds (distance from the correlation vector in the stimulus matrix). Orthogonal discrimination systematically improves with increasing dissimilarity. With extreme dissimilarity, orthogonal discrimination is significantly better than consistent discrimination across all testing blocks. Results cannot be explained by weighting of individual stimulus dimensions. A principal components analysis network model for extracting covariance structure that successfully predicted listener performance in previous tasks predicts some but not all of these findings. Implications for perception of complex sounds including speech will be discussed. [Work supported by NIDCD.]

MONDAY MORNING, 23 MAY 2011 DIAMOND, 8:00 TO 11:40 A.M.

Session 1aSA

Structural Acoustics and Vibration, Noise, and Engineering Acoustics: Fluid-Structure Interaction, Computational and Experimental

Dean E. Capone, Cochair
Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Robert L. Campbell, Cochair
Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Invited Papers

8:00

1aSA1. Fluid-structure interaction and inverse design simulations for highly flexible turbomachinery. Robert L. Campbell (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

Highly flexible turbomachinery offers substantial advantages for biomedical implantation but can suffer from performance losses due to blade deformations during operation. The objective of this work is to develop a method to define an impeller shape that deforms into the design shape during operation, thereby making feasible a collapsible impeller for medical implantation without incurring performance losses due to blade flexibility. A fluid-structure interaction (FSI) solver is developed and validated for quasi-steady operation, capable of modeling the time-dependent deformation inherent to the impeller polymeric material. The solver is validated using experimental data for a modified NACA 66 fin at various angles of attack. Inverse design simulations are used to define blade geometry that deforms into the design shape after being subjected to quasi-steady flow at prescribed conditions for a specified amount of time. The validated FSI solver is used to confirm performance of the inverse design shape. Evaluation of the FSI solver convergence shows the fluid and structure are nearly fully converged after only a few sub-iterations for these quasi-steady simulations. Performance studies show that the flow solver consumes most of the processing time, with the mesh motion and structure solutions requiring a much smaller fraction of the total processing time.

8:20

1aSA2. Scattering of acoustic quasi-Gaussian beams and Bessel beams by fluid-loaded spheres: Exact solutions and physical interpretation. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Exact solutions are available for the scattering of sound by spheres placed on the axis of certain types of acoustic beams. In the case of an acoustic Bessel beam the scattering for various elastic spheres is greatly modified relative to the standard case of plane-wave illumination [P. L. Marston, J. Acoust. Soc. Am. **122**, 247–252 (2007)]. Symmetry and geometric considerations explain why selected sphere modes can be suppressed in a way that depends on the Bessel beam parameters. The quasi-Gaussian case is treated by an appropriate superposition of zero-order Bessel beams displaying the focal properties of a Gaussian beam [P. L. Marston, submitted]. This superposition, unlike the case of a paraxial Gaussian beam, is an exact solution of the Helmholtz equation. The sphere is centered on the focal point of the beam and the scattering follows by superposition from the Bessel beam case. While not prone to suppress specific modes of the sphere, the quasi-Gaussian case can also modify the scattering for a wide range of beam parameters and frequencies. Mode amplitudes are modified by a function depending on a ratio of incomplete gamma functions and the size of the focus of the beam. [Work supported by ONR.]

8:40

1aSA3. Experimental and computational studies of vortex shedding and aerodynamic sound for flow over a square cylinder near a rigid wall. Yatao Hu, Jun Chen, Kai Ming Li (Dept. of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031), Yatao Hu, and Keqi Wu (Huazhong Univ. of Sci. and Technol., 1037 Luoyu Rd., Wuhan 430074, China)

Flow around a bluff body near a solid wall, such as a circular or rectangular cylinder placed in a uniform flow near a solid boundary, is of fundamental importance. The reduction in the aerodynamic sound is a very important issue, for example, flow induced noise in tall buildings, bridges, high-speed trains, windmills, and computer cooling fans. This type of problem has been addressed extensively both experimentally and theoretically in the past. The flow past a long rectangular cylinder is usually associated with vortex shedding, which can be affected strongly by the presence of a nearby wall. Flow past a solid body induces vortices and the vortex acceleration will cause sound. In the present study, the aerodynamic sound generated by flow past a rectangular cylinder of various configurations was investigated experimentally in a low-speed quiet wind tunnel. A numerical study using large scale eddy Simulation technique has also been conducted to compute the sound generation due to vortex shedding. Experimental results are compared with the numerical predictions. [Work sponsored by the China Scholarship Council.]

8:55

1aSA4. Sound generated by an elastic wing actuated at its leading edge. Avshalom Manela (Faculty of Aerosp. Eng., Technion, Haifa 32000, Israel)

The sound generated by a thin elastic plate, subject to uniform low-Mach flow and to leading-edge pitching and heaving motions, is studied. The linearized plate motion is analyzed under conditions where the unforced plate (in the absence of leading-edge actuation) is stationary. When the frequency of applied forcing coincides with an eigenfrequency ω_{res} of the unforced plate, a resonance motion is excited and the plate oscillates at the corresponding eigenmode. The sources of sound in the problem include the plate velocity and fluid vorticity. Acoustic radiation of dipole type is calculated and discussed in the limit of an acoustically compact plate. It is found that plate elasticity has two opposite effects on sound radiation: close to ω_{res}, elasticity results in the generation of high sound pressure levels; however, far from ω_{res}, elasticity tends to reduce the amplitude of plate deflection (compared to that of a rigid plate), leading to reduced sound levels. It is also shown that the release of trailing-edge vortices is the main source of sound, dominating the radiation from the direct plate motion. The present theory is suggested as a simple tool for analyzing insect-flight sound and predicting the acoustic signature of flapping micro-air-vehicles.

9:10

1aSA5. SALINAS: A massively parallel finite element code for structural dynamics and acoustic analysis. Jerry W. Rouse, Timothy F. Walsh, and Garth M. Reese (Sandia Natl. Labs., P. O. Box 5800, 87185-0346, jwrouse@sandia.gov)

This talk shall present an overview of SALINAS, a massively parallel finite element code for structural dynamics and acoustics analysis that is being developed at Sandia National Laboratories. SALINAS allows for prediction of both the time and frequency domain responses of complex structural, acoustic, and fully coupled structural acoustic systems having millions of degrees of freedom. An overview of SALINAS capabilities shall be presented including development history, solver and element types, quadratic eigenanalysis and frequency response, direct frequency response, nonlinear acoustics, implicit transient dynamic analysis, and infinite elements with focus given to structural acoustics capabilities. The application of SALINAS to structural acoustics problems shall also be presented as well as future directions of research for the development of the code.

9:25

1aSA6. Effect of internal pressure on the vibrational response of a fluid-filled spherical shell: Experiment. Andrew A. Piacsek and Robert P. Taylor (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, piacsek@cwu.edu)

The resonance frequencies of a spherical aluminum shell (radius 3.0 in., thickness 1/8 in.) filled with water have been measured for several different values of static water pressure. It is found that a pressure increase of 100 psi causes resonance frequencies associated with axisymmetric bending modes to shift higher by about 0.15%, consistent with predictions of elastic shell theory [DiGiovanni and Dugundji, Air Force Office of Scientific Research Report No. 65-0640 (1965). The shell is suspended by elastic cords attached to an inlet valve and is excited acoustically with a swept sine wave; the vibrational response is measured with small accelerometers mounted on the shell surface. Techniques for identifying frequency shifts associated with very small pressure changes (less than 1 psi) will be discussed. The effect reported here may have an application in the development of noninvasive methods for measuring intracranial pressure changes. [Work supported by National Science Foundation STEM program and Central Washington University.]

9:40

1aSA7. Effect of internal pressure on the vibrational response of fluid-filled shells: Finite element model. Sami Abdul-Wahid and Andrew A. Piacsek (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, sipho.1989@gmail.com)

A finite-element model of the vibrational response of fluid-filled shells with arbitrary shape and composition has been developed using the COMSOL multi-physics modeling package. The user can specify the properties of the fluid inside the shell including the static pressure. The shell is surrounded by air, which is enclosed by a perfectly matched layer boundary, and an acoustic source is positioned just outside the shell. The frequency response of the shell due to a swept sine acoustic excitation can be recorded at multiple locations. Model results for a spherical aluminum shell filled with water at different static pressures are compared with experiment. Results are also shown for a shell with geometry and material properties similar to a human skull. The goal is to apply this model to predict vibrational response due to changes in intracranial pressure. [Work supported by Central Washington University Science Honors Program.]

9:55—10:10 Break

10:10

1aSA8. Coupled structural-acoustic computations for interior problems using a modal solution for the structural vibrations and a direct solution for the acoustic pressure field. John B. Fahnline (Garfield Thomas Water Tunnel, University Park, PA 16802)

Structural-acoustic computations involving enclosed volumes of fluid using finite elements are often performed by coupling *in vacuo* structural modes with acoustic modes derived assuming rigid-wall boundary conditions, sometimes referred to as Dowell's method. This approach works well for problems with "light" fluid coupling, but is very slow to converge for problems with "heavy" fluid coupling. Several authors have suggested methods to derive terms to add to the acoustic basis set to account for the residual contributions of truncated modes. Another possibility, which will be the focus of this paper, is to directly compute the acoustic field in terms of nodal pressure variables rather than as a modal summation. Despite the relative simplicity of this approach, it does not appear to have explored previously in the literature. As will be shown, the coupled structural-acoustic equation system closely resembles that for coupled finite element/boundary element problems because both are written in terms of the structural variables only. The main goal of the presentation will be to demonstrate that using a direct

solution for the acoustic field alleviates the slow convergence difficulties encountered with Dowell's method for coupled structural-acoustic analyzes.

10:25

1aSA9. On the use of a variational boundary element/finite element approach to predict the vibroacoustic response of structures. Franck Sgard, Kamel Amichi (Service de la recherche, IRSST, 505 Boulevard de Maisonneuve O, Montreal, PQ H3A3C2, Canada, frasga@irsst.qc.ca), Noureddine Atalla (Univ. of Sherbrooke, Sherbrooke, PQ J1K2R1, Canada), and Hugues Nlisse (IRSST, Montreal, PQ H3A3C2, Canada)

This paper deals with the application of a variational boundary element /finite element approach to predict the vibroacoustic response of structures. First, the theoretical formulation is presented. Then two specific examples are considered and numerical results are presented. The first case concerns the calculation of the sound transmission through a curved orthotropic sandwich panel. The blocked pressure field acting on the structure is first calculated and then used as a loading in the computation of the forced response of the panel. The second case consists of the calculation of the sound radiation of an assembly of plates at different angles excited by uncorrelated point forces. This configuration is of special interest to evaluate the accuracy of analytical Leppington's radiation efficiency correction factors related to baffle angles to predict the acoustic power radiated by a box-shaped acoustic enclosure.

10:40

1aSA10. Analysis of results of numerical calculations of sound scattering by inhomogeneous elastic shells aiming to create a physical model of the scatterer. Mikhail B. Salin (Inst. of Appl. Phys., Russian Acad. of Sci., 46 Uljanov St., Nizhny Novgorod 603950, Russia, mikesalin@hydro.appl.sci-nnov.ru)

Determining vibration and acoustic characteristics of elastic bodies in water (especially elastic shells) is a challenging problem that becomes more complicated if non-homogeneities of various types are present or a finite length object is considered (~1–10 wavelengths). Scattering by such bodies is considered here. This paper describes methods of analysis of results of numerical calculations that allow one to find the scattering mechanisms that make the greatest contribution to the scattered directivity pattern. The method allows one to distinguish the following factors: scattering as if it were a rigid body and excitation of an eigenmode, whose form can also be roughly estimated. The scattering object is replaced virtually by a discrete receiver-transmitter array whose transmitted signal is defined as the product of the received signal and the scattering matrix. Elements of the scattering matrix are found in such way that calculated scattered field values in the far field are close to the real ones. The results of this research can help in such areas as designing structures with required scattering characteristics, solving inverse problems and in classification and remote sensing. [The author gratefully acknowledges the generous support of the U.S. Office of Naval Research, ONRG.]

10:55

1aSA11. An efficient back-scattering model for arbitrarily shaped objects. Edward Pees (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, edward.pees@navy.mil)

A method is presented for efficiently computing the propagating pressure field back-scattered from arbitrarily shaped, 3-D objects. This is accomplished by drawing upon a previously reported relationship between the boundary condition on a 2-D radiating aperture and the pressure propagating along an axis normal to the aperture, and the diffraction slice theorem, which relates the Fourier transform of an object function to its scattered pressure field. Together, these two results allow for the derivation of an integral formula that expresses the pressure field back-scattered from an object as a 1-D Fourier transform of its scattering amplitude. Use of this formula is demonstrated for computing the back-scattered pressure field from a uniform sphere in the first Born approximation (weak scatterer) and with a Kirchhoff boundary condition (rigid scatterer); the results of which are compared to the corresponding rigorous partial wave expansions.

11:10

1aSA12. Low-wavenumber turbulent boundary layer wall-pressure measurements from vibration data on smooth and rough cylinders in pipe flow. Neal D. Evans (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, nue110@psu.edu), Dean E. Capone, and William K. Bonness (Penn State Univ.)

The vibration response of a thin cylindrical shell excited by a fully-developed turbulent boundary layer is measured and used to extract the fluctuating pressure levels generated by the boundary layer. Parameters used to extract the turbulent boundary layer pressure levels are determined via experimental modal analysis of the water-filled pipe and measured vibration levels from flow through the pipe at 6 m/s. Hydrostatic head from a large reservoir provides the low-noise source of steady flow for measuring the low-wavenumber fluctuating pressure levels. Measurements are reported for smooth, transitionally rough, and fully rough conditions and are compared to the turbulent boundary layer pressure models of Chase, Smol'yakov, and Howe.

11:25

1aSA13. Determination of pipe interior pressures using external accelerometers. Alexandria R. Salton, Dean E. Capone, and William K. Bonness (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ars328@psu.edu)

A non-invasive method utilizing a ring of accelerometers to measure pipe interior pressures is presented. The internal acoustic pressure of the fluid inside a cylindrical pipe is directly related to the vibrations at the surface due to the coupling of the fluid and structure. The $n=0$ breathing mode provides the basis for this relation and is extracted from operational data using a circumferential modal decomposition routine. The measurement of pipe interior pressures can be useful in determining the integrity of a piping system by detecting high pressures, which may lead to the fatigue or failure of the system. Within the field, a non-invasive method for measuring these pressures is more practical in comparison to the intrusive but more direct method of using hydrophones. Measurements using the ring of accelerometers are compared to hydrophone measurements to determine accuracy.

Session 1aUW

Underwater Acoustics and Acoustical Oceanography: Sediment Acoustics and Geological Processes I

Charles W. Holland, Cochair
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Allen Lowrie, Cochair
U.S. Naval Oceanographic Office, Balch Blvd., Stennis Space Center, MS 39522-5001

Chair's Introduction—8:25

Invited Papers

8:30

1aUW1. Low-frequency geoacoustic modeling in shallow water sediment environments. N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada)

The goal in geoacoustic modeling is to develop realistic geophysical models of the ocean bottom that can be used in numerical calculations of the acoustic field in the ocean. The ocean bottom is assumed to be a layered structure of different types of sediment material that have been deposited over geological times. However, in shallow water the bottom is generally much more complex. The sediment material is variable on different spatial scales horizontally and is inhomogeneous in depth below the sea floor. Despite this complexity in realistic bottom environments, there has been considerable success using the simplified approach of a layered, range-independent geology in low-frequency (20–500 Hz) applications with inversion techniques that provide estimates of geoacoustic model parameters and their uncertainties. This paper reviews some of the most effective inversion techniques and compares their performance in estimating realistic and effective geoacoustic profiles in applications with data from the recent Shallow Water '06 experiments on the New Jersey continental shelf. Conditions are discussed that limit the performance of present day inversion techniques. These include rough interfaces on and below the sea floor, consolidated material that supports shear wave propagation, and range variation of sub-bottom structure.

8:50

1aUW2. Stability of unconsolidated sediments and acoustic transmission. Allen Lowrie (U.S. Naval Oceanograph. Office, Code NP-53, Stennis Space Ctr./NASA, MS 39522-5001)

The degree of compaction in marine sediments has a direct effect on their velocity and density, components in the Navy's geoacoustic databases. Acoustic transmission through sediments necessitates grain-to-grain contact. Such a conceptual model coincides with geology of sediments settling onto seafloor to consolidation as generally continuous. However, such continuity may not be justified. Understanding of global/regional/local processes and impacts suggests that sediment grains are jostled, causing rearrangement/breaking /damaging/retarding contact between grains, reducing rigidity, and intergranular fluids "flushed" by passing ocean waves. Resultant of these processes is that sediments remain unconsolidated to unexpected depths. Improved knowledge of global/regional/local stress-fields reveals heretofore unappreciated complexity maintaining individual grains apart with retardation of contact. Meteor impacts and major earthquakes and volcanic eruptions form "instantaneous" stress-waves that can jostle sediments globally, forcing high-frequency inter-granular motions, rupturing, and interstitial fluid movements. Dynamic sedimentation episodes from glacial-lake releases and continental margin collapses provide lower frequency stress-waves causing lateral motions among sediments, re-forming grain arrangements. Wave migration over seafloor and de-watering fluctuates inter-granular fluids, maintaining interstitial spaces open. Tectonics form faults /breakages along with movements and fluids and material flowage, creating differing domains/units within sediments. These dynamic inputs within a complex-geographic-matrix assist in maintaining sediments unconsolidated.

9:10

1aUW3. Measurement and modeling of high frequency sound speed and attenuation in sandy sediments since the Sediment Acoustics Experiment 1999. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

During the Sediment Acoustics Experiment in 1999 (SAX99), measurements were made of the sound speed and attenuation in a sandy sediment, supplemented by detailed environmental characterization. While the dispersion was consistent with Biot theory, the attenuation at high frequencies had a linear frequency dependence that was consistent with models based on losses at grain contacts. These results led to the development of a number of competing models of sound propagation in sand sediments. Subsequent to SAX99, measurements of sound speed and attenuation have been made in several ocean sediments as well as in a number of laboratory sediments composed of either sand or glass beads. Many of these experiments have been accompanied by careful measurements of the sediment properties and, in some cases, these properties have been varied to assess their impact on sound propagation. An overview of these results will be given and the implications of these measurements for high frequency sediment acoustics modeling will be discussed. [Work supported by the Office of Naval Research.]

9:30

1aUW4. Geological processes affecting high-frequency acoustics. Kevin B. Briggs (Naval Res. Lab., Seafloor Sci. Branch, Stennis Space Ctr., MS 39529-5004, kbriggs@nrlssc.navy.mil)

Two seafloor properties that control high-frequency (greater than 10 kHz) acoustic scattering from and propagation into the sea floor are interface roughness and sediment inhomogeneities. Geological processes that affect the statistical distribution of these properties include sediment transport and sediment diagenesis. Sediment transport is a hydrodynamic process driven by waves and currents, which are results of storms and tides. Seafloor features that are significant for acoustic scattering and are created by sediment transport include sand waves, ripples, ripple-scour depressions, lag layers, flasers, and sand lenses. Scattering from these features can be predicted from the statistical characterization of interface roughness and sediment inhomogeneities. Sediment diagenesis is a physical, chemical, and sometimes biological process that affects the sediment bulk density through processes of dewatering, consolidation, precipitation, or cementation. Sediment diagenesis affects the gradient and fluctuations in bulk density and thus the speed and attenuation of sound. Scattering model inputs related to these features are derived from measurements of vertical fluctuations of sediment bulk density and sound speed. Examples of how sediment transport and diagenetic processes affect high-frequency acoustic bottom-interactions are presented.

9:50

1aUW5. The myth that sediment mean grain size provides a useful prediction of high-frequency acoustic bottom-interaction. Michael Richardson (Marine Geosciences Div., NRL, Stennis Space Ctr., MS 39529)

Mean grain size is the most common descriptor found in seafloor databases and is often the only sediment description available to acoustic modelers. Grain size (phi) is traditionally characterized by a base-2 logarithmic scale. This makes geological sense, as natural processes including weathering and transport tend to create lognormal distributions of particles. The mean describes the central tendency of phi-transformed size distributions and values of sorting, skewness, and kurtosis describe deviations from lognormal distributions. The presence of two or more grain size modes is often evident in grain size histograms. Packing in sand, compaction in mud, and sorting in sand/mud mixtures explain the high variability in mean grain size, density (porosity) relationships. Density provides a better predictor of reflection from and propagation within the seafloor. Although sediment transport is dependent on grain size, seafloor roughness characterized either by rms height or values of spectral parameters is poorly predicted by grain size. These relationships are overwhelmed by temporal variations due to biological and hydrodynamic processes. Prediction of roughness scattering requires concurrent measurements of seafloor roughness. Use of additional grain size statistics and sediment density and understanding the affects of hydrodynamic and biological process should provide better prediction of high-frequency bottom interaction.

10:10—10:30 Break

Contributed Papers

10:30

1aUW6. Sonic speed estimates for muddy sediments with plausible microbubble distributions. William M. Carey and Allan D. Pierce (Mech. Eng., College of Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Measurements of the sound speed characteristic of the high porosity Dodge Pond mud were found to have a sonic speed less than that observed by Wood and Weston [Acustica **14**, (1964)], a compressional speed 3% less than that of water. Other experiments performed on muddy sediments at frequencies greater than 1 kHz are consistent with the Dodge Pond observations when microbubbles are present. The presence of bubbles is known to be an important factor in decreasing the sound speed. A theoretical treatment of "muddy sediments," the card house theory [Pierce and Carey, POMA **5**, 7001, (2009)], estimated the slow sound speed and frequency dispersion proportional to mud porosity, $C_{\mathrm{mud}} \approx (0.91\text{–}0.97)C_{\mathrm{w}}$. The presence of microbubbles can lower the sound speed consistent with the Mallock–Wood equation when the bubble size distribution and mean bubble separation are less than the wavelength of the propagating wave. Since measurement of the bubble size distribution within the mud is difficult; theoretical limits on the size distribution in the complex card house structure can be useful in interpreting measurements on muddy sediments and provide a basis for acoustic distribution measurement. [Work supported by the ONR OA and the NSWC PCD.]

10:45

1aUW7. Estimation of the shear wave speed in mud based on a card-house theory with elastic platelets. Joseph O. Fayton (Rensselaer Poly. Inst., Troy, NY 12180), Allan D. Pierce, William M. Carey (Boston Univ., Boston, MA 02215), and William L. Siegmann (Rensselaer Poly. Inst., Troy, NY 12180)

Principal constituents of mud ocean sediments are small platelets of minerals such as smectite and kaolinite. Isomorphous substitution produces negative charges on platelets, and salt water ions rearrange to cause an effective quadrupole moment per unit area of platelet. A card-house structure results, where an edge of one platelet is bonded to the face of another and where each platelet is in a state of electrostatic equilibrium. Platelets are idealized as elastic plates, with a bending modulus proportional to Eh^3. The elastic modulus E can be estimated from chemical physics principles as a dimensionless constant times $\hbar^2/(m_e a_B^5)$, where a_B is the Bohr radius. When an elastic platelet is perturbed from its equilibrium position, the restoring forces near a contact point (joined edge) are exceptionally large so that an appropriate boundary condition is that the platelet is cantilevered. Shearing forces cause the platelet to bend like a cantilevered beam with distributed electrostatic restoring forces. Solution for the platelet deformation leads to estimates of the shear modulus and shear speed, which compare favorably with existing data. [Work partially supported by the ONR and NSWC PCD.]

11:00

1aUW8. Compressional and shear wave modeling in underwater granular media. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

A model of attenuation and sound speed in water-saturated granular media, based on a combination of the Biot–Stoll and contact squirt and shear drag model (BICSQS) [Chotiros, and Isakson, J. Acoust. Soc. Am. **116**(4), 2011–2022, (2004)] and the frame virtual mass (FVM) model [Chotiros, and Isakson, J. Acoust. Soc. Am. **121**(2), EL70 (2007)] is reconciled with the physical dimensions of the area and thickness of the fluid film at the grain-grain contact. The results are consistent with recent experimental observations of enhanced viscosity in nanometer-scale interfacial water films due to

molecular mechanics [Goertz *et al.*, Langmuir **23**, 5491–5497 (2007)]. The resulting model has a reduced set of input parameters and it is able to match both the sound speed dispersion and attenuation measurements from a large number of sites, including the Sediment Acoustics Experiments (SAX99 and SAX04) in the Gulf of Mexico, the Shallow Water Experiment on the Atlantic coast, and the Yellow Sea [Zhou, J. Acoust. Soc. Am. **78**(3) 1003–1009 1985]. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

11:15

1aUW9. Laboratory measurements of sound speed and attenuation in water-saturated sand and glass beads. Theodore F. Argo, IV and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, IX 78713-8029)

The exact nature of sound propagation in water-saturated granular sediments, across the range of frequencies of interest in underwater acoustics, remains insufficiently understood. Well-controlled laboratory measurements are useful for validation and continued development of predictive models. Toward this end, a time-of-flight technique operating in the 250 kHz–750 kHz range was used to determine the sound speed and attenuation in a variety of artificial sediments including sand and glass beads of varying grain size and varying levels of homogeneity. Measurements will be presented and compared to existing model predictions.

11:30

1aUW10. Geoacoustic modeling based on sediment particle analysis. Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu)

Geological underwater processes are affected by physical properties of sediment particles, their size, shape, and spatial variability. In geological modeling, these parameters are normally provided by analysis of sediment cores and documented in certain geologically relevant terms. What is the set of parameters most relevant from acoustics standpoint, or in “acoustically relevant” terms, is an open question. It is addressed using an acoustic scattering model directly based on sediment particle analysis, geo-acoustic model of bottom interaction taking into account the sediment discrete heterogeneity. The particle size distribution in this model is comprised of central and coarse parts. The central part describes the sediment matrix and its large scale variability, or continuous heterogeneity, critical for modeling of acoustic propagation in the sediment. To account for discrete component of heterogeneity, the coarse part of size/shape distributions is attributed to “inclusions” in the sediment matrix and provides direct inputs for evaluation of volume scattering in the sediment. An example of such analysis and modeling based on the SAX04 geo-acoustic data set is presented and used for calculating the frequency-angular dependences of bottom backscattering strength. Comparisons with acoustic backscatter data measured at this site and possibilities for geoacoustic inversions are discussed.

MONDAY AFTERNOON, 23 MAY 2011 — GRAND BALLROOM B, 1:00 TO 3:20 P.M.

Session 1pAAa

Architectural Acoustics and Underwater Acoustics: Computational Methods for Auralization in Air and Water II

Jason E. Summers, Cochair
Applied Research in Acoustics, 1222 4th St., SW, Washington, DC 20024

Michael Vorländer, Cochair
Inst. fur Technische Akustik, RWTH Aachen Univ., D-52056 Aachen, Germany

Chair’s Introduction—1:00

Contributed Papers

1:05

1pAAa1. Psychoacoustic limitations of discrete infinite impulse response and finite impulse response auralizations. Jon W. Mooney (Acoust. Team, KJWW Eng. Consultants, 623 26th Ave., Rock Island, IL 61201, mooneyjw @kjww.com)

Auralizations based on discrete infinite impulse response (DIIR) filters having coefficients derived from statistical reverberation type analyzes are relatively quick and easy to produce but have limitations. Auralizations of very small rooms, or very large rooms having little acoustic treatment, or rooms having dissonant eigenmodes may sound less than real. Auralizations based on discrete finite impulse response (DFIR) filters having coefficients derived from finite element type analyzes can avoid these effects. In both DIIR and DFIR methods, filter coefficient spacing and polarity has a large effect on simulation quality. In this study, auralizations of a test room are created from recorded impulses, derived DFIRs and calculated DIIRs. Sound clips produced by each method are rated using audition and psychoacoustic measures of roughness, fluctuation and sharpness. This paper addresses (1) the derivation of DIIR based auralization from statistical reverberation analysis, (2) the derivation of DFIR based auralization from finite element analysis, (3) psychoacoustic limitations imposed on auralizations, and (4) DIIR/DFIR auralization limits indicated by audition and psychoacoustic rating of a limited number of simulation comparisons.

1:20

1pAAa2. Practical applications and limitations for analog auralizations. Richard A. Vedvik (Acoust. and Vib. Team, KJWW Eng. Consultants, 623 26th Ave., Rock Island, IL 61201, vedvikra@kjww.com)

Developing auralizations of irregularly shaped or small spaces proves to be a challenge for computer software to achieve a natural representation of the original recording. Analog auralizations are discussed using scaled wavelengths, high-bandwidth loudspeakers, and high-bitrate and high-bandwidth measurement equipment, effectively capturing the sound of the analog room. This paper discusses the practical applications and limitations of scale model analog acoustic auralizations. Impulse responses and auralizations of anechoic musical performances in a 1:5 scale music practice room are presented to demonstrate the fine tuning capability of the technique. Interference from outside sources such as ultrasonic emissions of modern electronics is also discussed.

1:35

1pAAa3. Trapping of sound in an open rectangular cavity. Duncan P. Williams (Dstl. Physical Sci., Porton Down, Salisbury SP4 0JQ, United Kingdom), Cristina V. Sargent, and Elizabeth A. Skelton (Imperial College, London SW7 2BZ, United Kingdom)

The trapping of sound in a rectangular enclosure, such as a corridor or street, and the noise disturbance caused by trapping, can be a serious problem to understanding speech or other acoustic events. In contrast to confined systems that sustain so-called trapped modes, the behavior in a partially open enclosure is associated with leaky modes of the system that are "nearly" trapped but are still radiating some energy out of the system. This paper presents numerical modeling that has been developed to calculate the sound that can be nearly trapped in an open rectangular cavity such as the rectangular cross-section of a street canyon that is excited by a number of sources switching on and off. The cavity is surrounded by an elastic half space, i.e., surrounded on three sides by absorbing walls, for which it is necessary to implement a set of numerical grids that are staggered in space and in time. Results are shown for combinations of different walls and different shaped geometries that minimize the radiation losses out of the cavity. The results are used to comment on the problems that can be caused by the trapping of sound in acoustic systems.

1:50

1pAAa4. Application of the mirror source method for sonic booms outdoors. Amanda B. Lind and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, amanda.blair.lind@gmail.com)

Image source theory is commonly applied in architectural acoustic modeling and auralization software, with particular utility in prediction of early reflections. Here, image theory has been employed to model specular reflections of low amplitude sonic booms in outdoor environments. Mechel's presentation of image source theory, the mirror source method (MSM), was implemented and tailored to outdoor environments and plane wave incidence. Given the low-frequency nature of the incident sonic booms, and the high-pass filtering associated with specular reflection from a finite surface, the frequency content of each specular reflection is of particular importance. The specular reflection impulse response (IR) is generated in two steps. First, the MSM is applied to obtain the arrival times and amplitudes of specular reflections, populating the initial IR. Next, each specular reflection is filtered, compensating for the finite size of the reflector. The magnitude response of the minimum phase filter is approximated by the frequency content reflected by a similar finite disk. [Work sponsored by the FAA through the PARTNER Center of Excellence and the Applied Research Laboratory at Penn State.]

2:05

1pAAa5. Combination of wave-based and geometrical-acoustics simulation to study acoustic properties of recording rooms. Wolfgang Ahnert, Stefan Feistel, and Holger Schmalle (Ahnert Feistel Media Group Technologies, Berlin, Germany)

In small recording rooms the acoustic behavior is determined by geometric as well as wave acoustic phenomena. After introducing basic acoustic parameters of these rooms some tools to acquire the necessary acoustic parameters are presented. For this study, a known recording room is used to establish a geometrical-acoustics model using EASE and to derive its geometrical acoustic properties, primarily the transfer function for a certain listener position valid for different monitor speakers. A new tool allows now to derive the modal distribution for the selected speaker-listener positions. Based on the obtained results a transfer function for the low frequency part is derived. Combing the transfer functions of both the geometrical acoustic and the wave acoustic part, the Fourier transform results in broadband impulse responses, which will be compared with the experimentally measured ones.

2:20

1pAAa6. Acceleration of acoustic raytracing using graphics processors. Zuofu Cheng and Lippold Haken (Dept. of Elec. and Comput. Eng., Univ. of Illinois, 1406 W. Green St., Urbana, IL 61801, zcheng1@illinois.edu)

OpenCL presents an industry standard interface to accelerate computationally expensive algorithms on consumer grade graphics hardware. Presented is a system in which raytracing for room acoustic simulation is accelerated with OpenCL. The system is primarily geared toward realtime applications such as video gaming or virtual reality, but may be extended to higher quality simulations for architectural acoustics. Different methods for mapping the raytracing problem onto the GPU are discussed, as well as strategies for optimizing the algorithm for graphics processors from different manufacturers.

2:35

1pAAa7. How to get to Carnegie Hall: Real time signal processing. Alexander E. Post, Shane Cotter, and Palmyra Catravas (Dept. ECE, Union College, 807 Union St., Schenectady, NY 12308, posta@garnet.union.edu)

Since the acoustics of a practice room differ so greatly from a performance hall, it could be beneficial for musicians to prepare for a particular venue using a simulated version of that environment. The system described here makes use of a microprocessor and headphones to realize this virtual environment. The nominal room response is attenuated by the headphones, replaced by the response of the desired location. Usability goals include simplicity and portability, as well as the ability of the system to fade in to the background while in use, so that the user may focus on practice. We will report on the design and implementation of this system, the details of the real time processing system, and present measurements made to characterize the impulse response of practice rooms and performance halls. [Work supported from the National Science Foundation MRI 0923384 is gratefully acknowledged].

2:50

1pAAa8. Acoustic performance of an installed real-time three-dimensional audio system—Part II. Kenneth J. Faller, II (NASA Postdoctoral Program, Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA 23681-2199), Stephen A. Rizzi (NASA Langley Res. Ctr., Hampton, VA 23681-2199), and Aric R. Aumann (Analytical Services and Mater., Inc., Hampton, VA 23666)

The exterior effects room (EER), located at the NASA Langley Research Center, was recently upgraded to allow for simulation of aircraft flyovers in a three-dimensional (3-D) audio and visual environment. The 3-D audio server employs an implementation of the vector base amplitude panning (VBAP) method to position virtual sources at arbitrary azimuth and elevation angles in the EER. Recent work focused on the development of loudspeaker equalization first using high order FIR filters [POMA **9** 015004 (2010)], and later using low order IIR filters [J. Acoust. Soc. Am. **128** 2482 (2010)]. The latter, in conjunction with full-path time delay compensation and relative gain compensation, were implemented in real-time and shown to reproduce the desired sound field to within about ± 5 dB over an extended frequency range for stationary and moving sources. In the present work, the performance is further characterized both objectively and subjectively. Addressed are calibration of the system for absolute sound pressure level reproduction and measurement of the spatial uniformity of the generated sound field. Further, localization of sound sources will be subjectively measured to assess the efficacy of the VBAP implementation in the EER.

3:05

1pAAa9. Effects of excessive noise and reverberation on listening and learning in a simulated classroom. Daniel L. Valente, Dawna Lewis, Elizabeth Heinrichs, Jody Spalding, John Franco (Ctr. for Hearing Res., Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, daniel.valente@boystown.org), and Hallie Plevinsky (Univ. of Maryland, College Park, MD, 20742)

Elementary students often learn in dynamic discussions during typical classroom lessons. Many classrooms, though, have poor signal-to-noise ratios and long reverberation times. The presence of excessive noise and reverberation may increase a student's listening effort and result in reduced performance during classroom learning. A simulated classroom environment was created which allowed for varying degrees of room reverberation and background noise. In this experiment, groups of elementary-aged students were seated in the center of the simulated classroom environment and were presented a story read by either five talkers positioned around the student or a single talker in front of the student (reproduced by LCD monitors and loudspeakers). A post-test was used to assess listener comprehension. Com-

prehension scores are compared to a group of adult subjects as well as a sentence-recognition task in the same condition. Significant differences were seen in comprehension scores as a function of age and condition, both increasing background noise and reverberation degraded performance in comprehension tasks compared to little or no differences in measures of sentence-recognition. Finally, comprehension scores are correlated to measures of the speech transmission index in each of the simulated classroom environments.

MONDAY AFTERNOON, 23 MAY 2011 GRAND BALLROOM B, 3:35 TO 5:55 P.M.

Session 1pAAb

Architectural Acoustics and Noise: Acoustics of Green Buildings

David M. Sykes, Cochair
Remington Partners LLC, 23 Buckingham St., Cambridge, MA 02138

Brandon Tinianov, Cochair
Serious Materials, 1250 Elko Dr., Sunnyvale, CA 94089-2213

Ralph T. Muehleisen, Cochair
Illinois Inst. of Technology, Civil and Architectural Engineering, Chicago, IL 60616

Chair's Introduction—3:35

Invited Papers

3:40

1pAAb1. Efforts to improve the acoustics requirements in the proposed International Green Construction Code. Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, asaseattle@sacnc.com)

In the spring of 2010 the acoustics community became aware of a proposed set of acoustical requirements in the International Green Construction Code being developed by the International Codes Council. The acoustics section as proposed contained many problems. Several members of the Acoustical Society of America submitted comments pointing out the problems and suggesting changes or recommending that action be delayed until appropriate changes could be developed. The ICC accepted only a few of the comments submitted for inclusion in the next draft. Subsequently, a member of the ICC committee that reviewed the comments offered to help with comments on the new draft and suggested that what the ICC needed was a recommendation endorsed by the acoustical community. An effort was made to develop a recommendation endorsed by various organizations involved in acoustics to be submitted in early January 2011. This paper will discuss the original ICC proposal, the efforts to change it, the changes proposed, and the results if known of the hearings to be held the week before the ASA Seattle meeting.

4:00

1pAAb2. New Center for the Built Environment data reinforces poor acoustical performance of "green" office buildings. Kevin Powell (US General Services Administration, Office of Appl. Res., 555 Battery St., Rm. 518, San Francisco, CA 94111)

Analysis by the US General Services Administration of new end-user-satisfaction data compiled by UC Berkeley's Center for the Built Environment (CBE) continues to reinforce that green building interiors built according to the LEED Rating system suffer from a decline in acoustical performance. The facts are that occupants of offices that have been upgraded to achieve LEED certification express high levels of satisfaction with the Indoor Environmental Quality with one exception—acoustics. What is it about LEED-rated offices that causes this unintended consequence to occur? Is it because there are more reflective, hard surfaces causing reverberation? Or because the HVAC system has become quieter? Or because partition heights have been lowered to allow better airflow and more natural light, resulting in less privacy? Since increased occupant dissatisfaction may result in reduced productivity or increased absenteeism, GSA's Office of Applied Research recently held an Acoustics Workshop in Washington DC to develop new guidelines for the construction of office environments that will eventually affect the GSA's entire 8600-building portfolio.

4:20

1pAAb3. Current and future acoustics credits in the leadership in energy and environmental design rating system. Daniel C. Bruck (BRC Acoust. & Technol. Consulting, 1741 First Ave. S., Seattle, WA 98134), Charles M. Salter (Charles M. Salter Assoc., San Francisco, CA 94104), and Alexis Kurtz (New York, NY)

The next version of LEED, currently in development, includes a new acoustics credit that is directed toward acoustical performance for all types of new construction projects. The credit includes performance requirements for background noise, speech privacy, sound isolation, sound reinforcement, and sound masking systems. As an acoustical performance credit for all types of new construction, the credit allows flexibility to address a wide range of performance criteria for various types of projects. This paper, coauthored by the acoustics working group within the U.S. Green Building Council, will present the intent, provisions, and background on the development of the new credit. A review of existing acoustics credits in LEED and approaches for compliance with existing credits will also be discussed.

4:40

1pAAb4. Laboratory experimental investigation of the acoustical characteristics of vegetated roofs. Maureen Connelly and Murray Hodgson (Dept. of Interdisciplinary Grad. Studies., Univ. of British Columbia, 2206 East Mall, Vancouver, BC V6T 1Z3, Canada, mconnell@interchange.ubc.ca)

A laboratory-based experimental investigation was made of the sound-absorption and sound-transmission characteristics of vegetated roofs and/or their components. Impedance-tube measurements of the normal-incidence absorption coefficients of different substrates were related to their physical characteristics, finding that percentage organic matter, volumetric water content, and compaction are the key influencing factors. Spherical-decoupling-method measurements of the normal-incidence absorption coefficients of different thicknesses of substrate without and with vegetation showed that the absorption of substrates tends to increase with frequency, substrate depth, and to decrease with moisture content and vegetation. Sound-intensity measurements of the green roofs with different thicknesses of dry or wet substrate without and with vegetation, made in a purpose-built facility, showed that transmission loss increased rapidly with frequency, and increased with substrate depth and water content.

Contributed Papers

5:00

1pAAb5. Acoustical design of heathcare facilities for leadership in energy and environmental design green building certification: Odd harmonic filter for mechanical system noise control. Seth M. Harrison (Acoust. and Vib. Group, KJWW Eng. Consultants, 623 26th Ave., Rock Island, IL 61201, harrisonsm@kjww.com)

To achieve the leadership in energy and environmental design (LEED) indoor environmental quality credit 2: acoustic environment, healthcare facilities must meet stringent sound and vibration criteria. Mechanical system noise presents one of the largest obstacles to meeting the room noise design requirements. Code requirements intended to promote indoor air quality, and infection control limits the use of typical absorptive materials such as fiberglass duct liner, packed duct silencers, and other porous absorbers. Energy code requirements force mechanical system designers to locate large noise sources such as air handlers, fan-powered terminal air boxes, and exhaust fans close to the spaces they serve, adding to background noise levels. One innovative, practical solution to tonal mechanical system noise is an odd harmonic filter. Acoustical performance of odd harmonic filters and strategies to achieve the room noise criteria for the LEED acoustic environment credit will be discussed.

5:15

1pAAb6. Optimization of silencers for interior natural ventilation openings. Chris Bibby (Dept. of Mech. Eng., Univ. of British Columbia, 3rd Fl., 2206 East Mall Vancouver, BC, Canada V6T 1Z3) and Murray Hodgson (Univ. of British Columbia, Vancouver, BC V6T 1Z3, Canada)

Naturally ventilated buildings are being constructed around the world due to advantages they offer in reduced HVAC operating costs, increased indoor air quality, and government subsidies for “green” construction and certification. Unfortunately, natural ventilation is often associated with a decrease in occupant acoustical satisfaction. One reason for this is the requirement to create acceptable air exchange rates in naturally ventilated buildings that have very limited pressure differentials; airflow paths must offer very low resistance. This leads designers to create paths by connecting adjacent spaces with apertures in the partitions or to use short transfer ducts rather than conventional duct systems. These apertures can be detrimental to the noise isolation provided by the partition. Products exist to attenuate noise through these apertures, but little exists in the literature about their performance. In this work, current methods and new methods of passively silencing interior natural ventilation openings have been evaluated and optimized using finite element simulations. To facilitate the optimization, silencer evaluation and selection criteria based on air flow and acoustical characteristics are proposed.

5:30

1pAAb7. Using a sustainable material to produce a sound absorbing panel. Shane J. Kanter (The Univ. of Kansas School of Architecture, Design, & Planning, 1465 Jayhawk Blvd., Lawrence, KS 66045, shanekanter@gmail.com)

A recent university research project involved the creation of an environmentally friendly sound absorbing panel using bamboo. The goal of this research was to produce a panel with a sustainable material that could be used in similar applications where wood fiber sound absorbing panels would be employed. The test bamboo panel has been made from bamboo rings approximately 1.6 mm thick cut from bamboo stalks with diameters varying from approximately 6 to 50 mm. The bamboo rings were randomly formed into a panel about 25 mm thick using an environmentally friendly adhesive with very low viscosity. Panel sound absorption has been determined using the reverberation room method, an *in situ* method, and an impedance tube method. [This research was partially funded by an Acoustical Society Robert W. Young award.]

5:45—5:55 Panel Discussion

Session 1pABa

Animal Bioacoustics: Long-Term and Cumulative Effects of Sound on Animals, Including Acoustic and Non-Acoustic Stressors

Christine Erbe, Chair
JASCO Applied Sciences, 9116 N. Twilight Ct., Spokane, WA 99208

Invited Papers

1:15

1pABa1. Assessing cumulative impacts of underwater noise with other stressors on marine mammals. Andrew J. Wright (Natl. Environ. Res. Inst., Dept. of Arctic Environment, Aarhus Univ., Frederiksborgvej 399, postboks 358, DK-4000 Roskilde, Denmark, awr@dmu.dk) and Lindy Weilgart (Dalhousie Univ., Halifax, Nova Scotia B3H 4J1, Canada)

Cumulative impact assessments (CIAs) are an often unmet requirement in many environmental impact assessment processes. However, marine mammals are typically exposed to multiple human activities and pollutants including noise, which can combine in various ways including through chronic stress responses. To address the issue, the Okeanos Foundation held an international, multi-disciplinary workshop in Monterey, CA (August 2009). Participants considered three aspects: how currently available tools for regionally mapping several anthropogenic pressures on the environment could be applied to species management, how the reported consequences in marine mammals of exposure to these pressures and their known interactions within an individual could be modeled, and how population modeling could include cumulative impacts. Participants felt that all three approaches could be realized in certain data-rich marine mammal populations, which could then be used as examples for informing management decisions in other marine mammals. The population modeling for cumulative impacts on Western gray whales and Southern and North Atlantic right whales is currently underway. Participants believed that marine spatial planning would facilitate better CIAs and that reducing ocean noise is an achievable goal that will help marine life cope with less tractable threats such as climate change.

1:30

1pABa2. Terminology and conceptual frameworks for cumulative effects analysis. Robert Gisiner (Navy Energy and Environ. Readiness Div. (OPNAV N45), Arlington, VA 22202, bob.gisiner@navy.mil) and Samantha Simmons (US Marine Mammal Commission, Bethesda, MD 20814)

In the short history of underwater noise regulation most efforts have focused on the risk posed by a single source over a short period. However, noise in the workplace, cityscape, and terrestrial environment are typically expressed in terms of the cumulative consequences of repeated exposures from multiple sources over long periods. Recent publications and workshop reports on cumulative effects' analysis are reviewed, and a terminology and conceptual approach are proposed to facilitate analysis of the cumulative effects of underwater noise. The term cumulative effect is reserved for the accretion of effects over time while the term aggregate effect is proposed for the effect of multiple concurrent stressors, both acoustic and nonacoustic. Possible metrics for comparing effects across individual stressors are discussed, including measures of caloric cost and physiological stress. A modeling framework is proposed, by which the synergistic effect of combined stressors might be quantitatively assessed.

1:45

1pABa3. Assessment of cumulative effects of underwater sound: A collaborative approach. Roberto Racca (JASCO Appl. Sci., 2101-4464 Markham St., Victoria, BC V8Z 7X8, Canada, roberto.racca@jasco.com)

A multidisciplinary team of scientists working under a research agreement between the University of California, Santa Barbara and BP America Production Company has undertaken a 2-year project, currently ongoing, to develop one or more standardized and practical methods for assessing cumulative effects of anthropogenic underwater sound on marine mammals. The work of the team is based on access to existing scientific information without substantial involvement of additional primary research although topics for future research may be part of the eventual recommendations. While the final goal of the project is to conceptualize and specify widely applicable methodologies, case studies are being used to help formulate and test the feasibility of assessment frameworks. The primary case study involves the significant hydrocarbon related industrial activity that took place in the Beaufort Sea (Alaska and Canada) during late summer and autumn 2008 and its potential cumulative effect on the population of bowhead whales (*Balaena mysticetus*) that dwell and transit through the region in their annual migration cycle. Methods that include advanced numerical modeling of sound propagation, individual-specific dose exposure calculation, and sensitivity analysis of the range of potential responses as a function of various influence factors are applied in the execution of these studies.

2:00

1pABa4. A bioenergetics approach to understanding the population consequences of acoustic disturbance. Daniel P. Costa, Lisa K. Schwarz, and Patrick W. Robinson (Dept. of Ecology Evolutionary Biology, Univ. of California, Santa Cruz, CA 95060)

[A major hurdle with marine mammal conservation and management is to know if and when measurable short term responses result in biologically meaningful changes in populations. We are developing a bioenergetics approach to parametrize the transfer functions developed in the conceptual model developed by the NRC Committee on the population consequences of acoustic disturbance (PCAD). Our effort is directed at quantifying the life functions that are linked to vital rates, and how changes in these vital rates affect populations. Such an approach can identify species and or particular life history characteristics that are likely to be sensitive or resilient to acoustic disturbance. Using species that represent the range of life history patterns observed in marine mammals we are analyzing the existing data to determine whether there is a linkage between fine scale measurements of foraging behavior and reproductive success and survival. These data are being used to develop a time-activity budget to produce a first order quantitative assessment of the potential significance in terms of lost energy and/or time that foraging behavior or habitat utilization is potentially affected by acoustic disturbance.

2:15

1pABa5. A regulatory approach to assessing the cumulative effects of anthropogenic noise and other stressors on endangered and threatened marine animals. Craig Johnson (Office of Protected Resources, U.S. Natl. Marine Fisheries Service, 1315 East-West Hwy., SSMC3, Silver Spring, MD 20910, craig.johnson@noaa.gov)

The U.S. Endangered Species Act of 1973 requires the U.S. National Marine Fisheries Service (NOAA-Fisheries) to assess the direct, indirect, interactive, and cumulative impacts of anthropogenic noise and other stressors on species that are listed as endangered or threatened under U.S. law. NOAA-Fisheries' assessment method makes the cumulative impact component of these assessments more tractable by recognizing that individual organisms, populations, and species accumulate the effects of noise and other anthropogenic stressors differently. Assessing the cumulative impacts of noise and other anthropogenic stressors on the fitness of individuals remains the most critical and challenging component of these assessments. The concepts of canonical cost [McNamara and Houston, Am. Nat. **127**, 358–378 (1986)] and vitality [Anderson, Ecology **70**, 445–470 (2000)] currently provide the foundation for qualitative assessments of the cumulative impacts of stressors on individuals while the cumulative impacts of these stressors on populations and species are assessed using existing population viability and perturbation analyzes. Further development of the analytical methods associated with canonical costs and vitality would facilitate quantitative assessments of the cumulative impacts of noise and other anthropogenic stressors on endangered and threatened individuals, populations, and species.

2:30

1pABa6. The implications of long-term increases of anthropogenic noise on fish. Arthur N. Popper and Brandon M. Casper (Dept. of Biology, Univ. of Maryland, College Park, MD 20742, apopper@umd.edu)

Most of the focus regarding environmental sound and fishes has been on (potential) effects of intense sounds such as pile driving and seismic air guns. There has been less concern, however, about the (potential) effects on fishes of less intense, but continuous increases in the overall noise environment in which fishes live. Such sources may include increased shipping in harbors, operations of off-shore wind farms, added noise in aquaculture facilities, etc. While the increase in ambient noise may be only a few decibels, such increases could have substantial effects on fishes resulting from the masking biologically relevant sounds and/or the overall acoustic scene. Moreover, and in contrast to intense sounds which generally ensonify any area for only a short period of time (e.g., sonar on a moving ship) or only ensonify relatively small areas (e.g., pile driving) that fish can leave, longer term sources may ensonify such large areas that many fish species cannot move from the area or would have to leave feeding and/or breeding sites in order to get to a quieter environment. This paper will review some data on potential effects of increased environmental sound on fishes and consider the long-term implications of this increase.

2:45

1pABa7. The effects of noise on birds. Robert J. Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742, dooling @psyc.umd.edu)

The effects of noise on birds can be considered as falling into four overlapping categories: hearing damage and permanent threshold shift (PTS) from acoustic overexposure, temporary threshold shift (TTS) from acoustic overexposure, threshold elevation and masking of important biological sounds, and finally, any other physiological and behavioral responses that might occur to a novel sound that is sufficiently above the natural background noise level so as to be detectable. In the first three cases at least, auditory effects depend strongly on the level and duration of noise exposure, which is highly correlated with the proximity of the bird to the noise source. Drawing on both laboratory and field data, the current state of knowledge in the field is summarized. In addition, this review will identify general principles and important exceptions, clarify and define the potential adverse effects, and suggest future approaches to understanding the effect of noise on acoustic communication in birds in their natural enviroment. [Work supported by the NPS.]

3:00

1pABa8. Assessing the effects of mid-frequency sonar on beaked whales in Southern California. Mariana L. Melcon, Amanda J. Cummins, Anne E. Simmonis, Simone Baumann-Pickering, Marie Roch, Sean M. Wiggins, and John A. Hildebrand (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA 92093-0205, mmelcon@ucsd.edu)

Naval exercises include the use mid-frequency active (MFA) sonar, which emits high intensity sound to obtain an acoustic picture of the environment. Little is known about the effects of MFA sonar on cetaceans, but correlations were observed between naval exercises and anomalous massive strandings of beaked whales. Passive acoustic monitoring was used to start studying the possible impact of MFA sonar on beaked whales. For this, high-frequency acoustic recording packages were deployed to continuously record sounds between 10 Hz and 100 kHz. First, the probabilities of beaked whales given MFA or no-MFA were calculated for one site in the Southern California Bight. The acoustical presence of the animals was 50% or less when MFA was present compared to the no MFA situation. Furthermore, the proportion of time with beaked whale calls decreased proportionally to the intensity of sound in the frequency band of MFA sonar. Finally, the acoustic presence of beaked whales and the intensity of mid-frequency noise at five different sites of the Southern California Bight were analyzed for 1.5 years. Possible effects of MFA sonar on these animals will be discussed as a function of exposure and geographical movements.

3:15—3:30 Break

Contributed Papers

3:30

1pABa9. Effects of human-generated sounds on saltwater fish. Caitlin N. Conway (Dept. of Phys., Salisbury Univ., 1101 Camden Ave., Salisbury, MD 21801, cc48792@gulls.salisbury.edu) Mark W. Muller (Dept. of Phys., Salisbury Univ., 1101 Camden Ave., Salisbury, MD 21801)

There is increasing apprehension of the potential adverse effects on marine mammals and fish caused by human-generated sounds. This study demonstrates the effects of low-frequency and high-frequency sounds on three spcies of fish, as such effects have been shown to be species specific. Three representative species from saltwater fish are chosen for this study. These species include Damselfish (*Chrysiptera parasema*), Clownfish (*Amphiprion percula*), and Molly fish (*Poecilia sphenops*). Ten of each species of fish are placed in a thirty gallon tank filled with water of the appropriate salinity level. The fish are monitored for one week in the absence of anthropogenic noise. The experiment portion consits of emitting short pulses of high-frequency sound waves and then long pulses of low-frequency sound waves over the fish for a predetermined duration. The fish are monitored for any changes in their behavior between each exposure of sound waves. This study may provide further insight into the effects of human-generated sounds on fish in their respective natural environments.

3:45

1pABa10. Behavioral response of Australian humpback whales to seismic surveys. Douglas H. Cato (Defence Sci. & Tech. Org. and Univ. of Sydney, P.O. Box 44, Pyrmont, New South Wales 2009, Australia, doug.cato@sydney.edu.au), Michael J. Noad, Rebecca A. Dunlop (Univ. of Queensland, Gatton, Queensland 4343, Australia), Robert. D. McCauley, Chandra P. Salgado Kent (Curtin Univ., Perth, Western Australia 6845, Australia), Nicholas J. Gales (Australian Antarctic Div., Kingston, Tasmania 7050, Australia), Hendrick Kniest (Univ. of Newcastle, Newcastle, New South Wales 2308, Australia), John Noad (Univ. of Queensland, Gatton, Queensland 4343, Australia), and David Paton (Blue Planet Marine, Canberra, Australian Capital Territory 2614, Australia)

The first of four major experiments in project behavioural response of australian humpback whales to seismic surveys (BRAHSS) was conducted on the east coast of Australia in September and October 2010. The project aims to understand how humpback whales respond to seismic surveys and to provide the information that will allow these surveys to be conducted efficiently with minimal impact on whales. It also aims to determine how the whales react to ramp up or soft start, and to assess how effective this is in mitigation. The 2010 experiment used a single air gun. Four air guns will be used in the next two experiments and a full seismic array in the final experiment in 2013. During the 2010 experiment, behavior and tracks of whales were recroded by four theodolite stations on elevated coastal positions and DTAGs used on some whales. Vocalizing whales were tracked with a wide base line hydrophone array. A further four acoustic recorders were used to measure propagation loss and to characterize the sound field throughout the area. A wide range of variables likely to affect whale response was measured. [Work sponsored by the JIP E&P Sound & Marine Life and Bureau of Ocean Energy Management, Regulation and Enforcement.]

4:00

1pABa11. Noise exposure and acoustic behavior of beluga whales (Delphinapterus leucas) in an outdoor exhibit. Cara F. Hotchkin and Susan E. Parks (Penn State Univ., P.O. Box 30, State College, PA 16804, cfh121@psu.edu)

Marine mammals are exposed to a wide variety of noise sources on a daily basis both in the wild and at zoos and aquaria. Filtration systems, aquarium visitors, and other anthropogenic sound sources (construction, road noise, weather, etc.) add noise to typically reverberant animal habitats in aquariums and zoos. Despite widespread concern about the effects of noise on behavior and welfare of wild marine mammals, there has been little research on the effects of chronic noise exposure on whales. This study used a single hydrophone system to characterize the acoustic environment, including noise and beluga whale vocalizations, in the Arctic Coast exhibit at Mystic Aquarium, a Division of Sea Research Foundation, Inc. between 28 October and 16 November 2010. Diel patterns in noise and beluga vocalizations, overlaps in call and noise frequencies, and general behavioral patterns were studied. Spatial variation in exhibit noise was also examined. Whale vocalizations had a distinct diel pattern, including changes in call rate and types during day and nighttime hours. Variations in noise levels were often associated with discrete events, including cleaning dives, filter backwash, and other exhibit maintenance.

4:15

1pABa12. Sealscarer induce behavioural responses of harbor porpoises (*Phocoena phocoena*). Caroline Hschle, Miriam J. Brandt, Ansgar Diederichs (BioConsult SH, Brinckmannstrasse 31, Husum, Schleswig-Holstein, 25813, Germany), Klaus Betke (Itap, Marie-Curie-Strasse 8, Oldenburg, Niedersachsen 26129, Germany), and Georg Nehls (BioConsult SH, Brinckmannstrasse 31, Husum, Schleswig-Holstein 25813, Germany)

The construction of several wind parks in the North and Baltic Sea and the associated pile driving could negatively affect the only resident cetacean species, the harbor porpoise. We studied the efficiency of a Lofitech sealscarer that emits pulses at a main frequency of 14 kHz to temporarily deter harbor porpoises during construction periods in order to avoid physical injury. Reactions of harbor porpoises were studied by tracking the animals from a cliff 20 m above sea level using a theodolite. Six trials showed animals immediately disappearing after sealscarer activation of 300–700-m distance to the porpoise with no animal resurfacing within 1 km. A greater deployment distance of 1.2–2.6 km to the porpoise provoked immediate disappearance of two porpoises; four porpoises showed a clear avoidance reaction and change of swimming direction. Reactions of two animals were unclear and these consequently lost; one individual resumed with its normal behavior. These results show that the sealscarer caused porpoises to leave. However, the use of sealscarers cannot guarantee complete absence of porpoises within areas where noise from windfarm construction can potentially harm them since reactions possibly differ between, for example, individuals and/or motivational states.

4:30

1pABa13. Presence of harbor porpoises near a pile driving site and modeling of cumulative acoustic effects. Klaus Lucke (Res. and Technol. Ctr. Westcoast, Univ. of Kiel, Werftstrasse 6, 25761 Buesum, Germany), Paul. A. Lepper (Loughborough Univ., Loughborough LE11 3TU, United Kingdom), Michael Daehne, and Ursula Siebert (Univ. of Kiel, Werftstrasse 6, 25761 Buesum, Germany)

The construction of the a small wind farm in the southern North Sea has just been completed. Possible effects on harbor porpoise (*Phocoena phocoena*) distribution and habitat use caused by the pile driving impulses emitted during the installation of the foundations for 12 offshore wind turbines (OWTs) were assessed by ship surveys, aerial surveys, and static acoustic monitoring (SAM). SAM data and sighting data revealed that during the pile driving activities for the construction of the transformer platform as well as for the first OWTs the presence of harbor porpoises decreased

significantly. However, the number of porpoise detections by the SAM units increased during the pile driving activities toward the end of the construction period. Habituation, ecological parameters, and external effects are considered as potential reasons for this increased presence of harbor porpoises in the vicinity of the construction site. The cumulative acoustic energy received by the animals exposed to repeated pile driving impulses is modeled for different scenarios in order to assess potential negative auditory effects on the animals.

4:45

1pABa14. Assessing long-term impacts of vocal compensation to ambient noise by measuring the metabolic cost of sound production in bottlenose dolphins. Dawn P. Noren, Marla M. Holt (NOAA NMFS Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. East, Seattle, WA 98112, dawn.noren@noaa.gov), and Terrie M. Williams (Univ. of California, Santa Cruz, CA 95060)

The use of sound is essential to the survival of cetaceans. Therefore, anthropogenic sound exposure is a concern. Cetaceans can change the amplitude, duration, repetition rate, and/or frequency of sounds they produce to compensate for masking noise. Potential costs of such compensation are unknown, and no empirical data on the metabolic cost of sound production in marine mammals exist. This study aims to determine the metabolic cost of cetacean vocalizations to assess the biological significance of vocal compensation. Oxygen consumption, respiration rates, and vocalizations of two captive bottlenose dolphins were recorded during sound production at moderate levels. One dolphin produced his signature whistle while the other produced a pulsed squawk or squeaklike sound. Both types of vocalizations increased oxygen consumption while respiration rates did not change. This increased oxygen consumption is likely due to increased metabolic demand related to sound production, rather than changes in breathing patterns. Thus, acoustic signals performed in response to increased vessel presence and anthropogenic noise may increase metabolism in cetaceans. Depending on the duration and intensity of exposure, it is possible that vocal compensation responses increase total daily energy expenditure, which may impact daily prey requirements. [Study supported by the U.S. Office of Naval Research.]

5:00

1pABa15. Measurements of ship noise and calling behavior on bioacoustic probes during opportunistic exposure of blue whales to commercial ship noise. Megan F. McKenna (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093, mmckenna@ucsd.edu), Erin M. Oleson (Pacific Islands Fisheries Sci. Ctr.), Jeremy A. Goldbogen (Univ. of California San Diego, La Jolla, CA 92093), and John Calambokidis (Cascadia Res. Collective, Olympia, WA 98501)

Bioacoustic probes were deployed on blue whales in and around commercial shipping lanes in Southern California. The proximity of shipping routes to predictable blue whale feeding grounds makes this an ideal location to study the impact of intense low-frequency noise on whale behavior. The bioacoustic probe is outfitted with a calibrated hydrophone, and records temperature, pressure, and two-axis acceleration, allowing us to evaluate sound exposure levels in relation to swimming and calling behaviors. During eight tag deployments, large commercial ships came within 1000 m or less of the tagged whale. Received levels of ship noise were measured from the tag to estimate the sound exposure levels. Because of the high level of flow noise recorded on the tag, measurements of ship noise were only possible at minimum velocity and speed. Dive and surface behaviors were evaluated related to the passage of the ship and elevated noise levels. One calling whale was tagged and exposed to ship noise at a close approach; no major change in calling behavior was observed. The results of this research provide insight on blue whale response to ships and highlight some of the challenges when measuring low frequency noise from an attached tag. [This work was supported by the ONR.]

MONDAY AFTERNOON, 23 MAY 2011 ASPEN, 1:25 TO 4:45 P.M.

Session 1pABb

Animal Bioacoustics: Marine Mammal Depredation

Aaron Thode, Chair
Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Chair's Introduction—1:25

Invited Papers

1:30

1pABb1. Southeast Alaska Sperm Whale Avoidance Project: Collaboration among fishermen, acousticians, biologists and managers to reduce longline depredation in the Gulf of Alaska. Janice Straley (Univ. of Alaska Southeast, 1332 Seward Ave., Sitka, AK 99835, jmstraley@uas.alaska.ed), Aaron Thode (Scripps Inst. of Oceanogr., San Diego, CA 92093-0205), Victoria O'Connell (Coastal Marine Res., Sitka, AK 99835), Linda Behnken (Alaska Longline Fishermen's Assoc., Sitka, AK 99835), Kendall Folkert (Fisherman F/V Cobra, Sitka, AK 99835), Sarah Mesnick (NOAA Fisheries, La Jolla, CA 92037), and Chris Lunsford (NOAA, Juneau, AK 99801)

In the eastern Gulf of Alaska, depredation of demersal longline fishing gear set for sablefish (*Anoplopoma fimbria*) by sperm whales (*Physeter macrocephalus*) has occurred since at least the mid-1970s. In 1995, with the implementation of Individual Fishing Quotas, the season expanded from a 2 week derby style fishery to 8 months. This change allowed more opportunity for whales to depredate longline gear and reports of depredation increased resulting in economic loss to the fleet. Beginning in 2003, the North Pacific Research Board funded a collaborative study among fishermen, scientists, and managers to collect quantitative data on longline depredation. The goal of the Southeast Sperm Whale Avoidance Project (SEASWAP) is to determine the mechanics of the depredation, characterize the whales involved, and to recommend changes in fishing behavior to reduce depredation. The longline fleet provided fishing data and allowed researchers to collect behavioral, genetic, and acoustic data in conjunction with their fishing operations. Through this effort, SEASWAP

has estimated numbers and sexes of depredating whales, identified long-range acoustic cues that attract animals to the gear, and systematically tested various depredation reduction strategies, including decoy deployments, active playbacks, and modifications to fishing gear.

1:50

1pABb2. Differences between natural and depredation behaviors in sperm whales (*Physeter macrocephalus*), using dive, acoustic, and orientation data from short-term tags. Delphine Mathias, Aaron Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA 92093-0205), Jan Straley (Univ. of Alaska Southeast, Sitka, AK 99835), John Calambokidis, Gregory S. Schorr (Cascadia Res. Collective, Olympia, WA 98501), and Kendall Folkert (P.O. Box 6497, Sitka, AK 99835)

Sperm whales depredate black cod (*Anoplopoma fimbria*) from demersal longlines in the Gulf of Alaska. Over a 3-year period 11 bioacoustic tags have been attached to adult sperm whales off Southeast Alaska under both natural and depredation foraging conditions. Measurements of the animals' dive profiles, acoustic behavior, and angular velocity under both behavioral modes were distilled and examined for statistically significant differences. Two rough categories of depredation are identified. The dive depths and durations of "deep depredating" whales are similar to those of natural dives, but acoustic parameters and roll rates show significantly significant differences. By contrast, "shallow depredating" whales conduct dives that are much shorter, shallower, and more acoustically active than natural foraging dives. Acoustic "creak" rates, which serve as proxies for prey capture attempt rates, are four times greater during shallow depredation than during natural foraging. During all behaviors, whales generating creak sounds had significantly greater roll rates when the creak was followed by a pause, suggesting that acoustic monitoring might distinguish prey capture successes from failed attempts. Fish that naturally "spin-off" the line during a haul may explain facets of deep depredation. [Work supported by the North Pacific Research Board, the National Geographic Society, and OGP.]

2:10

1pABb3. False killer whales and Hawaii's longline fisheries. Erin M. Oleson (NOAA Fisheries Pacific Islands Fisheries Sci. Ctr., 1601 Kapiolani Blvd., Ste. 1110, Honolulu, HI 96814, erin.oleson@noaa.gov) and Karin A. Forney (NOAA Fisheries Southwest Fisheries Sci. Ctr., 110 Shaffer Rd., Santa Cruz, CA 95060)

False killer whales have been implicated in depredation, or removal of catch or bait, from pelagic longline fisheries throughout the Indo-Pacific. These interactions often result in significant financial loss to the fishing industry, and though uncommon, hooking, or entanglement of false killer whales during depredation often leads to death or serious injury of the whales. False killer whale bycatch in Hawaii's longline fisheries exceeds allowable limits established under the Marine Mammal Protection Act. In 2010, the National Marine Fisheries Service convened a Take Reduction Team to develop a plan for reducing serious injuries and mortalities of false killer whales caused by such interactions. Means of avoiding or minimizing depredation were an important aspect of the Team's deliberations. Among other factors, measures of noise produced by fishing vessels or gear and local acoustic propagation may help predict false killer whale-fishing gear interactions. Acoustics studies are now underway to evaluate noise produced by fishing vessels and the setting, soaking, and hauling of gear, and to evaluate the behavior of false killer whales when they interact with gear. Additional studies on echolocation jamming or acoustically disguising gear were also suggested by the Team.

2:30

1pABb4. Development of an acoustic-based toothed whale depredation detection system for longline fisheries. Geoff McPherson, Gary Clarke, Brian Hingley (Global Detection Systems, Ste. 3, Aria Bldg., 38-46 Albany St., St Leonards, New South Wales 2065, Australia, mcpherson.geoff@gmail.com), Craig McPherson, and Christine Erbe (Jasco Appl. Sci. Australia, Eight Mile Plains, Queensland 4113, Australia)

Depredation on baited hooks and catch of longline fishing gear by toothed whales is a serious source of fishery product loss and increased operating costs. Depredation in tropical waters is dominated by false killer, short fin pilot, and melon headed whales. Depredation is associated with increased acoustic communication, both echolocation and whistles. Longlines take many hours to set and haul over tens of kilometers. Depredation detection would assist operators make decisions to reduce fishery loss. Termination of setting when depredation was occurring on recently set gear, or hauling sections of gear where depredation had not as yet occurred, would help reduce losses. Depredation avoidance would reduce risk of bycatch. Global Detection Systems is developing a depredation proximity detector. A GPS-based and web-enabled buoy will indicate locations of the buoys on the line. Detection is by a sensor array developed for fishing vessels. Detection, irrespective of the toothed whale species involved within a radius of the buoy, will be processed on-board the buoy using an entropy based detection system [Erbe, C., J. Acoust. Soc. Am. **127**, 1959 (2010)]. The radius of detection is being determined in fishery trials in Australian and Central Pacific waters. [Work supported by AusIndustry and Commercialisation Australia.]

2:50

1pABb5. Examining the biosonar process involved with depredation of black cod by sperm whales in the Alaskan long line fishery. Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744), Aaron A. Thode (Scripps Inst. of Oceanogr., San Diego, CA 92093), Kendall Folkert (Alaska Longline Fishermen's Assoc., Stika, AK 99835), and Jan Straley (Univ. of Alaska, Southeast, Sitka, AK 99835)

The long line fishery of bottom dwelling black cod or sablefish has been plagued by depredation by sperm whales as the long lines attached to the array of hooks and caught fish are being hauled up by the fishing boat. The whales are probably attracted to a fishing boat by sounds associated with the boat being jockeyed into position so that the long line can be hauled up in a near vertical orientation. The sperm whales use their biosonar to localize and take the hooked black cods as they are being hauled up. In order to study the biosonar process, acoustic backscatter of several black cod specimens was measured using a simulated sperm whale biosonar signal having a peak frequency of 22 kHz as the specimens were rotated along both the lateral and dorsal planes. The backscatter of a rockfish (*Sebastes sp.*), a typical fishing hook, and a length of fishing line was also measured. The echo structure of black cod specimens allows sperm whales to discriminate between black cod, fishing hook, fishing lines, and by-caught rockfish. The target strength based on the incident and reflected energies varied from -27 to -37 dB for the black cod specimens.

3:10—3:30 Break

3:30

1pABb6. Assessment of specialized acoustic pingers to mitigate toothed whales depredation on Japanese tuna longline catches in the Central Pacific. Tom Nishida (Natl. Res. Inst. of Far Seas Fisheries, Fisheries Res. Agency, 5-7-1, Orido, Shimizu-Ward, Shizuoka-City, Shizuoka 424-8633, Japan) and Geoff McPherson (Eng. & Physical Sci., James Cook Univ., Cairns 4868, Australia)

Schemes to mitigate depredation by toothed whales (mainly killer and false killer whale) on industrial tuna longline operations have existed for over half a century in all major Oceans. We have assessed a range of avoidance and passive acoustic methods, some currently being further developed by industry. In 2010 using four longline vessels we tested the effectiveness of the newly developed dolphin deterrent device (DDD) pinger model suited for longline depredation and the dolphin interactive device (DiD) pinger manufactured by STM Products Inc. (Italy) in the high depredation area off south of Hawaii. Applying the experimental design, we investigated if DDD and DiD were statistically effective to reduce depredation. Preliminary assessment suggested that depredation rates on longline catches were probably reduced with DDD pingers and also with the interactive DiD pingers. Interactive and variable output mitigation pingers are important to prolong effectiveness as would be their capability to function with other mitigation techniques. Effectiveness of pingers in a fishery context may be based on economic significance (increased returns and reduced losses) as against statistical significance. [This project was funded by Fisheries Research Agency of Japan.]

3:45

1pABb7. Depredation of striped dolphin on squid fishery and behavioural responses to interactive pinger. Giuseppa Buscaino, Antonio Bellante, Gaspare Buffa, Francesco Filiciotto, Vincenzo Maccarrone, Vincenzo Di Stefano, Giorgio Tranchida, and Salvatore Mazzola (Inst. for Coastal Marine Environment-Capo Granitola, Natl. Res. Council of Italy)

The depredation of dolphins on some artisanal fisheries in the Mediterranean Sea is the major source of economic loss. This study aims to reveal the behavior of striped dolphin during interaction with flying squid fishery equipped with interactive pinger in the Ionian Sea (southern Italy). A four channel acoustics acquisition system was used during fishing hauls to record the clicks and to localize the positions of dolphins through the time delay of arrival method. The preliminary analysis shows that dolphins approached the artificial light, used for attract the squids, diving further on 100 m below the fishing boat. The number and the power of dolphins' clicks decreased after the signals emitted by the interactive pinger. The distances of dolphins from fishing boat do not seem change significantly before and after pinger emissions. Although the efficiency of DDD pingers to decrease the depredation level was demonstrated in some study, the functioning mechanism is still unknown. Our data could indicate that pinger works to reducing the dolphin's sonar activities.

4:00

1pABb8. Application of passive acoustic reflectors to mitigate toothed whale depredation on longlines. David M. Deveau (RMC Acoust. Eng., 37A Bliss Mine Rd., Middletown, RI 02842, dmdeveau62@gmail.com) and Geoff McPherson (James Cook Univ., Cairns Queensland 4868, Australia)

Indo-Pacific longline fisheries have noted for decades that entangled fish, particularly those with swallowed hooks or with elements of metal in the entangling gear, were invariably spared from depredation attacks. To enhance the capability of fishery objects to either highlight their presence as foreign or to interfere with the clarity of returning echoes, a range of sonar reflective material logistically acceptable to the fishing industry was modeled. Small alloy spheres were shown to have target strengths equivalent to that expected of the TS of an entire tuna from a rear approach, the normal approach on hooked swimming tuna during depredation events. The performance of reflectors is governed by the ratio of the dimensions of the object to the wavelength of the toothed whale broadband echo click. Known changes in hearing and sonar focus with increasing toothed whale age must be factored into a fishery based approach. Spheres developed for the Coral Sea were trialed by Japan Fisheries Research Agency. Logistic issues were a problem while other fisheries have expanded their commercial utilisation. Spheres with greater logistic simplicity and higher target strengths have been identified for further testing and if commercially constructed could be readily incorporated into fishing operations.

4:15

1pABb9. Depredation mitigation of bottlenose dolphins on coastal and oceanic set net and line based fisheries by dolphin dissuasive device and dolphin interactive dissuasor pingers. Alessandro Finezzo and Martin Ipuche (STM Products Inc., Via Morgagni 14, 37135 Verona, Italy, martin.ipuche@stm-products.com)

Depredation by marine mammals on coastal and oceanic net and line based fisheries is a major source of economic loss to fisheries where bycatch levels are low. To further minimize possible habituation to DDD pingers that feature variable (frequency and amplitude) signals an interactive model, the DiD, was designed. The DiD reacts to broadband frequency clicks comparable to delphinid clicks. The detection sensitivity is about 130 dB re 1 μ Pa at 1 m at the minimum threshold. The sensitivity curve matched peak output of dolphin clicks, particularly those of larger toothed whales responsible for depredation, achieving minimization of snapping shrimp interference in shallow water fishery applications. The DDD and DiD output's were changed to improve their effectiveness resulting in modulated tones and pseudo echolocation signals. Trials with dolphins in gillnet fisheries indicated DiD spacing could be 300 m. Spacing may change slightly with environment and fishery. DDD offers a reliable method for mitigating bycatch. DiD offers a choice to address long term effectiveness of pingers for depredation mitigation with limitation of power requirements, restriction of sound exposure in sensitive environments, and with benefits to industry and the environment. [Work supported by CNR, SeaMed, IAMC, Trapani, Italy.]

4:30

1pABb10. Acoustic pingers to mitigate dolphin bycatch and depredation, barely a one third octave between them. Geoff McPherson (School of Eng. & Physical Sci., James Cook Univ., Cairns 4868, Australia) and Neil Gribble (James Cook Univ., Cairns 4868, Australia)

Bycatch mitigation alarms/pingers for dolphins/porpoises and whales are developed in the mid 1980s in Japan and Canada, respectively. Alarms are attached to nets so the acoustic warning is associated with the obstruction, avoided, and the behavior reinforced via associative learning. A continuous association between alarms (the warning) and a net (the obstruction) is essential. What constitutes an appropriate alarm is not fully understood but should result in reduced entanglement in fishery conditions, irrespective of the mammals behavior observed by human observers. Dolphins are rarely deterred with bycatch pingers; they are alerted if they were inattentive but often maintain a close association with nets. Both bycatch and depredation may be reduced. These puzzling results should be investigated. At the other end of the pinger cline are the pingers that move dolphins from the vicinity of nets or lines to mitigate depredation, and by default, bycatch. The mechanisms are not known how these pingers can be successful at sound pressure levels comparable to dolphin whistles yet the results are becoming clear. The avoidance behavior from these pinger types should be investigated with captive animals. Interactive pingers and net material/reflectors with higher target strength are also seen as other important developments.

Session 1pAO

Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Ocean Observing Systems: Acoustical Observations and Applications II

Brian D. Dushaw, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Timothy D. Duda, Cochair
Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543-1053

Invited Paper

1:00

1pAO1. The Naval Postgraduate School Ocean Acoustic Observatory: Past experiences and current proposition. Ching-Sang Chiu, Christopher W. Miller, and John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Rm 328, Monterey, CA 93943, chiu@nps.edu)

The Naval Postgraduate School (NPS) Ocean Acoustic Observatory (OAO) utilized a cabled hydrophone array of the decommissioned Integrated Undersea Surveillance System (IUSS) Naval Facility at Point Sur, CA. Established in 1993, the NPS OAO became the prototype dual-use IUSS facility to release data to the education and research communities. The cable severed in January 2001 and was deemed unrepairable during a 2004 attempt. In this paper, scientific results from several of the research activities between 1993 and 2000 are presented. Some of these research projects were conducted in partnership with external academic institutions and some conducted solely by NPS investigators. Past research activities include signal propagation studies, acoustical thermometry and tomography, and ambient sound and whale monitoring. Since 2006, suboptimal monitoring efforts have continued using duty-cycled recordings from a moored hydrophone, lacking real-time access and continuity. Finally, a proposal to replace the existing terminal equipment building and install a new high-bandwidth cable terminating with an undersea distributed node seaward to enable emerging multifaceted, multidiscipline naval and academic research, developments, and applications is described. With strong endorsements by the NPS Dean of Research and Undersea Warfare Chair, among others, this dual-use plan has been submitted as a Military Construction project.

Contributed Papers

1:20

1pAO2. Interfacing a scientific echosounder with a cabled ocean observatory. David H. Barbee, John K. Horne, Samuel S. Urmy, and Richard B. Kreisberg (School of Aquatic and Fisheries Sci., Univ. of Washington, Box 355020, Seattle, WA 98195)

To demonstrate the utility of adding active acoustics to ocean observatories, an upward-looking, 38 kHz Simrad EK-60 echosounder package was connected to the Monterey Accelerated Research System (MARS) Observatory. The Deepwater Echo Integrating Marine Observatory System (DEIMOS) integrated the echosounder with commercial electrical and communications hardware in a glass pressure sphere, all mounted in an ROV-deployable frame. A cabled connection to the MARS node provided instrument power and real-time TCP/IP network communications to surface control and display systems. DEIMOS sampled the entire water column (875 m) at 0.2 Hz from February 27, 2009 to August 18, 2010. The instrument collected 50 Mbytes of raw data every 3 h resulting in a total of 167 Gbytes of raw data. Metrics developed to quantify distributions and abundances of pelagic organisms show temporal variability over seasons, diel cycles, and minute-to-minute predator-prey interactions. We will present design goals, deployment challenges, and operational experiences including *in situ* calibration and interactions with other instruments. Active acoustics is a vital and viable component of ocean observatories.

1:35

1pAO3. The Aloha Cabled Observatory. Lora Van Uffelen, Fred Duennebier, Roger Lukas, and Bruce Howe (Dept. of Ocean and Resources Eng., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96822, loravu@hawaii.edu)

20 months of deep-ocean acoustic spectra have been obtained from a single hydrophone 10 m above the ocean depth of 4720 m in an initial proof phase of the Aloha Cabled Observatory (ACO), the only abyssal ocean observatory in the world. A decommissioned telecommunications cable donated by AT&T provides power and communication to the observatory, which is located approximately 100-km north of Oahu, HI, at Station ALOHA, the field site of the Hawaii Ocean Time-series (HOT) program that has collected biological, physical, and chemical oceanographic data since 1988. The Aloha Cabled Observatory will be augmented in May 2011 with a general-purpose node outfitted with two hydrophones as well as CTD, ADCP, themistor, bio-optical, and visual sensors. The ambient acoustic data from the ACO hydrophone, which was collected from February 2007 to October 2008 contemporaneously with wind, wave, and seismic data, indicate that local wind speed is the most important parameter affecting background acoustic levels, and directional characteristics of the wind-driven ocean surface wave field also strongly influence the acoustic spectra. Correlation with seismic and wave spectra implies that wave reflections from coastal waters generate seismic energy that dominate the acoustic spectrum below 0.4 Hz.

1:50

1pAO4. Interpreted acoustic ocean observations from Argo floats. Jeffrey Nystuen (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Steve Riser, Tim Wen, and Dana Swift (Univ. of Washington, Seattle, WA)

While it is technically possible to record high bandwidth ambient noise, practically the ocean ambient sound data need to be subsampled and interpreted for users. A new configuration of Argo float has been augmented with passive aquatic listener (PAL) technology to monitor the ocean ambient sound during the drift phase of the Argo float mission, typically at 1000 m depth for 10 days at a time. These ocean ambient sound data are interpreted as wind speed, rainfall rate, and marine animal detections. These data are reported to users via two-way iridium satellite communication link when the float surfaces at the end of each dive cycle. Changes in sampling strategies can be implemented if desired. The first set of these floats has been deployed and additional deployments are ongoing. The data will be used to study the fresh water cycle of the ocean as part of the NASA Aquarius/SAC-D satellite mission scheduled to be launched this year.

2:05

1pAO5. Spatial characteristics of an Arctic ambient noise field. Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Thormoehlensgt. 47, N-5006 Bergen, Norway, hanne.sagen@nersc.no), Dag Tollefsen (Norwegian Defence Res. Establishment, N-3191 Horten, Norway), Stein Sandven (Nansen Environ. and Remote Sensing Ctr., N-5006 Bergen, Norway), and Elling Tveit (Norwegian Defence Res. Establishment, N-3191 Horten, Norway)

During multidisciplinary experiments in the 1980s and 1990s, ambient noise characteristics were found to be related to different ice types, distance from the ice edge, tidal current, wave conditions, and ice edge eddies. Primary noise generation mechanisms are ice floe collisions, ice cracking, ridging, and breakup caused by wave interaction with sea ice, internal ice stress, and changes in air temperature. Furthermore, sound propagation conditions determined by surface reflectivity due to sea ice and ocean stratification have a strong impact on the time averaged ambient noise spectrum. Ambient noise as a way to provide information about the environment is being explored in ongoing experiments within the ACOBAR and WIFAR projects. This paper presents spatial characteristics of the low-frequency ambient noise field from data collected simultaneously at 16 sites in the Marginal Ice Zone (Fram Strait) in October 2010. The acoustic measurements covered a 100 x 100 km^2 area from open water to within the ice cover at frequencies from 10 Hz to 2 kHz. Noise spectra will be presented and compared with earlier measurements from the same area.

2:20

1pAO6. Characterizing biological scatter before, during, and after a temporary ice retreat in the Bering Sea. Jennifer L. Miksis-Olds (Appl. Res. Lab., Penn State Univ., P.O. Box 30, Mailstop 6110D, State College, PA 16804, jlm91@psu.edu) and Joseph D. Warren (Stony Brook Univ., Southampton, NY 11968)

A three-frequency echosounder system was integrated into a NOAA oceanographic mooring in the central Bering Sea from 2008–2009. Diel vertical migration patterns were examined before, during, and after the formation and retreat of seasonal ice. Scattering aggregations identified in the year long time series were classified based on the differences in scattering amplitude between the three frequencies. Theoretical scattering curves for four different types of individual scatterers were generated and decibel-differences at the three acoustic frequencies used in this study were calculated for copepods with lengths of 1 and 8 mm, euphausiids with lengths of 15 and 30 mm, and organisms with a gas-inclusion such as a swim-bladdered fish or siphonophore. Aggregations were classified into one of the four categories by determining the shortest geometric distance between the 3 dB differences calculated for the aggregation and that of the theoretical scatterers. If the closest geometric distance exceeded 16 dB, the aggregation was classified as unknown. Results indicate that the vertical movement patterns and community structure of biological scatter during a 2 week temporary ice retreat more closely resemble that of the spring bloom than the period immediately prior to the seasonal ice presence. [Work supported by the ONR.]

2:35

1pAO7. A method for extracting ocean sound absorption from ambient noise measurements. Joel Paddock and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, joelpaddock@gmail.com)

This paper investigates a method for estimating the sound absorption of an ocean environment using passively measured ambient noise. Theory predicts that the level of absorption has a measurable effect on bottom loss estimation using ambient noise. An initial estimate of the absorption is obtained from this method, and is then refined through parametric optimization by comparing the measured data to a forward model of the ambient noise. The proposed method uses a passive sensor array which does not require extra sound energy to be projected into the ocean environment. Since the noise data consist of cross-correlations between sensors at individual frequencies, it is inherently spatially averaged so that the estimated absorption is potentially less sensitive to small scale fluctuations than methods that compare signal amplitudes over long propagation paths. The low frequency (less than 1 kHz) sound absorption mechanism of ocean water is dominated by the chemical relaxation of boric acid, the concentration of which is dependent on ocean *p*H levels. An estimate of ocean sound absorption could be valuable in tracking ocean acidity. The theoretical approach will be tested using simulated noise fields. Preliminary results using measured ambient noise will also be presented. [Work suppported by ONR Ocean Acoustics.]

Invited Paper

2:50

1pAO8. The acoustical components of the Arctic Ocean observing system. Hanne Sagen and Stein Sandven (Nansen Environ. and Remote Sensing Ctr., Thormoehlensgt 47, N-5006 Bergen, Norway)

The Arctic Ocean is severely under-sampled due to lack of regular observing systems. A future sustainable Arctic Ocean observing system will need to combine data from several underwater sensors with satellite data and models. During the international polar year (IPY) 2007–2009, new technologies for ocean observations such as acoustic thermometry/tomography, ice tethered buoys, floats, and gliders operating under the ice were developed and successfully tested. Based on experience from IPY, a future Arctic Ocean observing system should include a multi purpose acoustic network of source- and receiver moorings with capability to (1) measure the acoustic travel times to derive heat content and mean circulation on a regional or basin scale in minutes or hours, (2) provide an underwater GPS system for navigation and timing for under-ice Lagrangian systems, and (3) provide information about ice dynamics, earthquakes, and marine mammals through passive listening. Near-real time data from Arctic underwater systems can be improved by use of acoustic communication between underwater platforms and ice-tethered platforms with satellite communication. In the future cabled systems can enhance the capability to obtain near-real time data. The Svalbard Integrated Observing System will develop plans for an ocean observing system in the European sector of the Arctic.

3:10—3:35 Break

Contributed Papers

3:35

1pAO9. Exploring an under-ice ocean cavity with sound. Peter F. Worcester, Walter H. Munk, Bruce D. Cornuelle, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA 92093, pworcester@ucsd.edu), and Kathleen E. Wage (George Mason Univ., Fairfax, VA 22030)

Sensible predictions of the future rise in sea level require an understanding of the melting processes underneath the floating ice sheets of Antarctica and Greenland. Acoustic transmissions into the ocean caverns underlying the ice sheets could provide time series of temperatures and currents to augment intermittent measurements by instruments lowered through the ice and carried on autonomous underwater vehicles. The wedge of diminishing height formed between the underside of the ice and the shoreward sloping seafloor will horizontally refract sound transmitted into the ocean cavern by a source seaward of the ice edge. Incoming rays come to a coastal turning point near the grounding line of the ice sheet before returning seaward. A highly idealized model of vertical mode structure and horizontal ray structure provides estimates of the transmission paths in the wedge. In the ideal case, source-receiver separations of order 100 km for the broad Antarctic ice sheets are expected to provide good coastal returns, but separations of order 20 km for the narrow Greenland floating ice tongues are expected to result in severe reflection losses at the coastal turning points. Receivers lowered through the ice would provide valuable additional information. Further calculations using realistic propagation models are essential.

3:50

1pAO10. Acoustic thermometry of ocean climate and the global ocean observing system. Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105, dushaw@apl.washington.edu)

The uniqueness and sustainability of acoustic measurements of basin-scale temperature were demonstrated by the the Acoustic Thermometry of Ocean Climate and North Pacific Acoustic Laboratory programs in the North Pacific over the decade 1996–2006. Tomography has a role to play in the global observing system as a measurement type that complements altimetry and profiling floats. Tomography is a subsurface measurement of temperature; salinity contributes negligibly. Acoustic travel times are inherently precise integral measurements of temperature; mesoscale variability is supressed. Tomographic measurements offer one of the few ways to sample the abyssal ocean. The North Atlantic and Arctic have been highlighted by international conferences as regions that would be suitable for implementing tomographic arrays. Ocean basins can be measured by tomography using a few acoustic sources and receivers, employing platforms of opportunity such as existing or planned components of the ocean observing system. Extensive acoustic sampling can be achieved by sharing resources, while minimizing long-term operation and maintenance costs. Because of the integral nature of the data, tomography is best employed in conjunction with numerical ocean models and data assimilation. [Dushaw *et al.*, "a global ocean acoustic observing network," OceanObs'09: Sustained Ocean Observations and Information for Society, ESA Publication No. WPP-306].

Invited Papers

4:05

1pAO11. Ocean state estimation and basin-scale acoustic thermometry. Dimitris Menemenlis (Jet Propulsion Lab., California Inst. of Technol., Pasadena, CA 91109, menemenlis@jpl.nasa.gov)

This presentation describes a global ocean state estimation system and explores the role that sustained basin-scale acoustic thermometry can play in evaluating and constraining the resulting ocean state estimates. The state estimation system is that developed by the ECCO2 project. Solutions are based on an eddying, full-depth ocean, and sea ice configuration of the Massachusetts Institute of Technology general circulation model. Data constraints include altimetry, hydrography, sea surface temperature, and satellite observations of sea ice extent, thickness, and velocity. Green's functions and the adjoint method are used to adjust initial and surface boundary conditions and empirical parameters such as vertical diffusivity, albedos, and drag coefficients, in order to reduce model/data discrepancies. Early prototype basin-scale estimation systems that used acoustic data were deployed in the Eastern Mediterranean for THETIS-2 and in the North Pacific for ATOC. More recently, 1 decade of North Pacific acoustic thermometry data was compared with ocean simulation and estimation results. The comparisons with acoustic data provide stringent tests of the time mean hydrography and of the large-scale temperature variability in the models. The differences are sometimes substantial, indicating that acoustic thermometry data can provide significant additional constraints for numerical ocean models. Of particular interest is the deployment of a basin-scale acoustic array for monitoring changes in the deep ocean interior.

4:25

1pAO12. Comprehensive Nuclear Test Ban Treaty monitoring: The hydroacoustic network and its applications. Andrew M. Forbes and Georgios Haralabus (Comprehensive Nuclear Test Ban Organization, Vienna Int. Ctr., Wagramer Strasse 5, Vienna 1400, Austria, andrew.forbes@ctbto.org)

In the 14 years since the Comprehensive Nuclear Test Ban Treaty was signed, a network of 262 International Monitoring System (IMS) stations (mostly seismic) has been constructed out of a planned total of 337 facilities. 11 are hydroacoustic stations, of which six use hydrophones and five use T-phase seismometers. While the hydroacoustic network's primary purpose is to detect, locate, and identify underwater explosions, a secondary benefit is civil and scientific applications of the data. Given the ocean's remarkable transparency to low-frequency sound, it is not surprising that the hydrophone stations detect many other events and activities, e.g., marine mammal vocalizations, iceberg calving, underwater volcanoes, and seismic exploration. A variety of scientific studies have already used CTBTO hydroacoustic data, and the new virtual Data Exploitation Center (vDEC) is designed to facilitate and encourage more studies. Tsunami warning centers in eight countries around the world now receive real-time seismic and hydroacoustic data from the CTBTO. Future applications include ocean climate studies using acoustic thermometry, taking advantage of the permanency of the installed hydrophones and the repeatability of sources of opportunity, such as seismic exploration airguns. The challenge of sustaining the hydroacoustic network for decades must not be underestimated, however.

Session 1pBA

Biomedical Acoustics and Committee on Standards: Metrology of High Intensity Ultrasound

Peter J. Kaczkowski, Chair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Invited Papers

1:00

1pBA1. Challenges of clinical high intensity focused ultrasound: The need for metrology. Joo Ha Hwang (Dept. of Medicine, Univ. of Washington, 1959 NE Pacific St., Seattle, WA 98195 Jooha@medicine.washington.edu), Vera A. Khokhlova, and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105)

Metrology of high intensity focused ultrasound (HIFU) is critical to the advancement of clinical application of HIFU for safe and effective treatments in patients. Several methods for performing metrology of HIFU systems are available in the research laboratory setting; however, translation of these methods to the clinical setting remains in evolution. From our initial experience with clinical HIFU systems we have realized the importance of accurate acoustic characterization of HIFU systems in order to determine the parameters of the treatment protocol to result in safe and effective treatments. The acoustic parameters of the system, particularly at very high intensities, are very important to understand prior to delivering HIFU therapy to patients. Improved methods of HIFU metrology, especially to determine *in situ* exposure and dose, will result in a more rational approach to clinical HIFU therapy. Further advances in clinical HIFU therapy will require close cooperation between clinicians and scientists in order to make HIFU therapy safe and effective. Educating clinicians on the importance of metrology will also be important. [Work supported by NIH Grant No. EB007643.]

1:20

1pBA2. Evolution of metrology techniques for high intensity ultrasound. Mark E. Schafer (Sound Surgical Technologies, LLC, 357 S. McCaslin Blvd., Ste. 100, Louisville, CO 80027)

High intensity ultrasound device characterization presents a number of formidable measurement challenges. Whether the system is based on a high pressure, short temporal (i.e., shock wave), or a medium pressure, long temporal (i.e., HIFU) approach, the critical issue has been the destructive effects on the measurement sensor. Some shock wave systems have the additional issue of high shock to shock variability. This presentation reviews the evolution of the measurement and regulatory approaches used for characterizing high intensity devices. Every approach involves a compromise between the desire for a complete characterization of all possible parameters and the practical realities of making the measurements. Early work was done with very rugged sensors which provided nearly worthless data. The need for improved temporal and spatial resolution drove the development of a number of novel approaches including piezoelectric, force balance, and optical systems. Customized equipment has been developed for each particular type of HIU device. In some cases, measurements at low settings are used to predict results at higher settings. Improvements in measurement technology have often resulted in improved equipment designs, leading to better treatment efficacy and patient safety. However, there is still a gap between initial system characterization and day-to-day quality assurance needs.

1:40

1pBA3. Hydrophones for surface mapping of high-intensity transducers. George W. Keilman and Kyle P. Morrison (Sonic Concepts, Inc., 18804 North Creek Parkway Ste. 103, Bothell, WA 98011, gkeilman@sonicconcepts.com)

In order to fully characterize transducers used in high-intensity focused ultrasound, it is useful to measure the source field at the transducer's radiating surface while the transducer is radiating into water. A new, ruggedized hydrophone has been recently developed to enable direct-contact scanning at the radiating surface without damage to the source transducer. The hydrophone enables the acquisition of high-resolution images of the source fields for single-element and array transducers. This allows beam features such as unexpected side lobes or other anomalies to be traced back to local conditions at the radiating surface. The compact size of the hydrophone also allows source field measurements to be made in the presence of reflective structures located within the field of the transducer. This hydrophone is useful in the testing of transducer quality. In array transducers, it is also useful in verifying the geometric locations, radiating areas, and channel numbering of array elements, and in the measurement of channel-to-channel crosstalk.

2:00

1pBA4. Holographic reconstruction of therapeutic ultrasound sources. Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Oleg A. Sapozhnikov, Michael R. Bailey, Peter J. Kaczkowski, and Vera A. Khokhlova (Univ. of Washington, Seattle, WA 981050)

Clinical therapeutic ultrasound systems rely on the delivery of known acoustic pressures to treatment sites. Assessing the safety and efficacy of these systems relies upon characterization of ultrasound sources in order to determine the acoustic fields they produce and to understand performance changes over time. While direct hydrophone measurements of intense acoustic fields are possible, data acquisition throughout a treatment volume can be time-consuming and is only applicable to the specific source conditions tested. Moreover, measuring intense acoustic fields poses challenges for the hydrophone. An alternate approach combines low-amplitude pressure

measurements with modeling of the nonlinear pressure field at various transducer power levels. In this work, low-intensity measurements were acquired for several therapeutic transducers. Pressure amplitude and phase were measured on a plane near the test transducer; the Rayleigh integral was used to back-propagate the acoustic field and mathematically reconstruct relative vibrations of the transducer surface. Such holographic reconstructions identified the vibratory characteristics of different types of transducers, including a 256-element clinical array. These reconstructions can be used to define boundary conditions for modeling and to record characteristics of transducer performance. [Work supported by NIH EB007643, NSBRI through NASA NCC 9-58, and RFBR].

2:20

1pBA5. Measurement of focal peak intensity generated by high intensity focused ultrasound transducers. John Civale, Ian Rivens, and Gail ter Haar (Joint Dept. of Phys., Inst. of Cancer Res., Royal Marsden Hospital, Downs Rd., Sutton, Surrey SM2 5PT, United Kingdom)

High intensity focused ultrasound (HIFU) is a technique that is gaining widespread acceptance as a tumour treatment modality. Calibration of HIFU transducers poses a number of challenges, one of which is the need to measure the intensity generated at the focal peak. This quantity can be estimated by measuring the acoustic pressure using a calibrated hydrophone, but measurements are typically made at low drive powers in order to avoid damage to the sensor. An alternative is to use a radiation force or other techniques to measure the total power in the beam; estimation of the intensity at the focal peak is then ordinarily achieved with knowledge of the lateral 6 dB beam width together with assumptions about the shape of the focused beam. These underlying assumptions, however, may not be strictly valid, thus leading to overestimation of the focal peak intensity, particularly where the transducer contains a central hole for imaging. This paper will present results from simulations and hydrophone scans, which demonstrate the relationships between the spatially integrated (total power), averaged (−6 dB), and focal peak intensities. This information will be used to demonstrate possible strategies for determination of focal peak intensities from acoustic power measurements with improved confidence.

2:40

1pBA6. Full-diffraction and parabolic axisymmetric numerical models to characterize nonlinear ultrasound fields of two-dimensional therapeutic arrays. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, vera@apl.washington.edu), Petr V. Yuldashev, Mikhail V. Averiyanov (MSU, Moscow, Russia), Olga V. Bessonova (Physikalisch-Technische Bundesanstalt (PTB), Braunschweig, Germany), Oleg A. Sapozhnikov, and Michael R. Bailey (Univ. of Washington, Seattle, WA)

Numerical modeling has been shown to be an effective tool to characterize nonlinear pressure fields for single-element HIFU transducers, but it has not yet been applied for the much more complex three-dimensional (3-D) fields generated by therapeutic phased arrays. In this work, two approaches are presented to simulate nonlinear effects in the field of a 256-element focused array. A new full-diffraction approach includes rigorous 3-D simulations of the nonlinear wave equation with a boundary condition given at the elements of the array. A second simpler approach is based on the KZK model and a focused piston source as the boundary condition. The effective aperture and initial pressure of the piston source are set by matching linear simulations of the two models in the focal region. It is shown that as output power is increased, agreement in the focal waveforms of the two simulations, even when shocks were present, is maintained up to very high power outputs of the array. These results demonstrate the feasibility of using the simplified KZK model to evaluate the role of nonlinear effects in the fields of two-dimensional (2-D) phased arrays of clinical devices. [Work supported by NIH Grant Nos. EB007643 and RFBR 09-02-01530, and NSBRI through NASA Grant No. NCC 9-58.]

3:00—3:20 Break

3:20

1pBA7. Techniques for performing quantitative measurements of high intensity focused ultrasound intensity distributions noninvasively. Prasanna Hariharan, Dushyanth Giridhar, Ronald A. Robinson, and Matthew R. Myers (Food and Drug Administration, 10903 New Hampshire Ave., WO 62 RM2233, Silver Spring, MD 20993)

Estimation of the HIFU intensity field at clinically relevant power levels can be difficult due to the possibility of hydrophone damage or of sensor interference with the focused beam. To address this issue, we have developed two non-invasive methods for determining the intensity in free field and in tissue-mimicking material. In the first method, streaming field generated by the absorption of acoustic energy in liquid is measured using particle image velocimetry. The intensity distribution giving rise to the velocity field is computed by performing the operations of the Navier–Stokes equations upon the experimentally measured streaming field. The second technique (IR thermography) involves the use of an infrared camera to measure the temperature within a tissue-phantom. An air interface is required at the phantom boundary in order for the IR camera to observe the temperature rise. Quantitative determination of the intensity can be difficult due to the presence of the air interface (at which the intensity is zero), heat conduction within the phantom, and convection currents arising in the air. Mathematical relations that address these complications and allow for intensity measurements have been derived. Intensity fields obtained using both methods will be shown, along with comparisons with hydrophone measurements.

3:40

1pBA8. High intensity focused ultrasound characterization of deep bleeder acoustic coagulation cuffs. K. Michael Sekins (Siemens Medical Solutions, Ultrasound Div., 22010 S.E. 51st St., Issaquah, WA 98029-7298)

Research prototype therapeutic ultrasound “cuffs” have been recently constructed to demonstrate the feasibility of cauterizing bleeding (acoustic hemostasis) of deep limb wounds, such as those from military combat or other penetrating trauma. The deep bleeder acoustic coagulation (DBAC) cuffs required a variety of HIFU power and acoustic beam assessments to assure safety and efficacy. The cuff, capable of > 1000 W acoustic power, comprised multiple two-dimensional (2-D) HIFU arrays that surrounded the limb and could dose with multiple arrays simultaneously. Each array was capable of high power (>170 W), significant electronic steering (solid angle

= 57×34 deg), and depth focusing (5–20 cm). Device characterization required both individual array power measurements, and "integrated" whole cuff multiarray dosing assessment. The single array characterizations included (a) low-power beam plots, (b) high power focused power assessment in "apertured" radiation force balance tests, and (c) Schlieren beam pattern assessment. The integrated (whole cuff) behavior was evaluated using (1) integrated tissue mimicking material (TMM) phantoms replicating limbs, which included soft tissues, blood vessels, pulsatile arterial and "bleeder" blood flows, and were able to measure HIFU closed-loop targeting accuracy; and (2) TMM HIFU phantoms (measuring full power heating to 100 °C). [This research was sponsored by U.S. DARPA Contract No. HR0011-08-3-0004].

Contributed Papers

4:00

1pBA9. On acoustic absorption in tissue. Gregory Vilensky, Nader Saffari (Dept. of Mech. Eng., Univ. College London, Torrington Pl., London WC1E 7JE, United Kingdom), and Gail ter Haar (Inst. of Cancer Res., Sutton, Surrey SM2 5NG, United Kingdom)

This work extends the relaxation theory of sound absorption to the case of continuous distribution of relaxation times. Such an extension is needed when the absorption mechanisms are not confined to viscosity and heat conduction, but are due mainly to the excitation of a large number of internal molecular degrees of freedom. In this case, the conventional Navier–Stokes equations are not sufficient, and additional equations are required to model relaxation stresses. When relaxation frequencies form a sufficiently dense distribution, as is the case for many biological fluids, it makes sense to consider the limit of continuously distributed relaxation frequencies in order to obtain the required equation for normal stresses. The proposed theory provides a means of deriving the equations of motion in physical time-space from absorption data for a wide range of absorption laws that can occur in practice. A number of experimental absorption laws are considered in detail, and the related systems of equations of motion are derived and analyzed. The methodology developed in the literature based on modeling of absorption in biological soft tissue by fractional derivatives is a special case of the proposed theory. Numerical properties of different absorption models in tissue are also presented.

4:15

1pBA10. Understanding changes in tissue phantom material properties with temperature. Barbrina L. Dunmire, John C. Kucewicz, Stuart B. Mitchell, Larry A. Crum (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), and K. Michael Sekins (Siemens Medical Solutions, Issaquah, WA 98029)

Phantoms used for high intensity focused ultrasound (HIFU) applications require rigorous evaluation of material properties since, locally, the material experiences extreme changes in temperature and stresses with the HIFU treatment. Here we present the testing of an agar/gelatin phantom intended for both acoustic radiation force imaging (ARFI) and HIFU applications. The phantom shear modulus, speed of sound, attenuation, and thermal properties were all evaluated over the range of room temperature to 80 °C. With the exception of the thermal properties, all measurements were taken during both heating and cool down. Cavitation threshold and melting point were also tested. The change in material sound speed and thermal properties with temperature were quasireversible and similar to that of water. Material attenuation showed a slight decrease with temperature, but appeared to also be reversible. Shear modulus decreased significantly with temperature, going to near zero. The response was not reversible, returning to only approximately one-third of the starting value. These results demonstrate the complex material response that can occur with HIFU treatment. The results also raise the question of how well the test procedures, and thus results, properly reflect the true HIFU conditions. [This work was supported by the DARPA (Grant No. HR0011-08-3-0004).]

4:30

1pBA11. *In vitro* validation of three-dimensional cavitation-based pressure mapping for quality assessment of clinical high intensity focused ultrasound devices. Stuart Faragher, Jamie Collin (Dept. of Eng. Sci., BUBL, IBME, ORCB, Univ. of Oxford, Oxford OX3 7DQ, United Kingdom), Andrew Hurrell (Precision Acoust. Ltd., Dorchester DT2 8QH, United Kingdom), Paul Doust, Mark Tanner (DNV Ltd., Portland DT5 1SA, United Kingdom), and Constantin-C. Coussios (Univ. of Oxford, Oxford OX3 7DQ, United Kingdom)

Sufficiently robust and reliable quality assessment (QA) procedures are vital in assuring the widespread adoption of high intensity focused ultrasound (HIFU) for use in both thermal ablation and enhanced drug delivery. Mapping of broadband cavitation emissions in a tissue-mimicking material with a repeatable cavitation threshold offers the potential for rapid, 3-D, cavitation-based pressure mapping of the field produced by a given HIFU transducer. Previous work has demonstrated the viability of this concept, including the design and optimization of a 50-element, cylindrical array capable of mapping a collection of broadband sources distributed throughout a region comparable to the size of a typical HIFU focal volume. The work presented here relates to *in vitro* experimentation using the array to map cavitating fields produced by a number of HIFU transducers at a range of insonation amplitudes. Results are compared to the predicted size of the cavitation region determined via hydrophone-based characterizations of the HIFU field. Future work will involve using the array to map the instigation and evolution of cavitating fields in three-dimensions in *ex vivo* tissue during HIFU exposure.

4:45

1pBA12. Real-time acoustic dosimetry for focused ultrasound surgery. Jae Chun Jeon and Suk Wang Yoon (Dept. of Phys., SungKyunKwan Univ., Suwon 440-746, Republic of Korea)

Real-time acoustic dosimetry was investigated with acoustic attenuation and sound speed variation for focused ultrasound surgery (FUS). Bovine liver was used for the study because of its acoustic similarity to human liver. Acoustic attenuation and sound speed were measured *in vitro* with a pulse transmission method at the frequency of 1 MHz during FUS treatment on bovine liver. In bovine liver, acoustic attenuation and sound speed were increased with increasing temperature. Acoustic attenuation was steeply increased when the temperature of the tissue was increased around 55 °C. Once the tissue was necrotized, its acoustic attenuation was not returned to its initial value even though it was cooled down to the initial temperature, while its sound speed was almost returned to its initial value. This study indicates that acoustic dosimetry with acoustic attenuation can be very feasible for real-time FUS dosage control. [Work supported by Faculty Research Fund, Sungkyunkwan University 2010.]

Session 1pID

Interdisciplinary and Student Council: Introduction to Technical Committee Research and Activities, Especially for Students and First-Time Meeting Attendees

Lauren M. Ronsse, Cochair
Univ. of Nebraska, Lincoln, Architectural Engineering, 1110 S. 67th St., Omaha, NE 68182-0681

Dorea R. Ruggles, Cochair
Boston Univ., Biomedical Engineering, 44 Cummington St., Boston, MA 02215

Chair's Introduction—1:00

Invited Papers

1:05

1pID1. An overview of acoustic research in Speech Communication. Brad H. Story (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ 85721)

Speech Communication in the ASA covers a wide range of research areas and brings together scientists from many different academic disciplines. This presentation will provide a brief overview of acoustic research relevant to the production, perception, and acquisition of speech, speech processing, speech intelligibility, and disordered speech.

1:15

1pID2. Psychological and Physiological Acoustics: From sound to neurons to perception. Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E. River Rd., Minneapolis, MN 55455)

The area of psychological and physiological acoustics encompasses a wide and multidisciplinary range of topics. It is concerned with questions of what happens to sound once it enters the auditory system of humans and other species, and how sound is used to help organisms navigate their environment. Topics include the biomechanics of the middle and inner ear; the neuroscience of the auditory nerve, brainstem, and cortex; and behavioral and cognitive studies of auditory perception. This talk will give a brief overview of some of the many areas currently under investigation, ranging from basic questions about the neural codes used to represent aspects of sound to clinical applications, such as the development and improvement of cochlear, brainstem, and even midbrain implants that bypass the peripheral auditory system to restore hearing to people with profound hearing loss.

1:25

1pID3. Introduction to Animal Bioacoustics. David K. Mellinger (Oregon State Univ., & NOAA/PMEL, 2030 SE Marine Sci. Dr., Newport, OR 97365, david.mellinger@oregonstate.edu)

Animal bioacoustics (AB) covers all matters related to the production, transmission, and reception of sound in nature, as well as the investigation and use of natural sound by people and impacts of anthropogenic sounds by on animals. Topics include animal communication; sound production mechanisms; sound reception mechanisms (with P&P); evolution of sound production and hearing mechanisms; effects of acoustic propagation on natural sounds; sound detection, classification, localization and tracking; estimating populations and population density; impact of human-generated noise on animals; and a variety of other topics. All animals, and indeed all organisms, are considered within the scope of AB, though the most common taxa are marine mammals, birds, primates and other mammals, fishes, frogs and other amphibians, and insects.

1:35

1pID4. An introduction to the technical committee on Noise. Erica E. Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, erica.ryherd@me.gatech.edu)

Noise is generally defined as "unwanted sound," or more specifically sound that is "generally unpleasant," or "interferes with one's hearing of something," or "lacks agreeable musical quality." The demand for noise research and consulting continues to intensify in concert with rising population densities, increasingly industrialized societies, growing demands from consumers, and increasingly common standards and legislation related to noise. The Acoustical Society of America Technical Committee on Noise (TC Noise) is concerned with all aspects of noise, ranging from noise generation and propagation, to active and passive methods of controlling noise, to the effects of noise on humans and animals. Some specific interests include source mechanisms, predictions, measurements, evaluation, analysis, effects, regulation, mitigation, and legal aspects of noise.

1:45

1pID5. Overview of current research activities in architectural acoustics. Ralph T. Muehleisen (Civil, Architectural and Environ. Eng., Illinois Inst. of Technol., Chicago, IL 60616, muehleisen@iit.edu)

One of the largest and most active technical committees in the Acoustical Society of America is the Technical Committee on Architectural Acoustics (TCAA). TCAA typically has six to eight special sessions at each meeting. Authors include university researchers, product manufacturers, and practicing acoustical consultants. This talk will focus on reviewing research and activities of TCAA. Some of the more popular TCAA research areas include Green Building Acoustics, Healthcare Acoustics, Classroom Acoustics, Room Modeling, and Auralization. Some of the more popular activities include student paper and design competitions, the development of acoustical design booklets, and work on acoustics standards.

1:55

1pID6. The Musical Acoustics technical committee. Andrew C. Morrison (Dept. of Phys., DePaul Univ., 2219 N. Kenmore Ave., Chicago, IL 60614, amorri29@depaul.edu)

The technical committee on musical acoustics (TCMU) is concerned with the application of science and technology to the field of music. Of particular current interest are topics which include the physics of musical sound production, music perception and cognition, and the analysis and synthesis of musical sounds and composition. The TCMU draws from a diverse body of scientists, engineers, musicians, instrument builders, psychologists, architects, and others interested in the study of the science of music. An overview of selected research topics and activities of the TCMU will be presented.

2:05

1pID7. Structural Acoustics technical committee. Dean E. Capone (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16803)

An overview of the Structural Acoustics and Vibration Technical Committee (SAVTC) will be presented. The talk will provide a broad overview of the many research areas of interest to the SAVTC. In addition, a few topics will be explored in more depth to provide background on some of the more common analysis methods used by members of the technical committee. The content of the talk will be tailored to students and relative newcomers to the Acoustical Society of America.

2:15

1pID8. Introduction to the Engineering Acoustics technical committee. Daniel M. Warren (Knowles Electrons, 1151 Maplewood Dr., Itasca, IL 60143)

"Engineering acoustics encompasses the theory and practice of creating tools for investigating acoustical phenomena and applying knowledge of acoustics to practical utility"; thus opens the newly revised scope statement for the engineering acoustics TC, which was an occasion of some sole-searching in our group. The former scope had us pigeon-holed as "transducer guys." While transduction is an important aspect of our field, this limited definition undermined the true role of engineering as an art common across and necessary for all other scientific endeavors. Behind every scientific discovery is a rack of test gear, and in front is a practical application of that discovery. This is what we do; we are the toolmakers, the designers, and the architects of acoustics.

2:25

1pID9. An introduction to the Physical Acoustics technical committee's activities. Steven L. Garrett (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

The primary activity of any Technical Committee is to exploit the collective wisdom of the Committee's membership to determine which research topics within its specialization area are most active and interesting. Based on that assessment, the Committee organizes special sessions at future meetings that will bring together experts from those areas, not necessarily limited to the Society members, who can share interesting results and provide guidance regarding the directions that will lead to further understanding. In Physical Acoustics, that is a particularly daunting challenge given the scope of topics that fall within its purview: use of sound to probe material properties, sound propagation and attenuation mechanisms on this planet and in other parts of the universe, and physical effects of sound and its interaction with other forms of radiation, all of which could also go well beyond the limitations of a linear acoustical theory. At just this meeting, PATC is sponsoring special sessions on gas hydrates and on violent cavitation. Needless to say, involvement in debates about "what is hot" is both interesting and valuable. Other activities include proposals for technical initiatives that allocate ASA resources. Recently, PATC received funding to sponsor a demonstration session at the Physical Acoustics Summer School.

2:35

1pID10. Biomedical Acoustics: Making a quantum leap in medicine. Tyrone M. Porter (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston MA 02215, tmp@bu.edu)

At the dawn of a new era in medicine, the future of biomedical acoustics is bright. In the past century, we witnessed the introduction of ultrasound contrast agents, lithotripsy, and the visualization of ultrasound images in three dimensions. Currently, scientists are developing acoustic-based techniques for opening the blood-brain barrier transiently in order to treat brain tumors and neurological diseases. Additionally, researchers are developing echogenic liposomes and microbubbles for targeted ultrasound image enhancement and drug and gene delivery. Further excitement has been generated by the advances made with focused ultrasound to ablate or mechanically erode solid tumors in a noninvasive and site-specific manner. As technology and protocols continue to evolve, biomedical acoustics will have a dramatic impact on the diagnosis and treatment of debilitating diseases, thus improving patient care and quality of life.

2:45

1pID11. Introduction to Acoustical Oceanography research and activities. Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201)

It could be argued that in many ways, we have a better understanding of the universe beyond earth than about the oceans right here. The difficulty in exploring the ocean might be put in perspective by recalling how long it took to find the Titanic after years of searching. Why is it so difficult to obtain information about the ocean? Consider the many tools we use in our terrestrial lives for exploration and convenience. These include GPS, mobile phones, remote sensing images (e.g., Google Earth), and wireless internet to name a few. Due to unfavorable propagation conditions for electromagnetics in the ocean, the equivalent tools mostly depend on acoustic signals. Therefore, exploring the ocean is highly dependent on using acoustic systems to do, for example, remote sensing, imaging, and communications. Acoustical oceanography is essentially concerned with the development and use of acoustical techniques to measure and understand the many parameters and processes of the sea. This presentation will provide an introduction to acoustical oceanography and describe some of the current research and activities. Topics include acoustic applications for ocean observing systems, remote sensing with ocean noise, acoustic techniques for studying marine life, and inverse methods for ocean and seabed properties.

2:55

1pID12. An introduction to the Underwater Acoustics technical committee. Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Although sound speed was measured astonishingly accurately as early as 1826 by Colladon and Sturm in Lake Geneva, there was not widespread interest in underwater acoustics until the dramatic sinking of the Titanic in 1912. Today, acoustics is considered the best means of remote sensing in oceans, lakes, and estuaries due to the high attenuation of light in water. The members of the Underwater Acoustics Technical Committee are concerned with the generation and propagation of sound in an underwater environment as well as acoustic reflection and scattering from the seabed, sea surface, and objects in the water column and/or beneath the seabed. In this talk, a short history of underwater acoustics will be followed by an overview of the current state of research in the field.

3:05

1pID13. Overview of Signal Processing in Acoustics. David H. Chambers (Lawrence Livermore Natl. Lab., P.O. Box 808, L-154, Livermore, CA 94551)

Signal processing is used to some extent in all areas of acoustics such as extracting relevant information from acoustic measurements made either in the laboratory or in the field, processing signals, and/or synthesizing data to cope with demanding tasks raised in acoustics. Techniques range from simple classical approaches based on Fourier transforms and Gaussian noise to sophisticated model-based techniques that incorporate physical/parametric models of the acoustical system. In this talk we highlight new approaches to signal processing that could be applied to a broad variety of acoustical problems. Examples of each approach will be shown to illustrate the class of problems addressed and the performance. [This work was performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.]

MONDAY AFTERNOON, 23 MAY 2011 WILLOW B, 1:30 TO 5:00 P.M.

Session 1pNS

Noise and Education in Acoustics: Public Outreach Workshop on Community Noise

Lawrence S. Finegold, Cochair
Finegold & So, Consultants, 1167 Bournemouth Ct., Centerville, OH 45459-2647

George Luz, Cochair
Luz Social and Environmental Assoc., 4910 Crowson Ave., Baltimore, MD 21212

Chair's Introduction—1:30

Invited Papers

1:35

1pNS1. Community-based environmental noise management. Lawrence S. Finegold (1167 Bournemouth Ctr., Centerville, OH 45459, lsfinegold@earthlink.net)

Many local noise control initiatives in the past have been very successful in engineering terms, but community noise problems still exist, more needs to be done, and without concerted and sustained action these problems will continue to get worse. However, deciding on the most effective noise control options for communities is not just a matter of promulgating legally required noise exposure criteria and/or mandating the development of action plans to achieve these required noise levels. There are often multiple, conflicting perspectives among the various stakeholders involved in the resolution of community noise issues, which must be resolved to achieve effective solutions. This presentation provides a brief introduction to some of the concepts and techniques that can be utilized to fill this need. It will introduce the idea of community-based environmental noise management, which includes a "tool-kit" concept, which can be used by local communities in developing effective, acceptable, and affordable noise policies. Major components of this approach include land use planning and urban design, the use of a modern environmental impact assessment Process (EIAP), implementation of state and local noise ordinances and building codes, plus related technical tools and various support requirements.

1:55

1pNS2. Proposed standard-Guidance for Developing State Noise Regulations and Local Noise Ordinances. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Sq., Vernon, CT 06066 bbrooks@brooksacoustics.com)

The American National Standards Institute (ANSI) Accredited Standards Committee S12 (Noise) Working Group (WG) 41 has been developing a draft standards document for over 10 years. The current document is now in draft 8, currently under development. This document represents the consensus of many stakeholders in the community noise arena, including industry, government, consulting, and the public. The purpose of the document is to provide guidance to government officials, acoustical consultants, and other interested persons on how to develop a community noise ordinance or regulation, which is appropriate for the existing local circumstances. The document addresses issues such as public and government priorities and values, and available resources, and also provides the technical basis to manage the local sound environment. The keys to the effectiveness of the document are that it provides a menu of options for the user, discusses the trade-offs involved for decisions that must be made by government officials, and emphasizes that enforcement of a community noise ordinance is crucial to its success. A description of the current draft is presented.

2:15

1pNS3. A tutorial on noise-sensitivity. George Luz (Luz Social and Environ. Assoc., 4910 Crowson Ave., Baltimore, MD 21212, luz_assoc.@msn.com)

About one in five people can be categorized as "noise-sensitive." An individual's degree of noise-sensitivity appears to be a personal characteristic, not an acquired attitude. Because environmental noise assessments within the United States are based on average response to noise, the existence of noise-sensitive people can be a challenge for land-use planning around airports and other noise-generating facilities. Although there has been some research on noise sensitivity within the United States, the bulk of research has been conducted in Europe and Japan. The purpose of the tutorial is to review key findings from some of the most recent research on this subject.

2:35

1pNS4. Lessons learned in noise control the public sector. Nancy S. Timmerman (25 Upton St., Boston, MA 02118-1609, nstpe @hotmail.com)

The author spent 15 years as Noise Abatement Officer at Logan International Airport, installing a state-of-the-art Noise Monitoring System and answering complaints from the public about aircraft noise. The airport has been a leader in soundproofing for homes, in restricting noisy aircraft, and in providing information to the public about noise from aircraft operations. The author currently is a self-employed consultant, and recently testified about aircraft noise at a public hearing. This presentation will address the interrelationship between the (complaining) public, the technical consultants, and the government. It will reveal some uncomfortable truths about each sector, and warn about "being careful what you wish for."

2:55

1pNS5. Washington state noise laws, traffic, and finding a quieter medium. Tim Sexton and Jim Laughlin (Washington State DOT, 15700 Dayton Ave. N, Seattle, WA 98133, sextont@wsdot.wa.gov)

Washington State is actively involved with community noise, especially roadway noise, and is involved in developing outdoor alternative noise evaluation techniques and implementing community noise reduction strategies where applicable. In this presentation Tim Sexton and Jim Laughlin will discuss the current status of Washington State noise law, regulation, transportation policy, and various methods for approaching community concerns where the propagation of noise is difficult to model and even tougher to reduce. Examples include the Interstate 5 Ship Canal Bridge in Seattle, the Tacoma Narrows Bridge, quieter pavement research, construction noise in relation to local city and county noise ordinances, and preliminary work with local cities and counties on noise compatible land use.

3:15

1pNS6. Seattle Nightlife Advisory Board addresses the issue of urban core nightlife noise. Julie A. Wiebusch (The Greenbusch Group, Inc., 1900 West Nickerson St., Ste. 201, Seattle, WA 98119, juliew@greenbusch.com)

Urban Core neighborhoods are gaining in popularity. Infill or adaptive reuse projects are creating inner city housing and supporting a sustainable lifestyle. These neighborhoods have created walkable communities, with pedestrian-accessible amenities and proximity to the workplace. Nightlife is also often a component of an Urban Core community. Seattle has embraced expanding residential areas into the downtown core. The conflict between the residential desire for an environment that does not intrude on quieter living activities and the commercial desire to promote a vibrant nightlife within the city becomes more critical. In 2007, the Seattle Mayor signed an ordinance addressing amplified sound from nightlife. The Seattle City Council subsequently appointed a Nightlife Advisory Board (NAB)

to advise the city on a number of issues relating to nightlife and noise. The NAB consisted of three nightclub owners, three residents, a police officer, a representative from the Liquor Control Board, and a noise expert. NAB collaborated to reach consensus regarding allowable sound levels, protocol for measuring the sound, and proposed modified Ordinance language. The findings were presented to the City Council and are currently under consideration.

Contributed Paper

3:35

1pNS7. Traffic and construction noise changes in communities next to highways. Laura Musso Escude (WSDOT, Seattle, WA 98133)

Residents located next to an existing highway are getting notified that their road is getting either wider or fixed such as paving or installing a traffic signal. This paper summarizes the most frequent questions asked by residents located to nearby highways where construction will occur. Some of the questions that will be addressed will be as follows: Are we getting a wall, is the noise environment will change permanently or for few month or weeks, my routines are going to change, etc. A summary of the protocol on how the state addresses long term noise impacts and temporary ones is given. It will provide an overview on what happens when a noise study is warrant and what happens after if you have mitigation such as a wall. With regard to the temporary noise impacts, the paper will include the protocol on how the agency addresses the process before and during construction.

3:50—4:00 Break

4:00—5:00 Panel Discussion

MONDAY AFTERNOON, 23 MAY 2011 WILLOW A, 1:30 TO 4:30 P.M.

Session 1pPA

Physical Acoustics: General Topics in Nonlinear Acoustics

Philip S. Spoor, Chair
CFIC-QDrive, 302 10th St., Troy, NY 12180

Contributed Papers

1:30

1pPA1. A *k*-space method for nonlinear wave propagation. Yun Jing and Greg Clement (Dept. of Radiology, Harvard Med. School, Brigham and Women's Hospital, Boston, MA 02115)

A *k*-space method for nonlinear wave propagation in absorptive media is presented. The Westervelt equation is first transferred into *k*-space via Fourier transformation and is solved by a wave-vector time-domain scheme. The present approach is not limited to forward propagation or parabolic approximation. One- and 2-D problems are investigated to verify the method by comparing results to the finite element method. It is found that, in order to obtain accurate results in homogeneous media (errors of the fundamental and first two harmonics being less than 0.5 dB), the grid size can be as little as two points per wavelength, and for a moderately nonlinear problem, the Courant–Friedrichs–Lewy number can be as small as 0.4. As a result, the *k*-space method for nonlinear wave propagation is shown here to be computationally more efficient than the conventional finite element method or finite-difference time-domain method for the conditions studied here. However, although the present method is highly accurate for weakly inhomogeneous media, it is found to be less accurate for strongly inhomogeneous media.

1:45

1pPA2. Application of vortex identification to acoustic streaming flows. Mei Sou, Chittaranjan Ray (Dept. of Civil Eng., Univ. of Hawaii-Manoa, 2540 Dole St., Honolulu, HI 96822), Chris Layman, and John S. Allen, III (Univ. of Hawaii-Manoa, 2540 Dole St., Honolulu, HI 96822)

Acoustic streaming has an important role in a variety of industrial and medical ultrasound applications. Enhanced mixing and chemical processing can be achieved with the addition of ultrasound. Coherent structures in streaming flows, such as vortices, contribute to the overall transport and convective heat transfer. Their unsteady development is not well understood and thus the subject of on-going investigations. Previous studies measuring both acoustic heating and streaming have used thermocouples and been limited to specific spatial locations within the flow field. In this study, simultaneous velocity and temperature measurements are made using synchronized particle imaging velocimetry (PIV) and infrared thermography in a model sono-chemical reactor. This experimental system is water-filled acrylic tank with a 20 kHz Langevin acoustic horn mounted on the side. With overlapping fields of view for an IR camera and the PIV camera, the spatial-temporal evolution of the acoustic streaming and heating at the liquid surface is investigated. A vortex pair adjacent to the horn is identified with respect to its swirling strength and unsteady flow regimes are quantified in terms of the Lagrangian coherent structures (LCS) methodology. The swirling strength reveals the location of the vortex cores and LCS shows the evolving boundaries of the vortices.

2:00

1pPA3. Nonlinear scattering of crossed ultrasonic beams in a constricted flow for the detection of a simulated deep vein thrombosis. Sean M. Mock and Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402)

A three phase experiment performed at the USNA Acoustics Laboratory uses turbulent flow to generate nonlinear scattering of crossed ultrasonic beams. The intent is to simulate blood flow through a vein with and without a constriction to flow and compare the nonlinear scattering effects. In phase 1, a submerged turbulent water jet (Re$=4\times104$ and diameter$=0.6$ cm) interacts with overlapping crossed focused beams ($f_1=1.8$ MHz, $f_2=2.0$ MHz, and 15 cm focal lengths), 20 diam downstream, to create nonlinear scattering at the sum frequency ($f_+=f_1+f_2=3.8$ MHz). A circular plane receiving transducer rotates azimuthally in the plane of the transmitted beam axes about the intersection to obtain angular dependent Doppler shift (about f_+) and spectral broadening results. In phase 2, the turbulence at the meeting point is channeled into a simulated 30 cm long, 4.5 cm diameter "vein" with a thin, polyethylene membrane. Results show negligible nonlinearly scattered sum frequency, suggesting laminar flow. In phase 3, a constriction to flow is introduced via a 4.0 cm ping pong ball caught in a narrowed segment of vein. Results show a scattered sum frequency with narrow broadening, indicating the return of some vortices or turbulence.

2:15

1pPA4. Faraday waves produced by periodic substrates: Mimicking the alligator water dance. Peter Moriarty and R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215)

Male alligators use surface waves to attract their mates. This phenomenon is colloquially known as the alligator's water dance. Male alligators vibrate their lungs at very low frequencies, creating unique wavy patterns and fountains on the surface of the water. We hypothesize that the water dance phenomena are an expression of Faraday waves, arising from the instability of the plane fluid interface to wavy perturbations above a threshold vibration amplitude. We performed laboratory experiments to study how the corrugations covering the skin of an alligator affect the onset, efficiency, and pattern of Faraday waves. A three-dimensional (3-D) printed plate was modeled after a molding of an alligator's back, mimicking the corrugation size, shape, and spacing. An electromagnetic induction transducer was used to vibrate this model plate normal to the water surface. Frequency sweep data were collected for a variety of amplitude settings. Sweeps were performed for different plate depths, with and without corrugations. Three diagnostics were employed: a piezoelectric hydrophone, an out-of-plane laser vibrometer, and a high-speed video. Onset, frequency, and pattern data will be presented as a function of driving parameters, and as a function of depth. Implications of this work for an eventual field experiment will be discussed.

2:30

1pPA5. On the axial drift of free pistons in acoustic systems. Philip S. Spoor (CFIC-Q, Dr. 302, 10th St., Troy, NY 12180)

Unlike the pistons in car engines, or other devices with crankshafts and other mechanical linkages, the pistons in acoustic devices such as pressure-wave generators, thermoacoustic coolers, or Stirling engines are part of a resonant system. The motion is induced by the overall dynamics of the system, but is not rigidly proscribed. It is therefore possible, and exceedingly common, for these pistons to experience axial "drift" from their nominal rest position while running. This most often occurs in devices that use radial "clearance seals" in which the working fluid flows in a very narrow annulus between the piston and the piston cylinder. There is no guarantee that the mass of fluid that "blows by" one way during one-half of the cycle equals the the mass that blows by during the second half; and in general these devices experience net mass transfer through the clearance seal, resulting in a buildup of dc pressure across the piston. This dc pressure tends to force the piston away from its natural center, shortening its useful stroke. Through our work with clearance seal devices, we have learned how to quantify the processes that cause this piston drift, and how to passively suppress it.

2:45

1pPA6. Dissonant resonant modes of bottle-shaped thermoacoustics engines. Bonnie Andersen (Dept. of Phys., Utah Valley Univ., 800 W Univ. Pkwy, Orem, UT 84058)

A key component of a standing-wave thermoacoustic engine is the resonator. A simple half-wave resonator is not ideal, due to the loss of energy to the higher harmonics. Cylindrical resonators with regions with differing cross sections have been shown to have overtones that are not harmonics of the fundamental. For resonators composed of two adjoining concentric cylinders with two different radii, the half-wave resonant harmonics that would be present if the radii were equal are altered. In particular, it is shown that the cylinder with the smaller radius that opens into the larger cylinder has a quarter-wave resonator effect, while the larger-radius cylinder has a half-wave resonator effect. Graphs of the resonant frequencies, found numerically, plotted as a function of one geometric parameter of the resonator, show that the frequencies bend toward the asymptotes created by these quarter-wave and half-wave influences and that where the asymptotes cross is where the dissonance is the greatest. Experimental results follow the model presented.

3:00—3:15 Break

3:15

1pPA7. Intensive acoustic wave propagating in the stratified atmosphere and acoustical influence on the atmosphere state. Vladimir A. Gusev and Ruslan A. Zhostkov (Phys. Faculty, Lomonosov's Moscow State Univ., Leninsky Gory 1, 119991 Moscow, Russia, vgusev@bk.ru)

The analytical solutions describing intensive wave propagation in the stratified atmosphere are found. The exponential increase of amplitude due to stratification and change of effective viscousity are taken into account. The evolution of the initial periodic signal and the single impulse is investigated. The main attention is payed to the fine structure of shock front. The analysis of the qualitative features of evolution of both initial signals is made. The influence of the atmosphere temperature dependence on the height from Earth surface is investigated. It is shown that the strong temperature rise at large heights leads to the increase of distance of wave propagation. The obtained analytical solutions are used for estimation of intensive wave influence on the atmosphere state at large heights.

3:30

1pPA8. Modeling thunder propagation and detectability on Titan. Peter Achi and Andi Petculescu (Dept. of Phys., Univ. of Louisiana, P.O. Box 44210, Lafayette, LA 70504)

This research is part of a study investigating the characteristics of thunder on Saturn's largest moon, Titan. In tandem with electromagnetic signatures, thunder can corroborate and quantify lightning discharges. A physical model for the propagation of thunder on Titan, based on the most recent data collected by the Cassini–Huygens mission, is being developed. The model approximates a tortuous 20 km cloud-to-ground lightning channel by an angle-wise random walk of small discharge segments, each generating a strong cylindrical shock wave, which acquires an N-wave shape after it travels through the relaxation radius, into the acoustic regime. These acoustic waves are then propagated to the far-field detector where they are added linearly to form long-range thunder. The detectability of thunder signatures by a sensor in Titan's lower atmosphere depends on the moon's atmospheric structure. In order to constrain the fraction of acoustic energy reaching a detector in Titan's troposphere, the model accounts for the upward-refracting sound speed profile up to the inversion point at ~45 km and also for ground effects. The sound speed and attenuation are computed along the length of the lightning channel (20 km) using altitude-dependent pressure, density, and temperature measurements by Cassini–Huygens, as well as thermophysical parameters (specific heats, viscosity, thermal conductivity, and diffusivity) extracted from NIST's Chemistry WebBook.

3:45

1pPA9. Gasdynamic modeling of strong shock wave generation from lightning in Titan's troposphere. Christopher S. Hill and Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504)

In an effort to predict the characteristics of thunder on Titan, a model is being developed for the formation of the initial strong shock wave by a short cylindrical lightning discharge "segment." (Such small discharge segments are later used to synthesize a tortuous cloud-to-ground 20 Km-long lightning channel in Titan's troposphere.) The shock wave is obtained numerically as a solution of the coupled gasdynamic equations of state and conservation of momentum, mass, and energy. The relevant acoustic quantities are the pressure, density, particle velocity, and specific internal energy of the shock wave immediately following the discharge. Different scenarios for the initial deposition of energy from the discharge into the shock wave are investigated. The altitude-dependent ambient conditions in Titan's lower atmosphere—temperature, pressure, and density—are extracted from *Cassini-Huygens* data, while specific heats and transport coefficients of the main constituents of Titan's troposphere (N_2, CH_4) are obtained from the NIST Chemistry WebBook and interpolated at each altitude. The CH_4 molar fraction, measured by the mass spectrometer onboard *Huygens*, varies from 0.0492 at 5 m to 0.0162 at 35 km (the latter marking the lower end of Titan's tropopause). These parameters are used as inputs to the model for a complete thermodynamic characterization along the length of the lightning

channel. [The work was funded by the Louisiana Space Consortium (LaSpace) and NASA.]

The time of appearance of optical radiation following the initial firing of the laser is found to decrease with increasing external pressure as well.

4:00

1pPA10. Enhancement of the photoacoustic effect and sonoluminescence in carbon suspensions through exothermic chemical reaction. Han Jung Park and Gerald J. Diebold (Dept. of Chemistry, Brown Univ., Providence, RI 02912)

Generation of the photoacoustic effect through chemical reaction of particulate carbon in chemically reactive solutions is presented. Experiments are carried out using a *Q*-switched Nd:YAG laser to heat carbon nanoparticles, initiating chemical reaction which results in sound production. The amplitude of the photoacoustic signal from a carbon suspension in H_2O_2–H_2O mixtures is shown to increase dramatically as the percentage of H_2O_2 in solution increases. Laser induced chemical generation of sonoluminescence in aqueous carbon suspensions is also reported. Following irradiation and chemical reaction, highly compressed gas bubbles are formed. The expansion of the gas bubble past its equilibrium diameter results in oscillation of the bubble diameter. Tens of μs after the initial formation of the bubble, sonoluminescence is found to take place on collapse of the bubble. Experiments show that the sonoluminescent intensity is increased by a factor of more than 10 as the external pressure is increased from 1 to 6 atm.

4:15

1pPA11. Modeling of photoacoustic Raman spectroscopy. David H. Chambers and J. Chance Carter (Lawrence Livermore Natl. Lab., P.O. Box 808, Livermore, CA 94551)

Photoacoustic Raman spectroscopy (PARS) is a technique used to identify chemical species mixed in a gas or liquid based on their pattern of vibrational energy levels. Raman spectroscopy differs from the more familiar absorption spectroscopy by using a nonlinear two-photon process that can be more sensitive to small differences in vibrational energy levels. Thus, it can detect defect sites in solid-state optical materials or low concentrations of chemical species in gases. The Raman scattering process generates acoustic pulses that can be detected with a microphone. In this talk, we present an overview of PARS and present a model of the production of these acoustic pulses from the energy deposited in the medium during the Raman scattering process. We will discuss the different types of PARS (cw, pulsed, and resonant) and their relative advantages. Finally, we describe an experimental platform that we plan to use for validating the model. [This work performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract No. DE-AC52-07NA27344.]

MONDAY AFTERNOON, 23 MAY 2011 GRAND BALLROOM C, 1:00 TO 4:45 P.M.

Session 1pPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Advances in Auditory Development

Lynne A. Werner, Cochair
Univ. of Washington, Dept. of Speech and Hearing Sciences, 1417 N.E. 42nd St., Seattle, WA 98105-6246

Emily Buss, Cochair
Univ. of North Carolina, Dept. of Otolaryngology/HNS, 170 Manning Dr., Chapel Hill, NC 27599-7070

Chair's Introduction—1:00

Invited Papers

1:05

1pPP1. Looking back and hearing progress in the noise. Edwin W. Rubel (Bloedel Hearing Res. Ctr., Univ. of Washington, Seattle, WA 98195, rubel@uw.edu)

The past 40 years we have witnessed an explosion of knowledge on the development and plasticity of the hearing organ, and the central pathways subserving hearing. Studies of the periphery have advanced from light microscopic descriptions of morphologic and cellular changes toward a detailed understanding of the molecular interactions responsible for cell specification and dynamic emergence of mature structures. Physiological studies evolved from measures of gross electrical potentials toward understanding the emergence of transduction events and the molecules controlling sensitivity and filtering of mechanics and electrical properties. Developmental analyzes of the central auditory pathways largely changed from descriptions of changing morphology and physiology that approach the mature phenotype toward the search for genetic and cellular interactions that underly such changes and there has been a growing interest in defining the developmental constraints on emergence of a mature structural and functional phenotype, the so-called plasticity of the system: real and potential therapeutic advances ranging from cochlear implants to hair cell regeneration and the potential for prevention of hearing loss of genetic and environmental origin. In this presentation I will try to wade through the noise and select some of the key areas of progress toward understanding the substrates of hearing development.

1:45

1pPP2. Individual and age related differences of normal auditory development in older (6–12 years old) children. David Moore (MRC Inst. of Hearing Res., Univ. of Park, Nottingham, NG7 2RD)

The ascending, sub-cortical auditory system is anatomically and physiologically mature by 2–3 years of age, but immature hearing has been found in children up to at least 10 years. This immaturity typically takes the form of elevated detection and discrimination thresholds, and increased listener variability. But individual differences indicate adult-like performance in some younger children, suggesting that the sensory mechanisms for hearing are established relatively early. This and high response variability in other children also suggest that higher-level, cognitive aspects of hearing may be responsible for later immaturity. These predictions were examined in studies showing that standardized measures of cognition (non-verbal IQ, working memory, language, and literacy), parental reports of communication, and speech-in-noise perception all relate significantly to response threshold and variability on single tests of temporal (backward masking, 1 kHz frequency discrimination) and spectral [notch noise masking (NNM)] hearing. However, when the sensory and cognitive contributions to those measures were partially dissociated by comparing performance on paired tests (e.g., simultaneous masking and NNM), most or all poor performance appeared due to the cognitive demands of the test. These findings indicate that the later functional maturation of hearing in older children is due to cognitive rather than sensory development.

2:05

1pPP3. Effects of stimulus variability in auditory development. Emily Buss (Dept Otolaryngol./HNS, Univ. of North Carolina, 170 Manning Dr., Chapel Hill, NC 27599-7070), Lori J. Leibold, John H. Grose, and Joseph W. Hall (Univ. of North Carolina, Chapel Hill, NC 27599-7070)

Stimulus variability can affect the performance of child and adult listeners differently. For informational masking, stimulus variability appears to have a greater effect on children than adults in some conditions and a comparable effect in others. For intensity discrimination, children are *less* prone than adults to threshold elevation with the introduction of interval-by-interval stimulus level jitter. These results have been interpreted as reflecting different aspects of auditory processing. Informational masking in children is thought to reflect greater difficulty focusing on the signal cue. The intensity discrimination data have been interpreted as reflecting higher levels of internal noise and reduced sensitivity for intensity in children relative to adults. In the case of informational masking, supra-threshold masker variability may have a greater disruptive effect on signal processing, and in the case of intensity discrimination, variability near threshold has a smaller effect due to reduced sensitivity. Susceptibility to the disruptive effects of stimulus variability will be discussed as a means of gaining insight into the factors limiting auditory processing in children.

2:25

1pPP4. Developmental effects in susceptibility to remote-frequency masking. Lori J. Leibold (Dept. of Allied Health Sci., Univ. of North Carolina, CB 7190, Chapel Hill, NC 27599, leibold@med.unc.edu)

Although the peripheral auditory system accurately encodes the fine acoustic characteristics of sound by about 6 months of age, infants and children are more susceptible to auditory masking than adults under many conditions. In particular, large developmental effects have been observed in the presence of maskers remote in frequency from the target signal. Here, we review results of behavioral investigations of remote-frequency masking across infancy and childhood. Consistent findings have been observed across studies, suggesting that the sources of age-related changes in susceptibility to remote-frequency masking are related to the maturation of perceptual auditory skills such as sound source segregation and selective attention.

2:45—3:00 Break

3:00

1pPP5. Deaf infants' attention to speech after cochlear implantation. Derek M. Houston (Dept. of Otolaryngol., Indiana Univ., 699 Riley Hospital Dr., Indianapolis, IN 46202, dmhousto@indiana.edu)

[Cochlear implants can provide deaf infants access to auditory information and the opportunity to learn spoken language. However, providing infants with access to auditory information does not guarantee that they will actively attend to speech. Attending to speech may be important for early language development. Recent work with normal hearing infants suggests that infants attend to speech more than a similarly complex nonspeech signal at a very young age [Vouloumanos Werker (2004); (2007)], and recent models of infant speech perception propose that early attention to speech is crucial for speech perception development [Werker Curtin (2005)]. I will present work investigating deaf infants' attention to speech after cochlear implantation. We found that deaf infants' attention to speech, on average, increases during the first 6 months after cochlear implantation and that there is a high degree of variability in attention to speech among infants with cochlear implants. We also found that attention to speech 6 months after implantation predicted speech perception performance 2 and 4 years later. Taken together, these findings provide evidence that attention to speech is important for speech perception development and that attending to speech may be challenging for children who experience auditory deprivation during infancy.

3:20

1pPP6. A perceptual-learning deficit in adults with reading disorders interpreted as evidence of a developmental delay. Beverly A. Wright, Julia J. Huyck, and Carmen Aliyeva (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu)

Individuals with reading disorders have difficulty reading despite having normal intelligence. Many also exhibit a number of auditory perceptual deficits. According to one proposal, these deficits arise because perceptual development in these individuals is delayed in childhood and then halted, for skills on which improvements would normally continue into adolescence, around age 10 (presumably due to brain changes associated with puberty). If so, perceptual skills with long developmental courses should be impaired in adults with reading disorders. While perceptual development is typically examined through naive performance, this prediction was tested here using a perceptual-learning paradigm. Ten adults with reading disorders were trained on a temporal-interval discrimination task using a multiple-session training regimen known to yield learning reliably in adults, but rarely in adolescents [Huyck and Wright, Develop-

mental Science (in press). Seven of the ten did not improve with training. Thus, the response to training in the adults with reading disorders more closely resembled that of adolescent than of adult controls. This result is consistent with the idea that perceptual development may be delayed and then halted in individuals with reading disorders and highlights prolonged maturational changes in a perceptual skill other than naive performance.

3:40

1pPP7. Developmental changes in perceptual weighting and organization of speech. Susan Nittrouer (Otolaryngol., The Ohio State Univ., Eye and Ear Inst., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, nittrouer.1@osu.edu)

Speech perception research and theory historically have focused on the role of discrete spectro-temporal bits of the signal, known as acoustic cues, in phonetic labeling. Following suit, work in development and/or disorders has asked if sensitivity to these cues is diminished in children or disordered populations relative to typical adults. Although outcomes with disordered populations provide a controversial account, clearer is the conclusion that typical children have sufficient sensitivity to these cues from infancy. Yet equally clear is the finding that abilities to recover phonetic structure in one's first language emerge only gradually over the first decade or so of life, suggesting more may be involved in mature speech perception than has heretofore been considered. This presentation reviews what research has been done on developmental changes in three phenomena that might help explain that protracted period of development for speech perception: (1) weighting strategies for the numerous cues available for separate phonetic decisions, (2) signal structure that is temporally longer and spectrally broader than acoustic cues, termed "global" structure, and (3) perceptual organization of various kinds of signal structure. These findings should prompt innovation moving forward in how investigators examine developmental change in speech perception. [Work supported by NIDCD Grant DC-00633.]

Contributed Papers

4:00

1pPP8. Electrophysiology of infant speech feature detection and discrimination. Barbara Cone (Speech, Lang. and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85718, conewess@u.arizona.edu), Kristin Baker, and Jessie Liu (Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85718)

Cortical auditory evoked potentials (CAEPs) were obtained from awake, typically developing, normally-hearing infants, aged 4–10 months, in response to speech sound tokens. In experiment 1, 50-ms tokens of "Ling sounds" (/a/, /i/, /u/, /s/, /ʃ/, /m/) and 50-ms tone bursts at 500–4000 Hz were used as stimuli and presented at 10 dB level increments to develop CAEP latency and amplitude input-output functions. Observer-based psychophysical methods were used to determine detection thresholds for the tone bursts and speech tokens. Infant thresholds were elevated by more than 20 dB for tone bursts and by more than 30 dB for speech tokens with respect to adults. CAEP input-output functions in infants had significantly steeper slopes than those for adults. In experiment 2, 500-ms synthesized vowel tokens (/a/, /i/, /o/, /u/) were used to evoke CAEP components P1-N1-P2 and the acoustic change complex (ACC). Observer-based psychophysical methods were used to determine the infants' ability to discriminate between these vowel sounds. Robust ACCs were obtained for all vowel contrasts even when behavioral tests of discrimination were not successful. The results of these experiments will be discussed with respect to the development of the neural generators responsible for CAEP and perceptual abilities.

4:15

1pPP9. The time course of maturation on auditory tasks can differ between the average and consistency of performance and also between the sexes. Julia J. Huyck and Beverly A. Wright (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, julia.huyck@queensu.ca)

Between infancy and adulthood, performance on auditory perceptual tasks becomes both better on average and more consistent. However, little is known about the relationship between these two aspects of performance during development. This issue was investigated by using a cross-sectional design to estimate the ages at which measures of average performance (the mean of multiple threshold estimates for each listener) and performance consistency (the within-listener standard deviation of those estimates) became adultlike on two temporal-interval discrimination and four masking conditions (ages: 8–30 years, n=50–142 per condition). Average performance became mature before consistency on two simultaneous-masking conditions, after consistency on two nonsimultaneous masking conditions (backward and forward), and at the same time as consistency on the two temporal-interval discrimination conditions. In addition, on five of the six conditions, males matured more quickly than females on both measures or on consistency alone, but did not differ from females on either measure in childhood or adulthood. These results suggest that different processes may underlie the average and consistency of performance on auditory tasks and that transient sex differences in these processes could lead to a male advantage during adolescence on some of the everyday skills that rely on auditory perception. [Work supported by the NIH/NIDCD.]

4:30

1pPP10. Effect of reward in a new localization test method for children under 5. D. A. McCartney, C. J. Church, and B. U. Seeber (MRC Inst. of Hearing Res., Univ. Park, Nottingham NG72RD, United Kingdom)

Absolute localization ability in children under 5 has rarely been measured. One of the reasons for this is the difficulty of keeping the child engaged long enough in the task to obtain sufficient data. This was addressed by developing an uninstructed game-like localization task in which children must *find* the location of sounds. Correct localization responses, defined as a head or eye movement toward the sound's location, are reinforced with visual rewards. These are presented on video walls surrounding the child. This study investigated how varying the location of the visual reward affected the number of correct responses obtained. The study used three visual reward strategies: (I) the reward always presented at 0 deg (front), (II) the reward presented at the sound's location, and (III) the reward varied randomly $\pm$ 20 deg about the sound's location. Preliminary analysis of data from 24 children (15–56 months) shows a significant effect of reward strategy and age. More correct responses were generally obtained using reward strategies (II) and (III). Reward strategy (III) is preferred for use in a refined localization test because the reward is not presented at a fixed set of locations, helping to avoid visual learning of the test directions.

Session 1pSA

Structural Acoustics and Vibration, Noise, and Engineering Acoustics: Designing Quiet Composite Structures

Gopal Mathur, Cochair
Boeing, 5301 Bolsa Ave., Huntington Beach, CA 92647

Dan Palumbo, Cochair
NASA Langley Research Center, Hampton, VA 23681

Invited Papers

1:00

1pSA1. A review of noise control methods for composite structures. Gopal P. Mathur (Boeing Res. & Technol., The Boeing Co., Huntington Beach, CA 92647)

Composite materials and structures offer great promise to all air vehicles and are being increasingly considered for lightweight advanced applications. However, composite structures are efficient radiators of noise mainly due to excitation and propagation of supersonic bending and/or shear waves in the structures. Composite skin-stringer structures mostly support supersonic flexural or bending waves propagating in the laminated structure whereas noise radiation by sandwich structures is dominated by supersonic shear waves in the core. The vibro-acoustic behavior of sandwich composites has been studied by several investigators. Wave number diagrams have been useful in identifying supersonic flexural and shear waves in composite structures. It has been shown that the radiating wave number components increase with increasing plate stiffness, causing increased sound radiation. Although design criteria for sandwich composites in terms of sub-sonic core wave speeds have been evolved, skin-stringer composite structures must use added noise control treatments. Noise control of composite structures, thus, still presents a significant technical challenge as the main advantages afforded by these structures, such as low weight and high strength, must not be compromised by added treatments. This paper presents a review of recent advances in the noise control techniques for composite structures.

1:20

1pSA2. Panel contribution analysis for the interior region of a complex structure using the Helmholtz equation least squares method. Logesh Kumar Natarajan, Sandeep Mylavarapu, and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detoit, MI 48202)

The paper presents an experimental study on using Helmholtz equation least squares based nearfield acoustic holography to perform panel contribution analysis inside a complex structure with overall dimensions of $21 \times 14 \times 14$ in.^{3} in the shape of an automobile passenger compartment. A point random force was used to excite this structure. The radiated acoustic pressures were measured by a linear array of 13 microphones scanning over the interior surface at close distances, resulting in 520 measurement points. These measured acoustic pressures were used to reconstruct the vibro-acoustic responses, including the normal surface velocity (NSV) and surface acoustic pressures (SAPs) of the cabin. Next, the reconstructed NSV and SAP are used to determine the normal-component of the time-averaged acoustic intensity on the surface, which was then correlated to the sound pressure level at any field point, say, at the driver ear position inside the cavity. The relative contributions from individual panels toward a specific field point are calculated by summing the acoustic power flow from individual panels, and their ranking is determined. The major advantage of this approach is that panel contributions toward any number of field points can be determined based on a single set of measurements.

1:40

1pSA3. On the study of wave propagation and the sound radiation of structures. Cliff Chin (5301 Bolsa Ave., H45N-E405, Huntington Beach, CA 92647, cliff.l.chin@boeing.com)

In this study, the wave propagation of structures in relation to the sound radiation and the sound transmission of structures are examined with analytical and numerical approaches. The 1-D structures (i.e., beams) demonstrate a simple example of how the wave propagation in sub- and supersonic regions affect the overall sound radiation of the beam. The same concept is extended to more complicated two-dimensional (2-D) structures such as metallic, composite, and sandwich panels, and the wavenumber spectral approach will be utilized to understand the sound radiation qualitatively. The wave speed may not be a sole factor in determining the final radiation efficiency of the structures, but it certainly helps engineers with designing light-weight, mechanically strong, and efficient sound insulator panels for aircraft vehicles.

2:00

1pSA4. Acoustical characteristics of honeycomb sandwich composite panels. Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843, joekim@tamu.edu)

For the purpose of investigating acoustical characteristics of various honeycomb sandwich composite panels, finite element method (FEM) combined with boundary element method (BEM) has been widely used. However, the latter approach is not always applicable to a relatively high-frequency analysis since it is required to use a large number of FEM/BEM meshes resulting in a high computational cost. In order to reduce computational resources as well as modeling times, a hybrid analytical/finite element method (HAFEM) is developed that uses a finite element approximation in thickness direction while analytical solutions are assumed in the directions parallel to a panel surface. By applying the HAFEM, three acoustical characteristics of honeycomb sandwich composite panels are identified: i.e., sound transmission characteristics, structural wave propagation characteristics, and sound radiation characteristics. By comparing

measured sound transmission losses and predicted ones, the proposed HAFEM procedures are validated. It is shown that structural responses converge asymptotically to flexural waves in low-frequency region, core shear waves in mid-frequency region, and skin flexural waves in high-frequency region. It is also shown that the corner and edge radiation modes of finite size panels below critical frequencies can be visualized from predicted supersonic intensities.

2:20

1pSA5. Noise mitigation and wave number characterization in sandwich structures. James J. Sargianis and Jonghwan Suhr (Dept. of Mech. Eng., Univ. of Delaware, 126 Spencer Lab, Newark, DE 19716, james.sargianis@gmail.com)

Sandwich structures are the ideal materials to use for many engineering applications since they are lightweight, strong, and stiff. While sandwich structures are mechanically advanced, their acoustic properties are not ideal since they are excellent sound radiators. Therefore by studying their wave number characteristics, the acoustic properties of sandwich structures can be improved to help mitigate noise propagation. This study used several different combinations of carbon fiber-epoxy, glass fiber-epoxy, and aluminum face sheets with honeycomb and Rohacell foam core materials. These beams were vibrated with an electro-dynamic shaker, while the vibration response was measured at numerous equidistant points along the beam with a micro-accelerometer. Results showed the frequency response functions, which were transformed to the wave number/frequency domain yielding dispersion plots. From this plot one can calculate the coincidence frequency of the sandwich structure, which determines how and when the structure will radiate noise. Experimental results showed that thin aluminum face sheets with Rohcell 51 WF foam core had the greatest noise mitigation properties with a coincidence frequency of approximately 4500 Hz. Furthermore, it is shown that the experimental wave speed values agree with analytical methods. [Work supported by The Boeing Company.]

2:40

1pSA6. Vibroacoustics response of sandwich-composite panels with attached noise control materials. Noureddine Atalla (Dept. of Mech. Eng., Univ. de Sherbrooke, 2500 Blvd. de l'universite, Sherbrooke QC, J1K 2R1, Canada noureddine.atalla@usherbrooke.ca)

This paper discusses the modeling of the vibration and acoustic responses of sandwich composite panels with attached sound packages, using both analytical and numerical methods. Special attention is devoted to the modeling using the transfer matrix method of these structures in various mounting conditions (single wall and double wall) together with the calculation of various vibroacoustics indicators (vibration response, radiated power, transmission loss, added damping, air-borne insertion loss, structure-borne insertion loss, etc.) under various excitations (acoustical, mechanical, and turbulent boundary layers). Examples are presented to compare the performance of the presented models and demonstrate their range of applicability and usefulness.

3:00—3:15 Break

3:15

1pSA7. Designs for construction of quiet composite sandwich structures. Steven Nutt and Christina J. Naify (Dept. of Mater. Sci., Univ. of Southern California, Los Angeles, CA 90089)

Sandwich structures are comprised of a low-density core sandwiched between two thin, stiff facesheets, often composite laminates. The construction of the sandwich structure is designed to optimize the high-bending stiffness-to-weight ratio necessary for aerospace applications. Although the high stiffness-to-weight ratio of sandwich structures is ideal for bearing mechanical loads, a trade-off exists between mechanical and acoustic performance, in that increases in one generally come at the expense of the other. Improvement of TL of sandwich panels can be divided into three general categories: altering the panel construction by changing the core/facesheet materials, use of active or semi-active vibration suppression, and finally, added treatments such as foams, fibers, or other structures to either the exterior of the panel, or inside the core. TL of sandwich structures using these strategies is presented, in addition to explanation of the physical principals responsible for the decrease in transmitted sound. Featured approaches include design of core and facesheet materials, attachment of layered gasses to the panel facesheet, and insertion of midplane membrane-type metamaterial to the core. Advantages and disadvantages of each strategy are also considered and discussed.

3:35

1pSA8. Novel sandwich composite aircraft sidewall architectures. Dan Palumbo (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, d.l.palumbo@nasa.gov)

An aircraft sidewall constructed of honeycomb sandwich composite materials could have a mass per unit area, which is 60% less than a comparable stiffened aluminum structure. With reduced weight, the sandwich composites have a high stiffness to mass ratio which yields high radiation efficiency with the result that the sandwich composite can have a transmission loss up to 10 dB less than its aluminum counterpart. A lightweight sandwich composite fuselage will then produce higher cabin noise levels and require additional treatment. The initial weight savings gained through the use of the composite material would be compromised by the need for the additional treatment. It may be possible to the reduce radiation efficiency of sandwich composites through novel features machined into the material's core. To explore this possibility, several Nomex honeycomb, sandwich composite panels were designed constructed and tested. The performance of the composite panels in terms of both transmission loss and vibration response will be compared to that of a stiffened aluminum panel. It will be shown that it is possible to recover a good portion of the transmission loss reduction incurred with respect to the aluminum structure without sacrificing panel strength.

3:55

1pSA9. Effect of surface treatment of graphite nanoplatelet on damping properties of polyetherimide nanocomposites. Bin Li, Erik Olson, and Weihong Zhong (School of Mech. and Mater. Eng., Washington State Univ., 405 NE Spokane St., Pullman, WA 99164, bin.li@email.wsu.edu)

Polyetherimide (PEI) with excellent mechanical properties and thermal stability is gaining popularity as infrastructure materials in aircraft and ground transportation. Exfoliated graphite nanoplatelet (GNPs) composed of stacked 2-D graphene layer, have shown great potential as a damping property modifier, owing to their high mechanical performances and strong interlayer frictional motion. Interface bonding between the nanofiller and the polymer matrix is critical to the properties of polymer nanocomposites. This paper reports the investigation on PEI/GNP nanocomposites for their dynamic mechanical properties, specifically to fabricate polymer composites with superior damping properties. In order to determine the effects of interfacial bonding between PEI and GNPs, the GNPs were surface treated by different amounts of silane ([3-(2-aminoethylamino) propyl] triethoxysilane. Preliminary results indicated that adding both types of GNP with and without the silanization treatment does increase the storage modulus of PEI, but surface treated GNP shows much better efficiency in improving the damping capability, compared with pristine GNP, especially at high dosage of silane surfactant.

Contributed Papers

4:15

1pSA10. Predicting response of a honeycomb sandwich panel to diffuse acoustic field vs turbulent boundary layer excitation using coupled finite element-boundary element approach. Reza Madjlesi (Bombardier Aerosp., 123 Garratt Boulevard, Toronto, ON, Canada, reza.madjlesi@aero.bombardier.com) and Noureddine Atalla (Univ. of Sherbrooke 2500, Univ. Boulevard, Sherbrooke J1K 2R1, Canada)

Composite materials are being extensively used in primary and secondary aerospace structures due to high specific strength and stiffness as well as low weight. In this study finite element (FE) and boundary element (BE) are used to study acoustical performance of honeycomb type structures subjected to common excitation sources in flight. Coupled FE-BE is applied to predict transmission loss (TL) of honeycomb type structure, excited by diffuse acoustic field and turbulent boundary layer. Structural FE model correlation is performed using experimental modal analysis. Transmission loss of honeycomb panel was measured at the GAUS TL Lab using diffuse acoustic field. TL results are used to validate coupled FE-BE vibro-acoustic model of a curved honeycomb panel. Correlated model is used to predict response of structure to TBL source. Effect of noise control treatments in reducing radiated noise from honeycomb panel excited by TBL and diffuse field are studied.

4:30

1pSA11. Acoustic analysis of honeycomb sandwich structures with perforated septa and membrane-type acoustic metamaterials. Christina J. Naify (Dept. of Mater. Sci. Univ. of Southern California, Los Angeles, CA 90089), Chia-Ming Chang, Geoffery McKnight (HRL Labs., 3011 Malibu Canyon Rd, Malibu, CA 90265-4797), and Steven Nutt (Univ. of Southern California, Los Angeles, CA 90089)

Composite structures, which are commonly used in aerospace applications due to their high strength and low weight, tend to exhibit poor acoustic properties. Sandwich structures are constructed of a low-density core adhered to two thin stiff facesheets. One common assembly for sandwich panels is a divided honeycomb core made of either para-aramid paper or aluminum with fiber-reinforced composite facesheets. Typical acoustic treatments for composite sandwich structures have included the addition of foams and fibers to either the exterior of the structure or inside the core. In addition, a currently implemented treatment involves the insertion of perforated septa to the midplane of the honeycomb core, though this method's effectiveness has not been established. Membrane-type acoustic metamaterials have been shown to increase transmission loss (TL) by up to 500% over the acoustic mass law, with multiple TL peaks achieved in a multicelled array. In this study, experimentally obtained TL behavior of a sandwich panel with a perforated mid-core septa is compared to a panel with a mid-core layer made of the membrane-type metamaterial. FEA generated predictions were used to validate the experimental data.

4:45

1pSA12. Cell size effects on material properties of foam-filled honeycomb sandwich structures using finite element analysis. Zhuang Li (Dept. of Eng., McNeese State Univ., Lake Charles, LA 70609, zli@mcneese.edu)

In recent years, sandwich structures have become used increasingly in many applications, especially in transportation vehicles, due to their high strength-to-weight ratio, excellent thermal insulation, and good performance as water and vapor barriers. The great advantage of sandwich structures is that optimal designs can be obtained for different applications by choosing different materials and geometric configurations of the face sheets and cores. In previous research work, the foam-filled honeycomb cores demonstrate higher stiffness and vibration damping than pure honeycomb and pure foam cores. However, other properties of the foam-filled honeycomb cores have not been investigated thoroughly. In this paper, the effects of honeycomb cell size on both Young's modulus and the shear modulus of the foam-filled honeycomb core were analyzed using finite element models developed in ANSYS. The polyurethane foam may produce a negative Poisson's ratio by the use of a special microstructure design. The influence of Poisson's ratio on the material properties is also presented.

5:00

1pSA13. Analysis of sound transmission through sandwich panels. Zhuang Li (Dept. of Eng., McNeese State Univ., Lake Charles, LA 70609, zli@mcneese.edu)

Due to their advantages over the conventional metal structure, sandwich structures have become used increasingly in transportation vehicles in recent years. However, they have some less favorable properties. In this paper, sound transmission through foam-filled honeycomb sandwich panels was studied using the statistical energy analysis (SEA). Modal density, critical frequency, and the radiation efficiency of sandwich panels were analyzed. The sound transmission loss and radiation efficiency of sandwich panels were predicted theoretically and measured experimentally. The measured results agree with the theoretical predictions quite well. An important observation is that, for foam-filled honeycomb sandwich panels, the radiation efficiency does not decrease to a plateau value at frequencies beyond the critical frequency. Although the high damping helps to improve the sound transmission loss in the critical frequency range, the overall transmission loss of sandwich panels is lower than their heavy metal counterparts. Therefore, the foam-filled honeycomb design must be modified to obtain good noise isolation capability, and a trade-off between the overall stiffness and the sound transmission properties should be considered in the design stage.

Session 1pSC

Speech Communication: Speech Perception (Poster Session)

Molly Babel, Chair
Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

1pSC1. An effect of phoneme frequency on stop place perception by English-speaking and Japanese-speaking listeners. Kiyoko Yoneyama (Dept. of English, Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi, Tokyo 175-8571, Japan, yoneyama@ic.daito.ac.jp), Keith Johnson, and Reiko Kataoka (Univ. of California, Berkeley, CA 94720-2650)

The effects of phoneme frequency on stop place perception were examined. In English, [t] is more frequently observed than [k] while the opposite is true in Japanese. If the sound frequency affects the phoneme perception, English listeners would identify more of the ambiguous [t]-[k] stimuli as "t" than do Japanese listeners. In this study, a 4IAX discrimination experiment and a phoneme boundary decision experiment were conducted with English-speaking and Japanese-speaking listeners to test this hypothesis. The stimuli of the two experiments were created by the Klatt synthesizer. [ta], [ka], [to], and [ko] were recorded by a native Japanese speaker and a native English speaker, respectively. Then, [ta], [ka], [to], and [ko] were synthesized based on the parameter values obtained from the measurements of the recorded tokens. They were used as the end-point tokens of a nine-step continuum. The rest of the tokens were created by manipulating frequency parameters of the synthesizer. Four sets of [k]-[t] continua were created (two languages: English and Japanese, and two vowel conditions: [a] and [o]). A preliminary analysis of a subset of data demonstrated that the more frequent sound in each language occupied a larger perceptual region of the [a] vowel continua.

1pSC2. The productivity of structure-dependent tone sandhi in Shanghai Chinese. Jie Zhang and Yuanliang Meng (Dept. of Linguist., The Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045, zhang@ku.edu)

Shanghai Chinese is noted for its structure-dependent tone sandhi patterns: its five lexical tones undergo two types of tone sandhi: a word with a modifier-noun structure undergoes a left-dominant sandhi, whereby the initial tone is spread across the entire word; a word with a verb-noun structure, however, may undergo a right-dominant sandhi that reduces the tonal contours on nonfinal syllables, but preserves the tone on the final syllable, especially if the word is of low frequency. A wug test was conducted to investigate the productivity of the two types of sandhis. Four types of disyllabic words were tested for each underlying tonal combination: real words, novel words composed of two legal syllables, and novel words in which either the first or the second syllable was a non-existing syllable in Shanghai. The syntactic structures of the words were cued by carrier sentences. 48 speakers of Shanghai (20 M, 28 F) participated in the experiment. The f0 results for the experimental stimuli will be reported and discussed in the context of the effect of structure dependency on tone sandhi productivity and how this effect interacts with those of lexical frequency and the phonetic properties of the sandhis.

1pSC3. The interlanguage speech intelligibility benefit-listeners for Chinese accented English liquids. Joo-Kyeong Lee and Xiaojiao Xue (Dept. of English Lang. and Lit., Univ. of Seoul, 90 Jeonnong-dong, Dongdaemun-gu, Seoul 130-743, Korea)

This study investigated the intelligibility of Chinese-accented English liquids for native English, Chinese, and Korean listeners. The lateral /l/ and the retroflex /r/ in syllable initial and final positions were examined in three different contexts: segment, word, and sentence. Three groups of listeners with different language backgrounds accomplished a forced-choice identification task. For Chinese talkers' production of the English liquids, both matched and mismatched interlanguage speech intelligibility benefits for listeners (ISIB-L) occurred in all the four positions except for the initial /l/ produced by low proficiency Chinese talkers. The ISIB-L, however, held only for high proficiency non-native listeners. The only exception was the syllable initial /l/ stimuli produced by low proficiency Chinese talkers in such a way that native English listeners were more accurate than Chinese and Korean listeners at identifying the strong Chinese accented syllable initial /l/ sound. Considering the contextual effects on ISIB-L, matched ISIB-L only occurred for high proficiency listeners in the segment context. There was evidence of both matched and mismatched ISIB-L effects only for high proficiency listeners in word context. In the sentence context, English listeners performed better than all non-native listeners; therefore, there was no evidence for ISIB-L.

1pSC4. Gradient perception of laryngeal contrast of stops in English and Korean: Eye-tracking evidence. Eun Jong Kong (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ekong@wisc.edu) and Jan Edwards (Univ. of Wisconsin-Madison, Madison, WI 53706)

Categorical perception implies that listeners discard subphonemic acoustic variation and attend only to higher-level phonemic representations. However, a number of studies have shown that listeners are also sensitive to subphonemic fine phonetic detail, given the appropriate task. The current study aimed to investigate gradient perception of the stop voicing contrast using an anticipatory eye movement paradigm. Specifically, our study examined English- and Korean-speaking adults' sensitivity to changes in VOT and $f0$ parameters that differentiated their native stop voicing categories. The stimuli varied VOT values in six steps (9–59-ms) and $f0$ in five steps (98–114 Hz) in synthesized CV syllables, with a vocalic source from either a male speaker of English or Korean. Adult listeners' anticipatory eye movements to the target categories were collected, as they listened to a series of stimuli. Looking latencies were assessed in a mixed-effect regression model. Preliminary results found gradient responses to VOT changes for both English and Korean speakers as well as evidence of sensitivity to $f0$ for Korean, but not English speakers. Results are also presented for bilingual English-

Korean speakers, who were asked to perform the task in both English and Korean. [Work supported by NIDCD Grant No. 02932 and NSF Grant No. 0729140]

1pSC5. Production and perception of voice onset time cues in spoken Japanese and Taiwan Mandarin. Naomi Ogasawara (Dept. of English, Natl. Taiwan Normal Univ., No. 162, Sec. 1, He-Ping E. Rd., Taipei, Taiwan, 106 Republic of China, naomi703@ntnu.edu.tw)

This study investigates (1) VOT length of Japanese and Taiwan Mandarin (TM) stops in word-initial and word-medial positions of disyllabic words and (2) VOT boundary between voiceless and voiced stops in word-initial and word-medial positions of Japanese disyllabic words perceived by TM listeners. Japanese has voiceless and voiced distinction of stops, while Mandarin has aspirated and unaspirated distinction of stops. This discrepancy makes it hard for TM listeners to perceive Japanese stops correctly. In addition, based on my observation, stops in word-medial position seem to be more difficult for them to hear. In the first part of the study, an acoustic analysis is conducted to investigate the difference in VOT length of stops of the two languages. Perception experiments are conducted to examine the relationship between VOT boundary and the positions of stops in words. Groups of TM listeners at different Japanese proficiency levels participate in the experiments in order to test the effect of second language acquisition.

1pSC6. Assessing the relative importance of consonants and vowels in Spanish and English. Michael E. McAuliffe (UBC Dept. of Linguist., Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada)

There appears to be a systematic difference in the dialectal variation of phonemes within languages. In his treatment of English dialects, Wells [(1982)] had three volumes detailing the vocalic differences of the dialects of English, but he remarked that these dialects "do not differ very greatly in their consonant systems" (1982:178). Conversely, Hualde [(2005)] detailed the dialectal variation of each consonant in detail, but only sad of vocalic variation that "vowel qualities are remarkably stable among Spanish dialects" (2005:128). These differences in variation suggest that English values consonants as more integral to a word, and Spanish values vowels more, as those are the more stable segments in their respective languages. To empirically test this observation, a lexical decision task [Goldinger (1996)] is employed, with native speakers of both English and Spanish, using nonwords varying from attested words either in a single consonant (such as *ship* versus *shid* in English) or a single vowel (such as *ship* versus *shup*). The predicted outcome is that languages that value consonants, such as English, will make the decision faster when the consonant is changed, and languages that value vowels, such as Spanish, will make the decision faster when the vowel is changed.

1pSC7. Perception of natural vowels by monolingual Canadian-English, Peninsular-Spanish, and Mexican-Spanish listeners. Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., Schl. of Elec. Eng. & Telecom., Univ. of New South Wales, Sydney, Australia, geoff-morrison@forensic-voice-comparison.net)

Based on a previously reported synthetic-vowel perception experiment, it was hypothesized that the location of the perceptual boundary between Spanish /i/ and /e/ differed for monolingual Peninsular-Spanish and Mexican-Spanish listeners (North-Central Spain and Mexico City), and this would affect the perception of the Canadian-English /i/—/ɪ/ contrast (Western Canada): Peninsular-Spanish listeners were predicted to identify almost all tokens of Canadian-English /i/ as Spanish /i/ and almost all tokens of Canadian-English /ɪ/ as Spanish /e/ (two-category assimilation), whereas Mexican-Spanish listeners were predicted to identify almost all tokens of Canadian-English /i/ as Spanish /i/, but identify some tokens of Canadian-English /ɪ/ as Spanish /i/ and some as Spanish /e/. Monolingual Peninsular-Spanish and Mexican-Spanish listeners' perception of natural tokens of English /i/, /ɪ/, /e/, and /ɛ/ produced by monolingual Canadian-English speakers was tested. Both the Peninsular-Spanish and the Mexican-Spanish listeners had results consistent with the perceptual pattern predicted for the Peninsular-Spanish listeners. Implications for second-language speech perception are discussed.

1pSC8. The discrimination and categorization of German /r/ allophones by American English speakers. Dilara Tepeli (Dept. of Communicative Disord., Goodnight Hall, 1975 Willow Dr., Madison, WI 53706, dtepeli@wisc.edu)

This study investigates the discrimination and categorization of two German /r/ allophones—the uvular fricative [R'] and the uvular trill [R]—by experienced and naive American English (AE) speakers. Two groups of AE speakers participated in a categorization experiment followed by a discrimination experiment. Preliminary results suggest that experienced AE speakers perceive both variants predominantly as a German /r/ sound, followed by an English /r/ sound. Inexperienced AE speakers also predominantly perceive both variants as a foreign /r/; however, their second most frequent selection during the experiment was "no category," reflecting their inability to categorize the two variants. These results suggest a qualitative perception difference between the two groups. The experiment also revealed no significant categorization difference between the two types of /r/ allophones. The results of the discrimination experiment revealed that both groups of AE speakers performed equally well on average; however, the standard deviation of percent correct scores is greater for inexperienced AE participants than experienced AE participants. The obtained results will be discussed in relation to existing second language acquisition models.

1pSC9. Preattentive language effects in bilinguals. Adrian Garcia-Sierra, Juan F. Silva-Pereyra, Nairan Ramirez-Esparza, Jennifer Siard, and Craig A. Champlin (Inst. for Learning and Brain Sci., Fisheries Ctr. Bldg., Box 357988, Univ. of Washington, Seattle, WA 98195-7988, gasa@uw.edu)

Event related potentials (ERPs) were recorded from Spanish-English bilinguals ($N=11$) to test if their pre-attentive speech discrimination changes depending on the language they are using at the moment. ERPs were recorded in two language contexts (i.e., participants silently read magazines in the language of interest). Two speech contrast conditions were recorded in each language context. In the unique to English condition, the speech sounds represented two different sounds for the English language, but represented the same sound for the Spanish language. In the unique to Spanish condition, the speech sounds represented two different sounds for the Spanish language, but represented the same sounds for the English language. We expected that bilinguals in the unique to English condition would pre-attentively discriminate the sound during the English language context, but not during the Spanish language context. The opposite was expected for the unique to Spanish condition. A monolingual control group ($N=9$) was recruited to test the validity of the speech sounds tested. According to our expectations, language contexts produced an effect in the amplitude of the mismatch negativity ERP component elicited by the deviant stimulus. The results suggest that language contexts can affect pre-attentive auditory change detection.

1pSC10. Neurophysiological evidence of preattentive English vowel perception in Japanese, Spanish, and Russian learners of English. Valerie L. Shafer, Winifred Strange, Kikuyo Ito, Yana Gilichinskaya, Jason Rosas, and Sarah Kresh (The Graduate Ctr., CUNY, New York, NY 10016)

The purpose of this study was to examine preattentive discrimination of English vowel contrasts in Japanese, Spanish, and Russian late learners of English compared to monolingual English listeners. The robustness of discrimination, measured using mismatch negativity (MMN), was predicted to be related to the importance of different types of cues in the native language of listeners (e.g., vowel length for Japanese). Event-related potentials (ERPs) were recorded from a 65-site geodesic net while listeners ignored the speech stimuli and performed a visual target-detection task. The stimuli were natural tokens of forms that differed in vowel (V) category in the first-syllable (e.g., Vpa). One phoneme category served as the frequently-presented standard and two as the rare (deviant) stimuli. Robust MMNs to all vowel contrasts were seen for the native group that was largest over left sites. The non-native groups generally showed smaller MMNs, less left hemisphere contribution, and evidence that the first language influenced pre-attentive perception. For example, the Japanese listeners showed no MMN for the [aepa] versus [apa] contrast, which differed the least in vowel duration. These findings reveal the highly automatic nature of native language speech perception and that the change-detection process indexed by MMN is sensitive to these processes.

1pSC11. Perceptual confusability of final nasals in Southern Min. Ying Chen and Susan Guion-Anderson (Dept. of Linguist., 1290 Univ. of Oregon, Eugene, OR 97403)

The three syllable-final nasals /m/, /n/, and /ŋ/ in old Chinese have merged into two (/n, ŋ/) or only one (/ŋ/) nasal in modern Chinese languages. The perceptual confusability of place of articulation was investigated with speakers of Southern Min, a Chinese language which preserves all three final nasals. Three experiments of forced-choice nasal-identification were conducted: (1) complete CVN syllables embedded in noise, (2) CV-truncations of the CVN syllables, and (3) the excised nasal murmur, $-N$. The first experiment revealed that /m/ was the most and /n/ was the least confusable. Responses were highly accurate in the second experiment (above 85%) and around chance in the third experiment (below 40%), which indicated that listeners relied on the information in the vowel rather than the nasal murmur to identify final nasals. The vowel /i/ resulted in the most and /a/ the least misidentification of final nasals among /i, ə, a/. Low-level and falling tones resulted in more misidentification of final nasals than mid-level, high-level, and rising tones. Segment duration, F0, and formant transitions are analyzed across the vowel and tone types for insight into these findings. Lexical familiarity ratings did not show significant correlation with the perceptual results.

1pSC12. Perceived similarity of English, Korean, and Japanese consonants for the native speakers of these languages. Takeshi Nozawa (Lang. Education Cntr., College of Economics, Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu, Shiga 525-8577, Japan) and Sang Yee Cheon (Dept. of East Asian Lang. and Lit., Univ. of Hawaii at Manoa, Honolulu, HI 96822)

The present study investigated how linguistic experience would influence the perceived similarity between native and non-native and between two non-native consonants. Three native speakers of English, Korean, and Japanese produced consonants of their respective L1 in /Ca/ context. Native speakers of these three languages served as listeners. They heard two stimuli per trial and rated how close the initial consonants of the stimuli are in seven-point scale. A two-way ANOVA yielded a significant main effect of consonant pairs and an interaction effect of consonant pairs and listener groups. The results generally show that the phonology of listeners' L1 affects the perceived similarity. Japanese listeners rated English and Japanese voiceless stops, which are rather distant in VOTs, more similar than English and Korean listeners did. Korean listeners rated English voiced stops and Korean lax stops more similar than did the other two listener groups. English listeners rated Japanese liquid and English /r/ or /l/ more different than did Japanese and Korean listeners. All the listener groups rated English /s/ and Korean /*s/ as very similar. However, only Korean listeners rated English /s/ and Korean /s/ as rather distant while English and Japanese listeners rated them as similar.

1pSC13. Tone perception cues: Pitch targets, trajectories, or both? Deepti Ramadoss and Paul Smolensky (Dept. of Cognit. Sci., Johns Hopkins Univ., 237 Krieger Hall, 3400 N. Charles St., Baltimore, MD 21218, ramadoss@jhu.edu)

Zsiga and Nitisaroj [(2007)] conducted tone perception experiments to test the Moren and Zsiga [(2006)] hypothesis that the principal perceptual cues to the five-way tonal contrast in Thai are high (H) and low (L) pitch targets aligned to (subsyllabic) moras. Their experiments involved perception of synthetic stimuli: manipulated peaks or troughs (corresponding to pitch targets) connected by line-trajectories. In the present study, the stimuli created are intended to be more naturalistic. The hypothesis is that not only peaks and troughs but also the trajectories between them are informative to and important for perception. Manipulations involve realignment of the peaks and troughs, with concomitant compression or expansion of the original trajectory between them. Native Thai speakers categorize the manipulated tone and give a goodness rating. Preliminary data indicate that some categorizations (e.g., manipulated rising perceived as high) can be explained by information present in the trajectory, and not in the peak/trough. However, theories that advocate movements (i.e., trajectories) as perceptual cues [e.g., Xu (2004)] cannot account for some goodness ratings observed in these experiments. Hence, it appears that a combination of the peak/trough information, as well as the trajectory, is employed in tone perception. Work supported by Dr. Luigi Burzio and Dr. Colin Wilson.

1pSC14. Hemispheric processing of tones by listeners with or without tone experience. Xianghua Wu, Murray Munro, and Yue Wang (Dept. of Linguist., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A1S6, Canada, xwa23@sfu.ca)

Tones are used phonemically in many languages such as Mandarin, Thai, and Norwegian. Previous research [Wang *et al.* (2004)] revealed left hemispheric preference for Mandarin listeners in the processing of native tones, but no preference for either Norwegian or English listeners, even though the latter two differed in familiarity with tones. The current study examined hemispheric processing of Mandarin tones by 20 native listeners each of Mandarin, Thai, and English. The Thai listeners had experience only with Thai tones, while the English listeners had no tone experience. In a two-response dichotic listening paradigm, listeners reported the tone they heard in each ear. The pooled results across tones indicated left hemispheric preference for the Mandarin listeners, no hemispheric preference for the Thai listeners, and right hemispheric preference for the English listeners. The findings suggest that native tone experience may increase left hemisphere involvement in the processing of tones in another tone language, whereas lack of tone experience in the native language tends to be linked to non-linguistic hemispheric lateralization of tone processing. [Work supported by SSHRC.]

1pSC15. Perception of final-consonant "voicing" in whispered speech. Yana Gilichinskaya and Winifred Strange (PhD Program in Speech-Lang.-Hearing Sci., CUNY Graduate Ctr., New York, NY, 10016, ygilichinskaya @gc.cuny.edu)

The purpose of this study was to examine the accuracy of final consonant perception in whispered speech. Of particular interest was the accuracy of the detection of "voicing" in whispered consonants and its relationship to the prominence of different acoustic cues to consonant voicing. This relationship was investigated for the effect of speaker and consonant. Stimuli were natural tokens that differed in final consonant and vowel preceding it. For example, [habæz] embedded in the carrier sentence "I said testword eight times" produced at a conversational speaking rate. Recordings from four monolingual speakers of American English were used. The consonants were two pairs of stops /b-p/, /g-k/, a fricative-pair /z-s/ and an affricate-pair d ʒ-ʈ ʃ. In an eight-alternative forced choice task, listeners indicated which final consonant they heard. Analysis of the results revealed that whispered consonants were identified with a relatively high overall degree of accuracy (84%–88%). ANOVA indicated that identification accuracy was affected by both speaker and consonant. A logistic regression model was used to establish the best predictors of consonant voicing in whispered speech for classification of intended consonants and in predicting the success in [+/− voicing] detection. Predictors that were considered included several acoustic measures, e.g., vowel duration.

1pSC16. Acoustic representations for tonal phonological categories. Kristine M. Yu (Dept. of Linguist., Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, krisyu@ucla.edu)

The learnability of tonal categories was investigated by studying how category separability is affected by different phonetic representations of tone. Gauthier *et al.* [(2007)] found that $f0$ velocity contours densely sampled over time were sufficient for near-perfect categorization of Mandarin tones. Since infant learners begin as "citizens of the world" before developing language-specific representations of sound categories in response to ambient language input [Kuhl (2004)], cross-linguistic data were studied. Based on tonal production data from Bole, Igbo, Mandarin, Cantonese, and White Hmong, we found that (1) densely sampled $f0$ velocity contours are insufficient for learning tones, but (2) coarse temporal sampling of phonetic features can produce well-separated tonal categories, and moreover, that (3) phonetic features for tonal representation necessarily extend beyond $f0$ to voice quality features, and (4) features for tonal representation are language-specific. Results (2) and (3) were also supported by tonal perception experiments in Cantonese. It was found that listeners can maintain tonal identification accuracy under coarse sampling (down to three to five samples per

syllable), with the speech signal interrupted with noise as in multiple phoneme restoration experiments. It was also found that listeners are biased in tonal identification by creaky voice in the speech signal.

1pSC17. Perceptual distinctiveness of vowels in relation to dialectal sound change. Ewa Jacewicz and Robert A. Fox (SPA Labs, Dept. of Sp. Hear. Sci. Ohio State, 1070 Carmack Rd., Columbus OH, 43210, jacewicz.1@osu.edu)

We examine distinctiveness among four vowels /ɪ, ɛ, æ, a/ in Appalachian English in the perception of children (9–13 years) and adults (50–65 years) who spoke the same local variety as the talkers who produced the stimuli. Each listener responded to unique exemplars of *bids*, *beds*, *bads*, *bides* (both stressed and unstressed) produced by 40 talkers, male and female, children (8–12 years) and adults (50–65 years). The highest identification rates were for beds, and the lowest for bides. For each vowel, stressed variants yielded slightly higher rates than unstressed. Examination of confusions among spectral neighbors revealed systematic confusions between bids and beds. Confusions for bads and bides varied as a function of talker age, gender, and listener age, reflecting effects of cross-generational sound change in this dialect. In a second task, the vowels were presented in *hVd*-frame and stimulus uncertainty was elevated by increasing the number of vowel categories (12), talker age groups (3), and the addition of another dialect. Similar pattern of responses was obtained by different listeners (children and adults) but overall identification rates were comparatively higher. Altogether, these results show listener sensitivity to cross-generational sound change, which affects both vowel position and formant dynamics. [Work supported by NIH.]

1pSC18. Active change in American English dialects and their role in age perception. Krista Hadeed and James Harnsberger (Dept. of Linguist., Univ. Florida, Gainesville, FL 32611)

Prior studies on vocal aging have varied with respect to the perceptual relevance of vowel formants. Two patterns have been (inconsistently) observed: formant lowering due to age-induced laryngeal lowering and formant centralization. A third hypothesis was investigated here: that active changes in the vowel systems of American English dialects, lead by younger (especially female) speakers, can themselves serve as vocal age cues independent of the cited (physiological) changes. One active vowel change is the California vowel shift, involving the lowering/backing of [I], [E], and [ae], one that has also been observed in young speakers in Florida and elsewhere. Would such systematic differences in vowel quality occurring frequently in younger speech play a significant role in female voice age perception? Speech samples were collected from 20 young and 20 middle-aged female residents of northern Florida under four conditions: (a) while conversing with another speaker matched versus mismatched in age category and (b) having been trained (versus untrained) to accurately simulate the California vowel shift in younger speakers. Acoustic analysis and age estimation studies were conducted to determine if the presence of a natural/simulated "younger" dialect shifted perceived age and the extent of any shift relative formant lowering and formant centralization.

1pSC19. Differentiating between gay and heterosexual male speech. Erik C. Tracy and Nicholas P. Satariano (Dept. of Psych., Ohio State Univ., 1827 Neil Ave., Columbus, OH 43210, tracy.69@osu.edu)

How many phones do listeners need to hear in order to discriminate between gay and heterosexual male speakers? Prior research [Munson *et al.* (2006) has found that listeners only need to hear a single monosyllabic word to make this determination. In a series of experiments, listeners heard monosyllabic words and portions of these words, and were able to discriminate between gay and heterosexual male speakers. Experiment 1 replicated the prior results. In experiment 2, listeners were above chance in differentiating between heterosexual and gay male speakers when presented with specific phonemes, such as /s/ and certain vowels (/eɪ/, /u/, /i/, /ɛ/, //aelig/). These data extend prior work demonstrating that gay and heterosexual men produce certain phones differently [Linville (1998); Pierrehumbert *et al.* (2004)]. When presented with other phonemes (/n/, /m/, /f/, /v/, /l/, /w/), listeners were not as accurate in differentiating between gay and heterosexual male speakers. These results suggest that listeners were able to accurately discriminate between gay and heterosexual speakers when presented with specific phonemes, although they were relatively more accurate in their determinations when presented with the entire monosyllabic word.

1pSC20. On the relationship between vocal aesthetics and speech perception. Grant McGuire (Univ. of California, Santa Cruz, CA 95064), Molly Babel (Univ. of British Columbia, Vancouver, BC, Canada), Joseph King, Teresa Miller (UC Santa Cruz, Santa Cruz, CA 95064), and Alyssa Satterwhite (Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada)

This study reports data from four experiments exploring the interplay of meta-linguistic analyzes and lower level tasks with the goal of understanding how judgments of vocal aesthetics and voice typicality affect voice and phoneme processing. In the first, speakers of west coast North American English rated the attractiveness of 60 American English voices. Results from this experiment were compared against the following ones. In the second experiment listeners were asked to rate the typicality of each voice for its sex. Ratings for both showed a strong correlation, suggesting that vocal attractiveness and voice typicality are related. In the next experiment listeners were asked to quickly classify the voices as male or female. Faster reaction times correlated with judgments of higher typicality, but not with attractiveness. Finally a group of listeners were asked to classify the vowels produced by the voices. Here a correlation was found with both rating tasks such that listeners were faster at vowel classification for both the more attractive voices and the more typical ones. Moreover, a correlation was found between both online tasks such that voices that listeners classified quickly by sex were also classified quickly by vowel.

1pSC21. On the intelligibility of fast synthesized speech for people who are blind: A cross-system comparison. Ann Syrdal, Taniya Mishra, and Amanda Stent (AT&T Labs—Res., 180 Park Ave., Bldg. 103, Florham Park, NJ 07940)

People who are blind increasingly use synthesized speech as a primary output modality when interacting with computers, mobile devices, and web-based services. They typically prefer to listen to the synthesized speech at speeds multiple times real time, and consequently develop strong preferences for particular text-to-speech (TTS) engines and voices. These usage-based preferences lead to potentially incorrect assumptions about which TTS approaches are "best" for fast speech. We report on a cross-system comparison of the intelligibility of fast synthesized speech for users who are blind from birth. We used male and female voices from multiple TTS engines representing the main approaches to TTS: diphone synthesis, unit selection synthesis, HMM-based synthesis, and formant synthesis. We recruited participants from organizations that work with the blind. Each participant listened to and transcribed semantically underspecified sentences from a single TTS engine and voice, spoken at speeds ranging from 300 to 550 words/min. We used TTS engine, algorithm for speeding up synthesized speech, and speaker sex as independent variables, and transcription accuracy as the primary dependent variable. We report differences by TTS approach, speed up method, TTS engine type, and speaker sex.

1pSC22. The waveform model of vowel perception and production. Michael A. Stokes (Waveform Commun., LLC, 3929 Graceland Ave., Indianapolis, IN 46208, waveform.model@yahoo.com)

The waveform model of vowel perception and production was published in late 2009, but it has not been publicized or exposed to a wide audience before now. In this presentation, the framework of the waveform model will be explained beginning with the categorization of the vowel space, and the distinguishing features for each categorical vowel pair. From the well defined categories, the position of the lips and tongue and their association with specific formant frequencies becomes apparent, as well as an explanation of perceptual errors. With this foundation, the experimental evidence that led to 99.2% accuracy across the 50 male talkers in the Hillenbrand *et al.* dataset (1995) will be presented. The 99.2% performance was achieved with 64 lines of computer code processing the data points of the 396 vowels stored in a database. The waveform model provides a unique categorization of the vowel space, which has led to a working explanation of vowel perception, vowel production, and perceptual errors. By explaining each of

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these aspects of vowel communication and achieving 99.2% accuracy with a computer algorithm, the waveform model will have an impact in more than one field of research.

1pSC23. The influence of concurrent working memory tasks on the visual contributions to speech. J. N. Buchan (Dept. of Psych., Queen's Univ., 62 Arch St., Kingston, ON K7L 3N6, Canada, julie.buchan@queensu.ca) and K. G. Munhall (Queen's Univ., Kingston, ON K7L 3N6, Canada)

Visual speech information can influence the perception of acoustic speech. In acoustically noisy environments, people can often understand more words correctly if they can see a talker as well as hear them. Additionally, conflicting visual speech information can influence the perception of acoustic speech (namely, the McGurk effect), causing a percept of a sound that was not present in actual acoustic speech. This auditory and visual speech information does not need to be perfectly synchronous in order to be integrated. Rather, there is a "synchrony window" over which this information can be integrated. The extent to which a distracting cognitive load task affects the influence of the visual speech is still not well understood. It is also unknown whether a distracting cognitive task has an influence on the integration of temporally asynchronous speech. A series of experiments using both speech-in-noise and McGurk tasks with concurrent working memory tasks was used to address this question. The temporal offset in some of the McGurk tasks was also manipulated. Overall, results suggest that while some interference of the cognitive task can be observed, this influence is quite small and does not have a substantial influence on the integration of asynchronous speech.

1pSC24. Perception of a computational model of hypernasality: A mixed methods study. Bartek Plichta (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, plich002@umn.edu) and Peter J. Watson (Univ. of Minnesota, Minneapolis, MN 55455)

Most synthesis of nasal resonance is implemented by means of a parametric synthesizer [Klatt & Klatt, (1990)], offering a high degree of control, but having machine-generated quality [Zraik *et al.*, (2000)]. Typically this method is used to synthesize only small speech segments (i.e., sustained vowels and syllables; generating sentence-long samples is time-consuming and difficult to implement in batch mode. The alternative to parametric synthesis is LPC analysis/resynthesis because it is capable of producing natural output, but attempts to synthesize hypernasality by means of LPC analysis /resynthesis have been largely unsuccessful [Eriksson *et al.*, (2003)]. We synthesized sentence-length stimuli with different degrees of nasality using a new technique based on the work of Chen [(1995)]. To evaluate our stimuli we used a sequential explanatory design: Quantitative data were used to rate hypernasality with different degrees of added nasal resonance; and qualitative data were used to evaluate the perception of quality and naturalness of the stimuli. The combined results of both analyzes showed that the ten speech-language pathologists (SLPs) rated the degree of perceived hypernasality in accord with the degree of nasality synthesized in the stimuli, and the SLPs queried rated the overall quality of the synthesis favorably and uniformly.

1pSC25. Does hyperarticulation facilitate phonemic categorization in non-native speakers of English? M. Uther, A. Giannakopoulou (Dept. of Psych., School of Social Sci., Brunel Univ., Uxbridge, Middlesex UB8 3PH, United Kingdom maria.uther@brunel.ac.uk), and P. Iverson (Univ. College London, United Kingdom)

Hyperarticulation of vowel sounds occurs in certain speech registers (e.g., infant- and foreigner-directed speeches). Hyperarticulation is therefore presumed to have a didactic function in facilitating phonetic categorization in language learners. This event-related potential study tests whether hyperarticulation of vowels actually results in larger phonetic change responses [as indexed by mismatch negativity (MMN)] in native and non-native speakers of English. Preliminary analysis of data from native English-speaking and native Greek-speaking participants suggests a possible marginal increase in phonetic change responses (as indexed by MMN) to hyperarticulated stimuli. However, further analyzes need to be completed before any firm conclusions can be drawn as to the benefit or otherwise of hyperarticulated speech.

1pSC26. Sentence recognition in noise for English-, Chinese-, and Korean-native listeners. Chang Liu, Su-Hyun Jin, and Chia-Tsen Chen (Dept. of Commun. Sci. and Disord., Univ. of Texas Austin, 1 University Station, Austin, TX 78712)

The present study aims to investigate the effects of background noise on English sentence recognition for English-native (EN), Chinese-native (CN), and Korean-native (KN) listeners. The hearing in Noise Test (HINT) [Nilsson *et al.*, (1994)] was used to measure the percent correct word identification in both quiet and noise conditions. Each sentence consists of uniform length with six syllables spoken by a female talker with general unaccented dialect-free American English. Two types of noise, multitalker babble and long-term speech shaped noise (LTSSN), were presented at various singal-to-noise ratios SNRs. Preliminary data demonstrated that first, the sentence recognition of non-native listeners was significantly lower than native listeners in both quiet and noise conditions; second, both native and non-native listeners showed significant amount of masking release at each SNR (−10, −5, and 0 dB SNRs); third, there was substantial individual variability in sentence recognition within each non-native group. This variability might be related with high variability in phoneme recognition in noise for the non-native individuals.

1pSC27. The classification of Greek fricatives with cepstral coefficients. Angeliki Athanasopoulou and Irene Vogel (Dept. of Linguist. and Cognit. Sci., Univ. of Delaware, 46 E. Delaware Ave., Newark, DE 19716, angeliki @udel.edu)

This study examines the classification of Greek fricatives based on place and manner of articulation. Greek is particularly rich in fricatives, so a total of 10 segment types are considered: voiced and voiceless segments at five places (labial, dental, alveolar, palatal, and velar). For each segment the first five bark-cepstral coefficients were extracted and the values were used in a linear discriminant analysis to separate the different place categories in a statistically optimal way. Data from ten native Greek speakers are currently being analyzed, and preliminary results indicate that the classification of Greek fricatives is generally comparable to that of a similar study with Romanian fricatives [Spinu, L. (2010); Palatalization in Romanian: Experimental and theoretical approaches, Ph. D. thesis University of Delaware.], where overall accuracy of the classification was 78%. Nevertheless, since the inventory of modern Greek fricatives is rather different from that of Romanian, and it crucially includes a contrast between labials and interdentals, which has traditionally posed problems to this type of analysis; it is anticipated that some differences will also be observed. This, in turn, will provide further insights into the nature of fricatives across languages.

1pSC28. Identifying gender in fricatives: A comparison of human performance with a classification based on Cepstral coefficients. Laura Spinu and Maica Machnik (Dept. of CMLL, Linguist. Program, Concordia Univ., 1455 de Maisonneuve West, Montreal H3G 1M8, Canada)

A recent gender identification study using Cepstral coefficients extracted from voiced and voiceless fricatives from four different places of articulation yielded 78% correct classification in a linear discriminant analysis (LDA) [Spinu and Lilley (2010)]. This high accuracy was somewhat unexpected, as previous cross-linguistic studies of fricatives generally found no significant effects of gender on various types of acoustic measures. The current study investigates whether humans can identify gender based solely on frication noise as successfully as the LDA. 30 English-speaking undergraduates were asked to identify the gender of fricatives produced by 10 speakers (5 males and 5 females)—a subset of the sounds used in the LDA study. The analysis, currently in progress, suggests that humans may not be as successful as the LDA. These findings are expected to shed light on both (a) the differential distribution of acoustic features related to gender in fricatives, and (b) the extent to which these features are available/employed in human perception.

1pSC29. A technique for adjusting Gaussian mixture model weights that improves speaker identification performance in the presence of phonemic train/test mismatch. Jack McLaughlin and Lane Owsley (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lane@apl.washington.edu)

Speaker identification is complicated by cases where training material is phonemically deficient. Misclassifications can result either because subsequent test material from that speaker contains primarily the phonemes missing from the training data or because that test material is phonemically most consistent with another talker's model. This situation can arise in any dialog where, for reasons of brevity and clarity, conventions must be imposed on phraseology. We present here a technique for detecting phonemic deficiencies in a speaker model, and then correcting that model to partially compensate for the biased training data. This technique relies upon a specially constructed universal background model (UBM) from which speaker models are adapted. This UBM is formed by weighting several dozen phoneme GMMs using EM training. As a result, each Gaussian component of the UBM (and of the resulting speaker models) corresponds to a specific phoneme. Analysis of the speaker model weights reveals whether the training data had the typical phonemic variety found in ordinary speech, and if it did not, the weights are adjusted. Using a specially designed corpus created from the TIMIT utterances, we show that this reweighting technique improves performance over non-reweighted models. Results are also given for the Air Traffic Control Corpus.

1pSC30. Perceptual similarity, response bias, and novel phonetic categories. Noah H. Silbert, Benjamin Smith, Susan Campbell, and Catherine Doughty (Ctr. for Adv. Study of Lang., Univ. of Maryland, 7005 52nd Ave., College Park, MD 20742)

Numerous studies show a close correspondence between categorization of stimuli from a voice onset time (VOT) continuum and native language category structure. Relatively little is known about the relationships between perception, response bias, and the assignment of novel, non-native categories to tokens varying in VOT. Data from two categorization tasks in which VOT continua spanned portions of English voiceless (task 1) and voiced (task 2) categories were analyzed with a hierarchical Bayesian signal detection model in an exploratory study of these relationships. In both data sets, high-accuracy listeners show roughly equally spaced perceptual distributions and no response bias, whereas low accuracy subjects exhibit substantial variability and consistent bias to respond with categories that were not those most closely aligned with the closest native category. The model also provides a rigorous, theoretically-motivated measure of perceptual similarity, which in turn enables quantitative exploration of links between categorization behavior and popular theories of speech perception (e.g., the perceptual assimilation model and native language magnet theory). Analysis of similarities between tokens suggests a small, but consistent, perceptual magnet effect in both high- and low-accuracy groups.

1pSC31. Category interaction and stimulus effects in the perception of synthetic liquid+stop clusters. Terrance M. Nearey and Benjamin V. Tucker (Dept. of Linguist., Univ. of Alberta, Edmonton, AB T6G 2E7, Canada)

There are numerous studies of the perception of English stops in the syllables /arga, alga, arda, alda/ [e.g., A. Lotto and K. Kluender, Percept. Psychophys. **60**, 602–619 (1998)]. Listeners give more /-da/ responses after /ar-/ syllables. There is controversy over the degree to which this effect involves general auditory contrast rather than phonetic context. The present study uses a two-dimensional continuum of 49 stimuli composed by crossing a seven-step /al-/ to /ar-/ series with a seven-step /-da/ to /-ga/ series. The variation in stimulus properties is localized to F3 only for both VC and CV stimuli. Listeners ($n = 34$) responded with all four categories. Results show clear phoneme level context effects. Ambiguous VC stimuli near the /al/-/arr/ boundary show significantly more /da/ responses in cases where the VC is heard as /ar/. This pattern is consistent with what has been called a diphone bias effect [T. Nearey, J. Acoust. Soc. Am. **101**, 3241–3256 (1997)]. Surprisingly, for these stimuli, evidence for more continuous tuning of the /da/-/ga/ boundary by preceding l/r is quite weak. Thus, in this experiment, phonetic effects appear to dominate auditory contrast effects.

1pSC32. Do language-specific differences in primary acoustic cues affect relative weighting of secondary cues to phonological contrasts? Alexander L. Francis (Speech, Lang. and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Fernando Llanos (Purdue Univ., West Lafayette, IN 47907), Olga Dmitrieva (Stanford Univ., Stanford, CA), and Rachel Chapman (Purdue Univ., West Lafayette, IN)

Cross-language differences in cue weighting are often attributed to differences in phonological inventories, in that listeners are assumed to give greater weight to those acoustic cues that best differentiate the specific phonological categories of their native language. However, subphonological properties of a given contrast may also affect weighting of acoustic cues for acoustic or perceptual reasons, in a manner independent of the structure of the listeners' phonological inventory. Here, this possibility is investigated by comparing Spanish and English listeners' relative weighting of two acoustic cues to the syllable-initial stop consonant voicing contrast /b/-/p/: Voice onset time (VOT) and fundamental frequency at the onset of voicing (onset $f0$). Spanish and English possess comparable word-initial stop consonant inventories, but differ in the phonetic realization of the voicing contrast in terms of VOT: Spanish contrasts a short lag (<10 ms) /p/ with a prevoiced (<0 ms) /b/, while English contrasts a long lag (> 50 ms) /p/ with a short lag (<20 ms) /b/. Preliminary results ($N = 7$) suggest that, although speakers from both languages depend mainly on VOT, Spanish speakers give more weight to onset $f0$ when VOT is ambiguous than do English listeners.

1pSC33. Comparing weights of cues with different numbers of levels. Vsevolod Kapatsinski, Irina Shport, and Susan Guion-Anderson (Dept. of Linguist., 1290 Univ of Oregon, Eugene, OR 97403, vkapatsi@uoregon.edu)

Cue weighting is a useful methodological tool in speech perception research: it allows to access within-group and between-group biases in sound categorization. Examination of cue weighting may also have major implications for phonological theory. For instance, significant individual variation in reliance on cues to a phonemic contrast within a speech community challenges the traditional assumption of the language-specific feature system. Morrison argued that logistic regression coefficients in identification tasks provide good estimates of cue weights [Studies in Second Lang. Acquisition 597–606 (2005)]. Unfortunately, coefficient estimates vary with the number of levels a cue has, which makes it impossible to directly compare weights of cues with different numbers of levels, a serious limitation. The present paper shows that, using a null hypothesis of zero cue weight and a fixed sample size, one can employ Monte Carlo techniques to estimate the effect of the number of levels on observed cue weights (regression coefficients). One can then test whether the difference in the number of levels between cues can account for the observed difference in weights. If not, then the cue weights are reliably different. The method is demonstrated on data investigating individual variation in attention to cues to pitch accent.

1pSC34. Why diverse spectral structure coheres for children. Eric Tarr and Susan Nittrouer (Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212)

Coherence masking protection (CMP) refers to the phenomenon in which lower thresholds for accurate vowel labeling are obtained when an informative F1 is presented with a constant, and therefore non-informative, F2/F3 cosignal than when F1 is alone. Tarr and Nittrouer reported this effect for adults and showed enhanced effects for children with synthetic speech [J. Acoust. Soc. Am. **126**, S2300 (2009)]. With sine waves, no CMP was found for either group. Two new experiments asked what accounts for these differences across age and stimulus type. Hypotheses focused on the principle of common harmonic structure and expectations of speech-like signals. Experiment 1 combined sine-wave F1 with three synthetic cosignals, one that was harmonically related to F1 and two that were not. Experiment 2 used all synthetic speech, but F1 and the cosignal differed in harmonic structure. All signals were speech-like. Stimuli were presented in white noise for labeling, with and without the cosignal. Labeling thresholds were measured adaptively for F1 only and all three formants. Adults showed no significant CMP in either condition, but children did in both. It was concluded that children perceptually organize auditory signals based on the expectation of speech and do not require harmonicity. [NIDCD grant DC-00633.]

1pSC35. The role of formant-frequency contours in the perceptual grouping of speech formants. Brian Roberts, Robert J. Summers (Psych., Sch. of Life & Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom), and Peter J. Bailey (Univ. of York, Heslington, York YO10 5DD, United Kingdom)

The perceptual organization of speech remains poorly understood. Recent research using sine-wave speech suggests that the ability of an extraneous formant to impair intelligibility depends on modulation of its frequency contour [Roberts *et al.*, J. Acoust. Soc. Am. **128**, 804–817]. This study examined the effect on intelligibility of manipulating the depth of this frequency variation. Three-formant (F1+F2+F3) analoges of natural sentences were synthesized using a monotonous glottal source (F0=140 Hz). Each formant-frequency contour was scaled to 50% depth about its geometric mean; this manipulation had relatively little impact on intelligibility. Perceptual organization was probed by presenting stimuli dichotically (F1+F2C; F2+F3), where F2C is a competitor for F2 that listeners must resist to optimize recognition. Different competitors were created by inverting the frequency contour of F2 about its geometric mean and varying its depth (100%-0%, 25% steps). Adding F2C typically reduced intelligibility; this reduction was greatest for 100%-depth, intermediate for 50%-depth, and least for 0%-depth (constant) F2Cs. These results indicate that competitor efficacy depends on overall depth of frequency variation, not depth relative to that of the other formants, and suggest that frequency-contour modulation influences across-formant grouping not only in sine-wave analogues but also in more speech-like simulations. [Work supported by EPSRC.]

1pSC36. Categorical processing of German vowel quantity. Fabian Tomaschek (Hertie Inst. for Clinical Brain Reseach, Dept. of General Neurology, Univ. of Tuebingen, Germany, fabian.tomaschek@uni-tuebingen.de), Hubert Truckenbrodt (Ctr. for General Linguist. (ZAS), Berlin, Germany), and Ingo Hertrich (Hertie Inst. for Clinical Brain Reseach, Dept. of General Neurology, Univ. of Tuebingen, Germany)

The present study investigates the behavioral and neural processing of duration in the German vowel /a/ using synthesized disyllabic nonsense words. The vowel /a/ was chosen since its long and short cognates are differentiated merely on the basis of vowel duration, unlike high and mid vowels which have additional differences in formant structure. By means of an identification test a sharp category border was found at 105.9 ms in the duration continuum from short to long category associated with a local peak in reaction time. Furthermore, an adaptive discrimination test applied to the entire continuum showed a minimum of the just-noticeable-difference for vowel duration at the category border. In an MEG study, the continuum of the disyllabic words was presented in randomized order. Besides two P50-M100 complexes in response to the first and the second syllable, the ERP data showed a secondary M100-like peak in case vowel duration exceeded the duration of the short category, which was not observed in a duration-matched nonspeech control condition. This secondary M100 might be interpreted as an additional phonological “event” or “slot”, indicating a nonlinear categorical difference in the processing across short and long vowels.

1pSC37. Brief accent exposure promotes early word learning in accented speech. Rachel Schmale (Dept. of Psych., North Park Univ., 3225 W. Foster Ave., Box 16, Chicago, IL 60625, rschmale@northpark.edu), Alejandrina Cristià (CNRS, 75005 Paris, France), and Amanda Seidl (Purdue Univ., W. Lafayette, IN 47907)

Foreign accents incur processing costs for monolingual listeners, compromising speech perception for both adults [Munro and Derwing (1995)] and infants [Schmale and Seidl (2009)]. However, some adult work suggests that this disadvantage is modulated by accent experience [Bradlow and Bent (2008)]. It remains unknown whether exposure to foreign-accented speech would reveal similar benefits for toddlers, who cannot exploit lexical, top-down information. Previous work demonstrates that English-learning 24-month-olds cannot learn words when trained by a native-English talker and tested by a Spanish-accented talker [Schmale, *et al.* (in press)]. Findings are presented from four experiments with 24-month-olds ($n = 88$) using this word-learning task preceded by a 2-min exposure phase. In experiments 1 and 2, toddlers heard speech from Spanish-accented speakers in exposure (single speaker in 1; multiple speakers in 2). Looking times revealed that exposure supported word learning ($p = 0.002$ and $p = 0.003$, respectively). To tease apart the role of accent and talker variation in bolstering learning, in experiments 3 and 4 toddlers heard speech from native-English speakers in exposure (single speaker in 3; multiple speakers in 4). Looking times revealed that they failed to learn words in both experiments ($p = 0.07$ and $p = 0.5$, respectively). These results demonstrate that accent exposure bolsters early cross-accent word learning.

Session 1pUW

Underwater Acoustics and Acoustical Oceanography: Sediment Acoustics and Geological Processes II

Charles W. Holland, Cochair
Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Allen Lowrie, Cochair
U.S. Naval Oceanographic Office, Balch Blvd., Stennis Space Center, MS 39522-5001

Chair's Introduction—1:00

Contributed Papers

1:05

1pUW1. From seismic stratigraphy and oil hunting to three-dimensional geo-acoustic databases for acoustic modeling. Allen Lowrie (U.S. Naval Oceanograph. Office, Code NP-53, Stennis Space Ctr./NASA, MS 39522-5001)

This work describes a conceptually possible methodology to construct 3D multi-layered geo-acoustic databases in absense of extensive data, based on geologic first principles. Remote sensing of the sediment-covered seafloor by seismic reflection data over continental margins provides for reflector interpretation with correlations between seismic and geologic data, identifying sedimentary processes with energy levels. Realization of individual reflectors denoting a later deposition onto a pre-existing surface opens the determination of individual depositional sequences from seismic reflection data. Such interpretation permits determining single deposition with sediments from one range of sealevels over definite regions and time-units. At periodicities from 20 000–100 000 years, sealevels oscillate 25–125 m due to climate with waters "stored" on continents as ice. A specific sequence suggests sediment-types deposited with geo-acoustic values applicable. From worldwide seismic data with sequence dates verified by drillhole-groundtruth, a stratigraphy of sealevel position with time exists. For past-time-ranges, sealevel position and depositional characteristics are known. Geologic history including globally dated climate/sealevel/sedimentation oscillations means that sediments and geo-acoustics for a given time may be interpreted without seismic and/or core/drillhole data. A synthetic database of geo-acoustic values in a 3D real-world format is constructable. Thus, actualistic databases based on data and without may be constructed.

1:20

1pUW2. Anisotropy, range dependence and seismo-acoustic propagation in shallow water. Robert I. Odom (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, odom@apl.washington.edu), Jeffrey Park (Yale Univ., P.O. Box 208109, New Haven, CT 06520-8109), and Darin J. Soukup (Univ. of Washington, 68 Johnson Hall Box 351310, Seattle, WA 98195)

The shallow water environment may be highly variable, with both range dependence and anisotropy almost ubiquitous in the seafloor bottom/subbottom regions. Some common causes of range dependence include marine-sediment composition, non-planar boundaries, rough surfaces, strong density, or velocity contrasts, and variation in water-column depth and/or sediment-cover thickness. Common causes of elastic anisotropy are compositional layering or vertically aligned cracks. There is an apparent trade-off between anisotropy and range dependence, and difficult to separate the two effects in a propagating signal. If the symmetry axis of a compositionally layered sediment is not exactly normal to the seafloor, the seismo-acoustic wave field has particle-motion polarizations in all three coordinate directions. Even in a one-dimensional medium, an explosion source excites sediment particle motion with all three polarizations. Assuming sediment isotropy when it is not justified can cause errors in layer-thickness computations, elastic property-gradient determinations, acoustic attenuation, and predictions of long-range sound propagation. Employing a modal representation of the seismo-acoustic wavefield, we illustrate the effects of anisotropy and range dependence on the modes, such as mode-identity switching due to angular dispersion and coupling of longitudinal, vertical, and horizontal particle motion polarization. [Work supported by ONR.]

1:35

1pUW3. Propagation in a waveguide with range-dependent seabed properties. I. Theory and implications. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, cwh10@psu.edu)

The ocean environment contains features affecting acoustic propagation that vary on a wide range of time and space scales. A significant body of work over recent decades has aimed at understanding the effects of water column spatial and temporal variabilities on acoustic propagation. Much less is understood about the impact of spatial variability of seabed properties on propagation, which is the focus of this study. Here, a simple, intuitive expression is derived for propagation with range-dependent boundary properties and uniform water depth. It is shown that incoherent range-dependent propagation depends on the geometric mean of the seabed plane-wave reflection coefficient and the arithmetic mean of the cycle distance. Thus, only the spatial probability distributions (pdfs) of the sediment properties are required. Also, it is shown that the propagation over a range-dependent seabed tends to be controlled by the lossiest, not the hardest, sediments. Thus, range-dependence generally leads to higher propagation loss than would be expected, due, for example, to lossy sediment patches and/or nulls in the reflection coefficient. The theory may be useful for other (nonoceanic) waveguides. [Research sponsored by the Office of Naval Research Ocean Acoustics Program.]

1:50

1pUW4. Propagation in a waveguide with range-dependent seabed properties. II: Principles and examples. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA, 16804, cwh10@psu.edu)

A recently derived theory of incoherent propagation in a waveguide with range-dependent boundaries suggests that propagation can often be understood in terms of a few principles. These principles will be reviewed, followed by several examples including fluid and visco-elastic layers that illustrate how a few simple principles can lead to a variety of apparently complex range- and frequency-dependent behaviors. Implications for geoacoustic inversion strategies will also be discussed. In a few instances, propagation over a range-dependent seabed can be calculated using range-independent sediment properties. [Research sponsored by the Office of Naval Research Ocean Acoustics Program.]

2:05

1pUW5. Effects of coupling of forward-backward modal components on continental slope ocean environment. David P. Knobles (Appl. Res. Labs., Univ. of Texas at Austin, P. O. Box 8029, Austin, TX, 78713-8029)

An exact solution to the two-dimensional (2-D) Helmholtz equation in an inhomogeneous environment may be expressed as a superposition of forward and backward traveling modes. The modal amplitudes of both components are generally coupled in the same manner that positron and electron amplitudes are coupled in the presence of an electric field, resulting in Zitterbewegung for the electron. In ocean acoustics, an accurate treatment of such coupling can become important over long propagation distances across continental slope environments with inhomogenities in either the surface or bottom boundary conditions, the volume of the water column, or both. The coupled integral equation for the range-dependent modal amplitudes is decomposed into the forward and backward modal components using the asymptotic boundary conditions of Green's function operator. In addition to an exact treatment to the coupling, a perturbation approach is introduced to include the forward-backward coupling effects within a one-way integral equation. The perturbations are introduced via a channel elimination approach that results in a nonlocal optical potential or effective interaction. The nature of this potential and the perturbation convergence criteria are examined for the case of bottom sand dunes and internal waves. [Work supported by the ONR.]

2:20

1pUW6. Transmission loss tomography of littoral hydrodynamics. Tokuo J. Yamamoto (Div. of Appl. Marine Phys., RSMAS, Univ. of Miami, 4600 Virginia Key, Miami, FL 33149, tyamamoto@bellsouth.net)

Experiments of transmission loss tomography for turbulent and Reynolds stress structures were carried out in Kanmom strait using eight 5.5 kHz source-receiver pairs. The fluid motion within the turbulent current scatters the acoustic wave. The scattered acoustic waves radiate, experience losses due to attenuation and spreading, and are recorded in the form of transmission loss. Because they are small, surface and bottom losses are neglected leaving turbulent scattering and geometric spreading as the dominant transmission losses. Therefore, accurate measurement of transmission loss extracts the scattered acoustic energy losses. Total transmission loss minus spreading loss yields transmission loss (of scattered energy). Transmission loss yields turbulent current motions with the aid of the hydro-acoustics-invariant. From the fluctuations of turbulence on the orthogonal axes, the Reynolds stress tensors are calculated. Reynolds stresses make it possible to calculate the rip currents, upwelling, surf beat, wave surge, sediment transport, and all the elements of littoral hydrodynamics. The imaging of hydrodynamic elements using transmission loss alone is done for the first time.

2:35—2:50 Break

2:50

1pUW7. Mid-frequency geoacoustic inversion using bottom loss data from the Shallow Water 2006 Experiment. Jie Yang, Darrell R. Jackson, and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu)

Geoacoustic inversion work has typically been carried out at frequencies below 1 kHz, assuming flat, horizontally stratified bottom models. Despite the relevance to Navy sonar systems, many of which operate at mid-frequencies (1–10 kHz), limited inversion work has been carried out in this frequency band. This paper is an effort to demonstrate the viability of geoacoustic inversion using bottom loss data in the frequency band of 2–5 kHz. The acoustic measurements were taken during the Shallow Water 2006 Experiment off the coast of New Jersey. A half-space bottom model, with three parameters, density, compressional wave speed, and attenuation, was used for geoacoustic inversion by fitting the model to data in the least-squares sense. Inverted sediment sound speed was compared with direct measurements and inversion results using different techniques in the same area. The comparison shows that bottom loss can be used to infer sediment geoacoustic parameters at mid-frequencies. In addition, observations and modeling results demonstrate that forward scattering from topographical changes is important at mid-frequencies and should be taken into account in sound propagation predictions and geoacoustic inversion. To cope with fine-scale topographic variability, measurement technique such as averaging over tracks may be necessary.

3:05

1pUW8. Bottom parameter behavior in slightly range-dependent shallow water. A. Tolstoy (ATolstoy Sci., Inc., 1538 Hampton Hill Circle, McLean, VA 22101)

An examination of the matched field processing behavior of key geoacoustic parameters as a function of frequency (25–100 Hz), range (250 m–2 km), and sediment layer thickness (12, 22, and 40 m) in a systematic manner can be very helpful in the design of inversion methods. Even in the presence of small expected geometric errors (in source depth, phone depths, and source range) plus small errors in ocean sound-speed and bottom depths (with a slight range dependence), some low frequencies and close ranges can be surprisingly sensitive to bottom properties while still very effectively and efficiently indicating correct bottom values. Simple averaging over all such frequencies and ranges may not be optimal for maximizing sensitivity and, thus, for finding geoacoustic parameters.

3:20

1pUW9. Striation processing for sediment geoacoustic characterization. Qunyan Ren (Environ. Hydroacoustics Lab., U.L.B., av. F. D. Roosevelt 50, CP 194/05, B-1050 Brussels, Belgium, qunyren@ulb.ac.be) and Jean-Pierre Hermand (Universit Libre de Bruxelles (U.L.B.), B-1050 Brussels, Belgium)

Broadband spectrogram of sound radiated by a moving source, i.e., surface ship, exhibits striations. According to normal mode theory, the slope and position of the striations are related to the modal group and phase speeds, which are determined by the propagation medium and source movement. Waveguide invariant theory provides an interpretation of the phenomenon and has been applied in underwater inverse problems including source localization, sediment geoacoustic characterization, etc. The successful extraction and identification of the striations is a critical step for these applications. In this paper, a multiscale line enhancement filter based on Hessian matrix eigenvalue analysis is adopted for the features' extraction. Typical shallow-water environment models are selected for numerical simulations. The interference structure is effectively enhanced, the striations are extracted, and their position and slope are estimated. The effects of varying the respective parameters of a sediment layer on the striations, i.e., position shift and slope change, are also discussed. Preliminary inverse results that use the striation features for sediment geoacoustic characterization are presented.

3:35

1pUW10. Wave scattering and interaction in elastic sea beds. Anatoliy N. Ivakin and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

A first-order perturbation model of scattering in an elastic medium is revisited and discussed. The material properties of the medium are defined by three spatially fluctuating variables, the density and two Lamé parameters. The wave interaction process is described in terms of four mechanisms of scattering and energy conversion: two without change of the wave type, from compressional to compressional and from shear to shear, and two with the type conversion, from compressional to shear and vice versa. The model is applied to the case of acoustic scattering from and propagation in underwater sediments of different types, sand and rock. Wave interaction and attenuation due to various mechanisms of scattering in the sediment are considered. An improved method for calculation of the seabed scattering strength is proposed, which takes into account the so-called "windowing" effect. It allows more accurate accounting for the contribution of volume heterogeneities near the sediment surface and its comparison with the first-order roughness mechanism of scattering. The frequency-angular dependencies of the scattering strength for elastic sandy and rocky seafloors are calculated, and behaviors of the volume and roughness contributions are compared.

3:50

1pUW11. Modeling backscatter from a series of sediment rough interfaces by a normal incident chirp sonar. Dajun Tang (APL-UW, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Chirp sonar is widely used as a survey tool in shallow water. A chirp sonar measures reflection and backscatter of normal incident sound from sediment interfaces and volume heterogeneity. Motivated by using such chirp sonar data to invert for goeacoustic parameters, a forward model is developed that uses an exact method on which practical models can be based. Here volume heterogeneity is ignored and only scattering by a set of rough interface is modeled. If these interfaces are flat, the received data will be a series of reflections from the interfaces. However, these interfaces are in general rough and a forward model should be able to quantitatively handle scattering from these rough interfaces, including full scattering from each interface and multiple scattering among the set of interfaces. A set of integral equations is derived, which provides the numerical mechanism to calculate the scattered field from the rough interfaces, and numerical examples are presented.

4:05—5:05 Panel Discussion

Payment of additional registration fee required to attend. See page A22

MONDAY EVENING, 23 MAY 2011 GRAND BALLROOM A, 7:00 TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Photoacoustic Sensing and Imaging

Thomas J. Matula, Chair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Chair's Introduction—7:00

Invited Papers

7:05

1eID1. The physics of biomedical imaging and therapy by interacting light and sound. Ronald A. Roy (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, ronroy@bu.edu)

Light and sound waves propagating through tissue are coupled in useful and interesting ways. When a beam of sound travels through diffuse light, photons passing through the beam are phase modulated at the ultrasound frequency, yielding information on tissue optical and acoustical properties via the acousto-optic (AO) effect. Alternatively, the absorption of short-pulse laser light induces rapid heating that launches acoustic waves that can then be used to determine the spatial distribution of optical absorption, a phenomenon referred to as the "optoacoustic" or "photoacoustic" (PA) effect. We will present a phenomenological introduction to the emerging field of dual-wave sensing and imaging using light and sound, give a few illustrative examples, and comment on the potential for applying these concepts to tissue imaging. We then transition from imaging mode to therapy mode by introducing a high-intensity focused ultrasound (HIFU) beam that can (1) pump the AO interaction, (2) enhance the level of PA emissions, or (3) promote cavitation-mediated therapy. We include a brief review of the physical and biological effects of HIFU-induced cavitation and will discuss how the addition of light and nanoparticles enhances both the efficacy and targeting accuracy of HIFU therapy in optically diffuse tissues such as breast and brain.

Session 2aAAa

Architectural Acoustics, Signal Processing in Acoustics, Engineering Acoustics, Musical Acoustics, and Speech Communication: Memorial Session for Manfred R. Schroeder

Ning Xiang, Cochair
Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180

Gerhard M. Sessler, Cochair
Univ. of Technology, Dept. of Information Technology, Darmstadt, D-64283, Germany

Chair's Introduction—7:30

Invited Papers

7:35

2aAAa1. Memories of Manfred Schroeder at Bell Telephone Laboratories: 1955–1987. Max Mathews (Dept. Music, Stanford Univ., Stanford, CA 94305, max.mathews@gmail.com)

Manfred joined Bell Labs in late 1954 and I joined in the middle of 1955. We both retired in 1987. We and our families became close friends. This was a golden age of research at Bell Labs. ATT, a regulated monopoly with a steady source of income and a great need for new technology, well supported the research. In 1969 Manfred took a second job as head of the Third Physical Institute (Acoustics) at Goettingen, Germany, but continued working at Bell Labs, interacting closely with Bishnu Atal. Manfred's speech work led to encodings to reduce the bits needed to transmit speech. Manfred showed one does not need to encode the parts of sounds which the ear does not hear due to masking by the ear itself. He was strongly interested in number theory and used it to design colorless diffusing panels for auditoriums. He developed reproducible white noise signals to test auditoriums. He also designed colorless digital reverberators for music. Manfred and I were incredibly lucky to have lived in such an exciting time and worked at such great institutions.

7:55

2aAAa2. Manfred R. Schroeder as an academic teacher: Minutes from an infinite paradise in acoustics, number theory, and "swinging physics." Birger Kollmeier (Medizinische Physik, Universitt Oldenburg, D-26111 Oldenburg, Germany, birger.kollmeier@uni-oldenburg.de)

Being the academic supervisor of more than 40 Ph.D. students (that later became university professors or chief scientists in companies themselves) constitute only a small part of the many activities and scientific fields Manfred R. Schroeder mastered in his academic life. After finishing his Ph.D. with Erwin Meyer in Gottingen (1954) Schroeder joined Bell Labs where he finally became director of acoustics and mechanics research. In 1969 he was also appointed Professor of physics and director of the Drittes Physikalisches Institut in Gottingen. The denomination of the institute "Schwingungsphysik" (i.e., physics connected to oscillations) was deliberately mistranslated by him into "swinging physics"—a term that characterizes well the light mood and creative atmosphere among the students that was inspired by Schroeder's supportive character in the 70's–90's. His inventions and impressive lectures about acoustics, number theory, speech, and nonlinear dynamics formed the basis of his very popular books. The "Schroeder diffusors," for example, are reflection gratings based on his number theoretic considerations that suppress the specular reflections while distributing scattered sound into all different directions. The talk will give an overview of Manfred Schroeder's life in academia—from the viewpoint of one of his former students.

8:15

2aAAa3. Manfred R. Schroeder: Challenge the present and there is a better way. Bishnu S. Atal (Univ. of Washington, Seattle, WA, bsatal@bishnu.net)

I worked with Manfred Schroeder for about 25 years at Bell Telephone Laboratories starting in May 1961. The entrance hall of the Bell Labs building at Murray Hill, NJ had an inscription attributed to Alexander Graham Bell that said "Leave the beaten track and you will find something you have seen never before." Manfred always inspired me to look for things one has never seen before. Digital computers were emerging in the 1960s with great promise and we used computers to do many new things: simulation of acoustics of concert halls, studying sound decay in rooms using ray tracing, and producing high-quality speech at low bit rates. This paper will highlight a few examples. Manfred had been deep in the area of vocoder research then, but we started on a new track leading us ultimately to code-excited linear prediction. This set the spark for expanding the use of cell phones worldwide. As a manager, Manfred Schroeder was uncompromising in pushing for scientific excellence resulting in success; Bell Labs had a fearsome reputation of being the best.

8:35

2aAAa4. Manfred Robert Schroeder: A personal memoir. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

This year we are celebrating the 36th anniversary of Schroeder's seminal paper on sound scattering from maximum length sequences. This paper, along with Schroeder's subsequent publications on quadratic residue diffusers and other number theoretic sequences, broke new ground, because they contained simple recipes for designing diffusers with known acoustic performance. This presentation will review the impact of Schroeder's innovation in diffusion on my life, RPG diffusor systems, and the acoustical industry. It will also show how the evolution, design, measurement, prediction, characterization, and optimization of Schroeder's sound diffusing concepts utilized diverse disciplines, including mathematics and number theory, microwave and crystallographic diffraction, fractal geometry and self-similarity, the Helmholtz–Kirchhoff equation, periodicity and aperiodicity, signal processing, and multi-dimensional optimization. As digital technology has resulted in advances in almost every discipline, Schroeder's genius gave birth to what I call digital acoustical surfaces, which have found widespread application in every aspect of acoustical architecture. Personally I am indebted to Manfred Schroeder for my career in acoustics and the acoustical community is indebted to him for leading the way and educating all of us with wit, scientific parsimony, and insight.

8:55

2aAAa5. Transducer research at Bell Laboratories under Manfred Schroeder. Gerhard M. Sessler (TU Darmstadt, Merckstrasse 25, 64283 Darmstadt, Germany, g.sessler@nt.tu-darmstadt.de)

Electro-acoustic transducers were one of the research topics in Manfred Schroeders Laboratory at Bell Labs. One of the early highlights was the invention of the polymer-film electret microphone in 1962. Over the years, the activities were extended to the investigation of appropriate electret materials, to directional microphones, and to microphone signal processing. Manfred himself always had a keen interest in this field and made some notable contributions such as the design of directional microphones based on gradients of various orders with toroidal and unidirectional directivity characteristics. The search for electret materials suitable for electro-acoustic transducers led to the discovery of the superior charge retention properties of the fluoropolymers. Apart from microphones, other types of transducers such as touch actuators, transducer arrays for acoustic holography, and sensors for the detection of acoustic ion waves were investigated. Much of the later work of the author, performed at the TU Darmstadt, was based on the results of the early studies at Bell Labs. Examples are the introduction of silicon-based condenser microphones made with the methods of micromachining, using electret or external biasing, and new types of piezoelectric transducers consisting of charged cellular polymers, so-called ferroelectrets.

9:15

2aAAa6. Spatial audio and room acoustics. James E. West and Josh Atkins (Elec. and Comput. Eng., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, jimwest@jhu.edu)

Schroeder's interest in the behavior of sound in small enclosures eventually lead to the use of statistical methods of digital signal processing to estimate and track a room's response for echo control. He showed that a small frequency shift (5 Hz) in a loudspeaker-enclosure-microphone system could be used to mitigate the coupling feedback in public address systems. He also employed frequency shifting for echo and feedback control in the first stereo teleconferencing system. We use many of the results from the early work on stereo teleconferencing in developing solutions for multichannel, immersive systems. Schroeder's work on simulating room acoustics form the fundamentals of how we simulate and test new algorithms. His work on stereo reproduction and perception forms the basis for our motivation to explore multichannel systems (specifically, the spatial release from masking effects). The work on frequency shifting in public address systems was a first attempt at a problem that research is still focused on, multichannel acoustic echo cancellation, which addresses the howling feedback and echo that results from a loudspeaker-enclosure-microphone system.

9:35

2aAAa7. Schroeder integration: Foundation for advanced sound energy decay analysis. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180)

Schroeder's backward integration method broke the new ground in classical architectural acoustics. Soon after publication of Schroeder's backward integration method for obtaining sound energy decay functions from room-impulse responses [Schroeder, J. Acoust. Soc. Am. **37**, 409–412 (1965)], research activities in this field have been frequently reported in major journal publications. For reverberation time estimation based on a traditional straight-line model, solutions to remedy problems related to the upper limit of backward integration and the background noise in experimentally measured Schroeder decay functions have emerged. Using a parametric model derived from the nature of Schroeder integration, this author proposed a nonlinear regression method [Xiang, J. Acoust. Soc. Am. **98**, 2112–2121 (1995)]. The nonlinear regression method yields reverberation time estimates, insensitive to background noise, and the upper limit of integration. Recent interest in acoustically coupled-volume systems has prompted new challenges in analyzing sound energy decay characteristics, which are more complicated than just single-rate decays. This paper will demonstrate a suitable framework for this room-acoustics application using Bayesian inference and the Schroeder backward integration as the foundation of the advanced model-based energy decay analysis. Based on the Schroeder integration, two levels of inference, decay order selection, and decay parameter estimation have been developed for practical applications.

9:55—10:05 Break

10:05

2aAAa8. Measurement of impedance at grazing incidence with a standing-wave tube. Volker Mellert and Roland Kruse (Inst. f. Phys., Oldenburg Univ., 26111 Oldenburg, Germany, volker.mellert@uni-oldenburg.de)

As guest editor of Ando's book "Concert Hall Acoustics" Schroeder wrote 1985 in his foreword "This lack of low frequencies in the first overhead reflection revealed another low-frequency deficiency that had hitherto gone unnoticed: a progressive attenuation of low frequencies in the *direct* sound as it grazes across the rows of seats. (This "seat effect" must exist in many other halls, but it is usually

2a TUE. AM

masked by the presence of low-frequency components in the early overhead reflections).” Schroeder addressed the acoustics in a famous concert hall. Early digital computing helped to suggest improvements. Even today it remains a problem to exactly determine properties of acoustic material. Determination of the effective impedance of acoustic materials at normal incidence is a standardized procedure (ISO 10534). However, at grazing incidence, the situation is more complicated. Using a modified impedance tube, in which the material under investigation is placed parallel to the direction of wave propagation, and applying Scott’s equation (1946) linking the (complex) wave number in the channel with the properties of the porous material, today’s laptop power can even provide efficient calculations of the impedance at grazing incidence. This paper will also discuss challenges on the uniqueness of the solution.

10:25

2aAAa9. Are impulse responses Gaussian noise? Jean-Dominique Polack (LAM/IJLRA, UPMC/CNRS/Ministre de la Culture, 11 rue de Lourmel, F-75015 Paris, France jean-dominique.polack@upmc.fr)

Many years ago, Manfred Schroeder proved that transfer functions in rooms follow complex Gaussian distributions. This property was extended to impulse responses by the author, following a suggestion by Moorer for simulating impulse responses. More recently, several authors have checked again the later property with modern signal analysis tools. They obtained mixed success, with results that strongly depend on the length of the analysis window. In order to understand these unstable results, the present paper takes a closer look at the statistical distributions of both impulse responses and transfer functions. It shows that an accurate model of both the impulse response and the transfer function is necessary in order to test the distribution. Further, it presents several sources of artifacts that skew the distributions and shows that they can be ascribed to the signal processing methods used to extract the responses. Finally, the conditions for obtaining true Gaussian distributions are specified.

10:45

2aAAa10. Neurally-based acoustic and visual design. Yoichi Ando (Graduate School of Sci. and Technol., Kobe Univ., Japan, andoy @cameo.plala.or.jp)

Our earlier approaches to acoustic design drew from sources outside the listener’s nervous system: external acoustical measurements, cochlear biophysics, and overt psychoacoustical judgments [Schroeder, “Model of hearing,” Proc. IEEE **63**, 1332–1352 (1972)]. In more recent years, we have also attempted to ground acoustic design in terms of brain processes that subserve auditory percepts and preferences. Neural response correlates associated with auditory qualities related to central autocorrelation functions (temporal sensations such as pitch, timbre, and loudness) and those related to interaural crosscorrelation functions (spatial sensations) were found to be dominant in one or another cerebral hemisphere. Features of these two central correlation functions describe primary auditory qualities well. The observable neural correlates of subjective preferences and annoyances involve the temporal persistence and spatial extent of EEG and MEG alpha rhythms in cortical regions. Analogous percepts, preferences, and correlation-based representations have been explored for a number of visual patterns (flickers, oscillating objects, and textures). Subjective scale values of listener preferences for sounds, sound fields, and visual patterns all show the same cortical alpha rhythm persistence and extent [Ando, Auditory and Visual Sensations, Springer-Verlag, (2009)]. Rational design of sensory experience is possible given a theory of stimulus generation and modification, percepts, and preferences. In acoustics, adaptive methods can be used to optimize concert hall sound fields, musical instruments, and speech recognition.

TUESDAY MORNING, 24 MAY 2011 GRAND BALLROOM B, 11:15 A.M. TO 12:15 P.M.

Session 2aAAb

Architectural Acoustics: Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture I: Musical Illusions, Absolute Pitch and Other Enigmas of Sound Perception

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

William J. Cavanaugh, Cochair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Rd., Sudbury, MA 01776

Chair’s Introduction—11:15

Invited Paper

11:20

2aAAb1. Musical illusions, absolute pitch, and other enigmas of sound perception. Diana Deutsch (Dept. of Psych., Univ. of California, San Diego, La Jolla, CA 93093-0109, ddeutsch@ucsd.edu)

Several phenomena that were discovered by the author are described, and their implications for music and speech perception are explored. First described is an algorithm for producing circular scales from sequences of single tones, with each tone comprising a full harmonic series. Next discussed are illusions, such as the scale illusion and the glissando illusion, which occur when two streams of sound emanate simultaneously from different regions of space. In several such illusions, a perceptual reorganization occurs, so that a melody formed of tones in one pitch range appears to be coming from one region of space, and another melody, formed of tones in a different pitch range, appears to be coming from the opposite region. Next explored is an illusion in which a spoken phrase comes to be heard as sung rather than spoken, simply by repeating it several times over. Finally, absolute pitch is examined. Although this has been considered a rare faculty, it is shown that speakers of tone language have a very high prevalence of absolute pitch; it is argued that this is due to the association of pitches with meaningful words during the critical period for speech acquisition. The talk is accompanied by sound demonstrations.

TUESDAY MORNING, 24 MAY 2011 ISSAQUAH, 8:00 TO 11:30 A.M.

Session 2aABa

Animal Bioacoustics: Memorial Session in Honor of Ronald Schusterman and David Kastak I

Robert Gisiner, Cochair
OPNAV N45, Navy Energy and Environmental Readiness Div., Arlington, VA 22202

Roger L. Gentry, Cochair
ProScience Consulting LLC, 22331 Mt. Ephraim Rd., Dickerson, MD 20842

Patrick W. Moore, Cochair
National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106

Chair's Introduction—8:00

Invited Papers

8:05

2aABa1. Acoustic work with Ron Schusterman and Dave Kastak. Roger L. Gentry (ProSci. Consulting, LLC, Dickerson, MD 20842, roger.gentry@comcast.net)

This paper describes auditory-related work done with Ron Schusterman from 1964 through 2002, and with Dave Kastak from 1999 to 2007. Some of Schusterman's early experiments failed to demonstrate echolocation in pinnipeds, resulting in a conflict with Poulter in 1965 that Ron did not resolve until 2000. He conducted several experiments to investigate what functions other than echolocation sea lion click vocalizations might serve. In 1965 he advised on measuring underwater auditory localization in sea lions, and between 1970 and 1976 he and his colleagues produced aerial and underwater audiograms for two seal species. His many other projects on hearing and vocalization, conducted with his colleagues, may be reviewed by other speakers in this symposium. Schusterman joined the NOAA panel writing noise exposure criteria for marine mammals from 1998 to 2002. In 2002 he asked Dave Kastak to replace him. Kastak then made significant contributions to the criteria from 2002 until his untimely death in 2007, just before the criteria were published. Auditory research lost two giants with their passing.

8:25

2aABa2. *In vivo* measures of hearing in seals via auditory evoked potentials, otoacoustic emissions, and computerized tomography. Darlene R. Ketten (Dept. of Biology, Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, dketten@whoi.edu), C. Rogers Williams (Natl. Marine Life Ctr., Bourne, MA 02532), T. Aran Mooney (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Keith Matassa, and Kristen Patchett (Univ. of New England, Biddeford, ME 04005)

Dr. Schusterman and Dr. Kastak made significant contributions to our basic understanding of pinniped hearing but recognized also that data, particularly from older animals, could reflect abnormalities. This concern is particularly acute for stranded animals and species for which we lack baseline data. Approximately 20% of stranded seals are estimated to have ear infections, which if undetected and untreated can lead to septicemia and death. In this study, five harbor seals (Phoca vitulina) with suspected infections were examined with CT, OAE, and AEP to determine hearing test diagnostic correlates of ear pathologies. OAEs were obtained between 0.5 and 15 kHz. AEPs were obtained from 1–30 kHz. CT, AEP, and OAE results were assessed independently. Four animals had moderately elevated thresholds but normal brainstem responses, consistent with CT findings of conductive loss from occluded middle ears but normal inner ears. Differences in percent occlusion over time and across individuals were consistent with threshold variations. A fifth animal with no overt signs of infection but scans showing aggressive inner, middle, and external ears disease, had no responses within 70 dB of normal response ranges. These data demonstrate that OAE/AEP features are important diagnostic tools for treatment and rehabilitation decisions. [Work supported by the ONR.]

8:45

2aABa3. Hearing pathways in the finless porpoise, *Neophocaena phocaenoides*, and implications for noise impacts. T. Aran Mooney (Dept. of Biology, Woods Hole Oceanograph. Inst., Woods Hole, MA), Songhai Li (Univ. of Hawaii, Kailua, Hawaii), Darlene R. Ketten (Oceanograph. Inst., Woods Hole, MA), Kexiong Wang, and Ding Wang (Inst. of Hydrobiology, The Chinese Acad. of Sci., Wuhan, 430072, People's Republic of China)

How an animal receives sound will influence how it uses or is impacted by sound. While the "jaw hearing" hypothesis is well supported, work has been limited to a few "representative" species. There are clear variations in the jaw and head morphologies of odontocete species suggesting subtle variation in sound reception. Here we address how a divergent cetacean species, the Yangtze finless porpoise (*Neophocaena phocaenoides asiaeorientalis*), receives sound. Noise impacts on this subspecies are a concern as they inhabit

waters with many acoustic sources. Hearing was measured using auditory evoked potentials. Click and amplitude modulated tone stimuli were presented at nine locations on the head and body using a jawphone transducer. Thresholds were compared to anatomical dissections and CT scans of porpoise heads. Minimum thresholds and best hearing locations were at the cheek-fat-pad and distal to the porpoise bulla. However, thresholds were not substantially different at locations from the rostrum tip to the ear (11.6 dB). This minimal variation is quite different from the 30–40 dB differences found across the head of bottlenose dolphins and belugas suggesting differences in how divergent odontocetes receive sound. Porpoises may have relatively less "shading" of sounds and are potentially more susceptible to masking effects.

9:05

2aABa4. Noise-induced temporary threshold shift in marine mammals. James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, San Diego, CA 92152, james.finneran@navy.mil) and Carolyn E. Schlundt (ITT Corp., San Diego, CA 92110)

In 1996, Dave Kastak and Ron Schusterman reported the first observations of noise-induced temporary threshold shift (TTS) in a marine mammal [D. Kastak and R. J. Schusterman, "Temporary threshold shift in a harbor seal (Phoca vitulina)," J. Acoust. Soc. Am. **100**, 1905–1908 (1196)]. Since that time, a number of studies have been conducted to characterize the effects of noise on the hearing abilities of pinnipeds and odontocete cetaceans. These studies compare hearing thresholds before and after subjects are exposed to intense sounds and relate feature of the noise exposure to the observed threshold shift. The results are analogous to data from terrestrial mammals, where TTS depends on the frequency, amplitude, duration, and temporal pattern of the noise exposure, as well as the hearing test frequency and the recovery time. This talk reviews the major findings related to the growth and recovery of TTS in marine mammals, with an emphasis on recent data from high-frequency tonal exposures in bottlenose dolphins. The relationship between onset-TTS levels across a range of frequencies and proposed auditory weighting functions will also be discussed. [Work supported by the ONR.]

9:25

2aABa5. Temporary hearing threshold shifts and recovery in a harbor porpoise (*Phocoena phocoena*) and harbor seals (*Phoca vitulina*) exposed to white noise in a 1/1-octave band around 4 kHz. Ronald Kastelein, Robin Gransier, Ron van Mierlo, Lean Hoek (Sea Mammal Res. Co. (SEAMARCO), Julianalaan 46, 3843 CC Harderwijk, The Netherlands, researchteam@zonnet.nl), and Christ de Jong (TNO Sci. and Industry, 2600 AD Delft, The Netherlands)

To set safety criteria for levels of sounds produced during pile driving for offshore wind parks, temporary hearing threshold shifts (TTSs) were studied in a harbor porpoise and two harbor seals. A psychoacoustic behavioral technique was used to quantify TTS and hearing recovery in the animals exposed to underwater sounds (white noise in a 1/1-octave band around 4 kHz). Hearing thresholds were determined once a day for a narrow-band frequency swept sine wave (3.9–4.1 kHz) before and after exposure to the fatiguing noise, which was offered at two sound pressure levels (131 and 119 dB *re* 1 μPa for the porpoise and 151 and 139 dB *re* 1 μPa for the seals) each at four durations (15, 30, 60, and 120 min). Recovery was quantified by measuring hearing thresholds at 4, 8, 12, and 48 min after the noise exposure. The results show that TTS in harbor porpoises occurred at a sound exposure level threshold approximately 20 dB below the threshold for which TTS occurred in harbor seals.

9:45

2aABa6. Behavioral reactions of dolphins and sea lions to sonarlike sound exposures. Dorian S. Houser, Laura Yeates (Natl. Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, dorian.houser@nmmpfoundation.org), Daniel E. Crocker (Sonoma State Univ., Rohnert Park, CA 94928), Steve W. Martin, and James J. Finneran (SSC Pacific, San Diego, CA 92152)

Colleagues Ron Schusterman and Dave Kastak were instrumental in the advancement of marine mammal cognition and sensory biology studies. In the past 2 decades, their work was heavily concerned with the potential impact of anthropogenic sound on pinnipeds. Behavioral reactions of marine mammals to anthropogenic sound exposure are one impact with a large range of potential consequences. Recently, 30 dolphins and 15 sea lions were exposed to sonarlike pings used in antisubmarine warfare. Exposures occurred while animals performed a behavior in which they traveled from one location to a second where they touched a paddle, and then returned for a food reward at the starting location. Subjects performed a ten trial baseline session followed by a ten trial experimental session in which sound exposures occurred. Each subject was randomly assigned a receive level from near ambient to 185 dB sound pressure level. For each subject, received levels were consistent across all experimental trials. Behavioral reactions were anticipated prior to the study and assigned a severity score by a panel of anonymous reviewers. Scores were used to explore relationships between the received level and the severity of the behavioral response.

10:05—10:25 Break

10:25

2aABa7. Auditory habituation as a diagnostic measure of domoic acid toxicosis in wild sea lions. Peter Cook (Dept. of Psych., Univ. of California, Santa Cruz, CA 95060), Colleen Reichmuth (Univ. of California, Santa Cruz, CA 95060), and Frances Gulland (The Marine Mammal Ctr., Sausalito, CA 94965)

Habituation measures are instrumental in behavioral neuroscience, but they have seldom been applied to research with marine mammals. In this study, an auditory habituation paradigm was applied as a diagnostic tool in California sea lions with suspected brain damage resulting from neurotoxic exposure to domoic acid. Domoic acid is a glutamate agonist that causes damage to the hippocampus and surrounding medial temporal areas. Such damage is typically revealed only by brain imaging or *postmortem* histology, and is therefore difficult to diagnose in a veterinary setting. Studies of laboratory animals suggest that habituation of defensive responses to aversive stimuli does not generally rely on hippocampal function, while habituation of orienting responses to non-aversive stimuli has been shown to rely on this brain area. Here it was found that behavioral orienting responses to a series of non-aversive sounds habituate more slowly in sea lions with domoic acid toxicosis than in sea lions with no apparent neurological deficits. A signal detection analysis

indicates that a measure based on this habituation shows significant diagnostic value. Preliminary results of a follow-up procedure probing habituation and dishabituation to auditory stimuli presented from multiple locations are also discussed. [Work supported by NSF Graduate Fellowship].

Contributed Papers

10:45

2aABa8. Preliminary investigation of sound reception in southern sea otters (*Enhydra lutris nereis*). Asila Ghoul and Colleen Reichmuth (Univ. of California, Santa Cruz, Long Marine Lab, 100 Shaffer Rd., Santa Cruz, CA 95060, asila@ucsc.edu)

Due to their dependence upon a highly restricted coastal habitat, sea otters are vulnerable to a variety of environmental and anthropogenic threats. Among these is the potential disturbance from human-generated sources of noise. Presently, there are no data on the auditory sensitivity of sea otters, and little evidence to suggest what sounds may be most relevant to these animals. As an initial step toward describing the acoustic sense of sea otters, we conducted a controlled exposure experiment, adapted from sound exposure studies used in behavioral field research, to efficiently measure the aerial frequency range of hearing in four captive sea otters. This approach was designed to determine which frequencies were audible to each animal, rather than to provide direct measures of auditory sensitivity. The maximum range of aerial hearing determined using this method was 0.125 to 32 kHz. These are the first direct measurements of hearing obtained for sea otters, and the results are relevant to improving understanding of their acoustic communication, evolutionary biology, and behavioral ecology, as well as in supporting ongoing conservation efforts. This research effort draws from the work of Kastak and Schusterman, especially with respect to the value of behavioral baselines in captive studies of marine mammals.

11:00

2aABa9. Developing auditory weighting functions in a bottlenose dolphin (Tursiops truncatus). Carolyn E. Schlundt (ITT Corp., 3276 Rosecrans St., San Diego, CA 92110) and James J. Finneran (Space and Naval Warfare Systems Ctr., San Diego, CA 92152)

The variation in susceptibility to noise as a function of frequency is handled by "weighting" sound exposures to emphasize frequencies where auditory sensitivity is highest. This technique allows the use of single, weighted numeric values for impact or damage-risk criteria, regardless of the sound frequency. Human weighting schemes were derived from measurements of equal loudness curves obtained from subjective experiments where listeners directly compare the loudness of sounds at different frequencies. Response times to acoustic detection tasks provide an indirect method to construct equal-latency contours in terrestrial mammals that are analogous to equal-loudness contours. The need for empirical measures of loudness contours or auditory weighting functions in marine mammals became especially apparent following experiments of temporary threshold shift (TTS) in dolphins that revealed frequency-dependent effects for onset-TTS levels. The objective of this effort was to develop auditory weighting functions for *Tursiops truncatus* by directly measuring subjective loudness as a function of the sound frequency. The resulting equal-loudness contours emphasize frequencies at which auditory sensitivity is highest and lessen the importance of other frequencies, similar to human A- and C-weighting networks. [Work supported by the ONR.]

11:15

2aABa10. The effect of age-related hearing loss on echolocation: Changes in click parameters and echolocation discrimination abilities are initiated by changes in auditory filters. Laura N. Kloepper, Paul E. Nachtigall, and Marlee Breese (Hawaii Inst. of Marine Biology Marine Mammal Res. Program/Dept. of Zoology, Univ. of Hawaii at Manoa, P.O. Box 1106, Kailua, HI 96734)

High-frequency hearing loss has been correlated with a reduction both in echolocation click parameters and in echolocation discrimination abilities in a false killer whale. During a 15-year time period, the whale demonstrated a significant decrease in peak frequency, center frequency, and source level of outgoing clicks between two studies. Echolocation clicks were analyzed from the most recent phase of the discrimination study to determine if there were significant differences for click parameters according to target condition. The whale consistently produced clicks with the same peak and center frequencies and source levels but varied the number of clicks according to experimental condition. The data suggest that the whale does not use spectral adaptations during discrimination. Likely, a gradual shift in click parameters resulted from a change in the auditory filtering processes initiated with age-related hearing loss.

TUESDAY MORNING, 24 MAY 2011 ASPEN, 8:00 A.M. TO 12:00 NOON

Session 2aABb

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Fish Bioacoustics I

Richard R. Fay, Cochair
Loyola Univ., Parmly Hearing Inst., 6430 N. Kenmore, Chicago, IL 60626

Joseph A. Sisneros, Cochair
Univ. of Washington, Psychology Dept., Seattle, WA 98195

Invited Papers

8:00

2aABb1. Speciation and sounds of fishes: Dividing up the bandwidth. Joseph J. Luczkovich (Inst. for Coastal Sci. and Policy and Dept. of Biology, East Carolina Univ., Greenville, NC 27858, luczkovichj@ecu.edu) and Mark W. Sprague (East Carolina Univ., Greenville, NC 27858)

Fishes in the drum family (Sciaenidae) make sounds to communicate, but they do not make the same sounds. The species-specific calls have different dominant frequencies, are produced in spawning aggregations at different times of the day and season, and there is spatial segregation among the spawning fish populations. We predicted that the pattern of bandwidth use by these species would show low overlap in space, time, and sound frequency. We monitored the seasonal pattern of sound production of Sciaenidae in Pamlico Sound (NC) using autonomous sound recorders that recorded 10 s of sound every 15 min during May–Nov. The observed bandwidth ranges and spawning season for species are weakfish 300–400 Hz in May–Aug, silver perch 800–1500 Hz May–Aug, spotted seatrout

200–400 Hz June–Sep, red drum 100–200 Hz in Sep–Oct. Overlapping calls in these species were rare temporally and spatially, as evidenced by long-term passive acoustic monitoring. Two other species of fishes (oyster toadfish and striped cusk eels), in unrelated families, also compete for acoustic bandwidth in Pamlico Sound, but overlap temporally with the sciaenids. There is low probabilty of signaling confusion for these species but higher probability for Sciaenids. It appears that bandwidth partitioning has evolved in the Sciaenidae.

Contributed Papers

8:15

2aABb2. Modeling fish aggregation sounds in very shallow water to estimate numbers of calling fish in aggregations. Mark W. Sprague (Dept. of Phys., East Carolina Univ., Greenville, NC 27858, spraguem@ecu.edu) and Joseph J. Luczkovich (East Carolina Univ., Greenville, NC 27858)

Many fishes in the Family Sciaenidae produce courtship sounds associated with spawning. Some estuarine sciaenid species such as weakfish *Cynoscion regalis*, spotted seatrout *C. nebulosus*, and silver perch *Bairdiella chrysoura* occur in very shallow water (water depths 3–10 m) aggregations so large that individual calls are no longer audible above the combined sound of calling fish in the aggregation. We have measured sound levels in these aggregations that are 30 dB greater than the estimated received level of an individual fish at a distance of 1 m. To estimate the number of individual calling fish that would result in these sound levels, we have produced a model of virtual fish aggregations based on echosounder data. We used a finite difference time domain (FDTD) model to calculate the pulsed sounds from the individual fish in randomized aggregations at our virtual hydrophone location. We produced many randomized instances of fish aggregations to obtain statistical distributions of sound levels produced by different densities of calling fish. The characteristics of aggregation sounds will be presented for different densities of calling fish for these species. These calculations could lead to better estimations of fish densities from single-hydrophone measurements.

8:30

2aABb3. Seasonal chorusing in deep-water coastal habitats recorded off Oahu, HI. Marc O. Lammers, Michael Richlen, Anne E. Rosinski, and Whitlow W. L. Au (Hawaii Inst. of Marine Biology, P.O. Box 1346, Kaneohe, HI 96744, lammers@hawaii.edu)

The acoustic behavior of animals living at the edge or below the photic zone is poorly documented. Limited opportunities to make recordings and/or behavioral observations in deep water environments have resulted in a paucity of information about sound production by animals living at depth. Here we report on an effort begun in 2006 to acoustically monitor deep reef environments off Oahu, HI. Ecological acoustic recorders (EARs) were deployed along the perimeter of the island and along the slope of a nearby bank at depths ranging between 115 and 575 m. At several locations, one of the predominant sounds recorded was a chorus of pulsed signals centered at approximately 4 kHz timed closely with the period following sunset. The duration of the chorus varies between 1 and 10 h in length and is tied to the lunar cycle. The chorus is seasonal, beginning in late December and ending in early June. This period corresponds closely with the seasonal presence of humpback whales (Megaptera novaeangliae) in Hawaii. The source of the chorus is presently still unknown, but being investigated using a combination of passive and active acoustic methods.

8:45

2aABb4. Long-term monitoring of bocaccio abundance using passive acoustics. Ana Širović (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu) and David Demer (NOAA Fisheries, La Jolla, CA 93027)

The bocaccio (*Sebastes paucispinis*) was a part of an important rockfish fishery to California anglers and commercial fishers until they were declared overfished by the Pacific Fisheries Management Council in 1999. Historically, rockfish stocks were estimated using ichthyoplankton sampling and catch per unit effort of the rockfish recreational fishery. More recently, with the closure of the fishery, non-lethal methods for stock assessment have been in development such as a combination of shipboard multifrequency echosounders and underwater cameras. To augment these methods, passive acoustics have been shown to be a possibility for long-term monitoring, but a further investigation in its use for rockfish population monitoring is necessary. Sounds produced by bocaccio were recorded off San Clemente Island in the Southern California Bight in the 1960s by Thompson, and again at a near-by location in 2007. A comparison of these recordings with the contemporary population estimates for bocaccio in the area will shed a light in the feasibility of using passive acoustics for long-term monitoring of changes in the population abundance of this commercially important species.

Invited Papers

9:00

2aABb5. Disruptive communication: Stealth signaling in the toadfish. Allen F. Mensinger (Dept. of Biology, Univ. of Minnesota Duluth, 1035 Kirby Dr., Duluth, MN, 55812, amensing@d.umn.edu)

Acoustic advertisement signals have evolved in many taxa and are best known among arthropods, anurans, and birds. In populations where individuals alternate signals, males often adjust the timing of calls to avoid overlap. However, specific attempts at jamming especially using a signal that differs temporally and spectrally in fundamental frequency from the advertisement signal appears rare. In the oyster toadfish, Opsanus tau, neighboring males will alternate production of long duration (~400 ms) advertisement calls or boatwhistles. However, males can also produce short duration grunts (~100 ms). These grunts are emitted almost exclusively during the boatwhistle of a conspecific male and specifically target the latter portion of the call. The fundamental frequency of the boatwhistle is modified by this disruptive grunt, jamming the signal by lowering its fundamental frequency. The disruptive counterpulse is initiated and completed during the second, tonal portion of the boatwhistle, and its brevity and timing may allow its emitter to remain undetected. The selective forces that are driving the evolution of complex signaling mechanisms remains to be determined; however, it is possible that the alternating boatwhistles evolved as a jamming avoidance function and the disruptive grunts arose to counter this strategy.

9:15

2aABb6. Convergent evolution and intermediate status of the sonic mechanism in a perciform fish *Glaucosoma buergeri (Glaucosomatidae)*. Michael L. Fine (Dept. Biology, Virginia Commonwealth Univ., Richmond, VA 23284-2012), Hin-Kiu Mok, Kai-Yun Tsai, Pai-Ho Chiu (Natl. Sun Yat-sen Univ., Kaohsing, Taiwan), and Eric Parmentier (Univ. of Liege, Belgium)

Little is known about evolution of swimbladder sound production in fishes. Typical fish sound production utilizes fast muscles that drive the swimbladder to produce sound as a forced but rapidly-damping response; a muscle contraction rate of 200 times/s will generate a sound with a fundamental frequency of 200 Hz. Recently, slow muscles have been demonstrated in a carapid fish. These muscles slowly extend the anterior swimbladder by stretching a swimbladder fenestra until the bladder snaps back exciting sound production. Here we describe sounds produced by a similar but phylogenetically-unrelated mechanism in a perciform fish. A pair of extrinsic sonic muscles originates on the pterotic bones and inserts on the anterodorsal swimbladder just forward of a stretchable swimbladder fenestra. A fan-shaped tendon that ends in a smooth muscle attaches to the bladder just forward of the fenestra. The tendon-smooth muscle pair will be stretched by contraction of the rostral sonic muscles. Strain energy stored in the tendon will cause the anterior bladder to recoil upon muscle relaxation. Sound pulses consist of two parts: a low-amplitude one followed by a higher frequency higher amplitude part. We suggest that the first part of the call is caused by contraction of the anterior sonic muscle (cocking) and the second (release) is forced by strain energy in the stretched tendon and smooth muscle.

9:30

2aABb7. Acoustic, visual, and chemical signaling during courtship in an African cichlid fish. Karen P. Maruska and Russell D. Fernald (Dept. of Biology, Stanford Univ., 371 Serra Mall, Stanford, CA 94305, maruska@stanford.edu)

The ability to attract and assess the quality of potential mates is critical for reproductive success, but few studies examine coincident multimodal sensory cues used for reproductive signaling within a single species. Here we examined acoustic, visual, and chemical signaling used by male African cichlid fish (*Astatotilapia burtoni*) during reproduction. Dominant males produced pulsed sounds in tight association with visual courtship displays where their bodies quivered vigorously in close proximity to a female. Sound production always occurred with visual courtship displays, but not all body quivers were associated with sound, which supports the hypothesis of intentional acoustic communication. Peak frequency of courtship sounds decreased with increasing male body size, possibly providing females with an honest signal of quality. Male chemical signaling, visualized as urine pulses, was higher in the presence of gravid compared to non-gravid females, but did not occur during any specific acoustic or visual courtship displays. These data suggest that dominant males use acoustic, visual, and olfactory channels at different times during courtship possibly to convey distinct types of information such as social/dominance status, motivation to spawn, and fitness.

9:45

2aABb8. Acoustic communication in the electric yellow cichlid, *Labiochromis caeruleus*. Dennis M. Higgs, Amanda N. Barkley (Dept. Biological Sci., Univ. of Windsor, 401 Sunset, Windsor, ON N9B 3P4, Canada, dhiggs@uwindsor.ca), and Craig A. Radford (Univ. of Auckland, Warkworth 0941, New Zealand)

The cichlidae represent an attractive model for acoustic communication as their well-characterized adaptive radiation can serve as a backdrop for testing evolutionary hypotheses. Despite interest in sound communication in cichlids, little is known of their hearing ability and less is known about hearing and sound production in the same species. The current study examined sound production, hearing, and auditory morphology in a Lake Malawi cichlid *Labidochromis caeruleus*. Males and females were paired in the laboratory and all behavioral and acoustic displays recorded. Hearing was tested to tone bursts and samples of recorded calls using auditory evoked potentials and morphology was assayed using MicroCT scans of intact fish. Males produced calls with dominant frequency of approximately 300 Hz but only when simultaneously performing a quiver display. Fish detected tones from 100–1000 Hz and were more sensitive to tones than to playbacks of call segments. Finally, MicroCT showed a heart-shaped swim bladder with anterior protrusions directed at the large saccular otoliths, possibly reducing self-generated noise by focusing the call away from the ears. With this combination of behavioral, morphological, and physiological approaches, we were able to fully characterize acoustic communication in this species for the first time.

10:00—10:15 Break

10:15

2aABb9. Derivation of a response severity index model for physiological quantification of fish response to impulsive sound. Michele B. Halvorsen, Christa M. Woodley (Pacific Natl. Northwest Lab., Marine Sci. Lab., Marine Biotechnology Group, Sequim, WA 98382), Brandon Casper (Dept. of Biology, Univ. of Maryland, College Park, MD 20742), Thomas J. Carlson (Marine Sci. Lab., Sequim, WA 98382), and Arthur N. Popper (Univ. of Maryland, College Park, MD 20742)

Assessment of the effects on fish from exposure to impulsive and other sound has been limited by the ability to quantify physiological injuries, which range from mortal to recoverable. Over the past several years, a panel of measures has evolved that includes postexposure assessments of external injury, internal injury in response to rapid decompression, impulsive sound, and explosive sound exposures. This has led to a model that permits reduction in a complex panel of injuries into a single metric. Computation of the metric considers the rank of the injury by its physiological significance to the condition of exposed fish and the severity of observed injuries. The metric has been found to have a wide dynamic range in that it is sensitive to the onset of injury at very low intensity exposures through mortal injuries at severe exposures. The model accommodates normalization of outcomes between experiments separated in time, which utilize separate fish populations by accounting for the condition of fish in addition to experimental control strategies. The model is explained, and its use in derivation of an exposure-response function for juvenile Chinook salmon exposed to impulsive sound is presented.

10:30

2aABb10. Responses of fish to underwater blasting signals: Application of the fish index trauma model. Christa M. Woodley, Michlele B. Halvorsen, and Thomas J. Carlson (Marine Biotechnology, Pacific Natl. Northwest Lab., Marine Sci. Lab., Sequim, WA 98382)

The Columbia River Channel Improvement Project lead by the U.S. Army Corps of Engineers, Portland District included removal of a basalt rock ledge extending into the navigation channel, using confined underwater explosions. We performed compliance monitoring and conducted a study of the response of juvenile Chinook salmon (126.1 mm FL SD 24.8) and rainbow trout (68.3 mm FL SD

4.1) to blast signals. Cages designed for high velocity conditions contained depth-acclimated neutrally buoyant fry and juvenile salmonids and tourmaline blast sensors were placed 100–300 ft from the center of arrays of explosive charges. Blast durations ranged from 0.42 to 1.38 s, positive peak pressures ranging from 27.6 to 163.4 kPa, SELs ranging from 194 to 205 dB. Exposed fish were examined for physical damage at 0, 24, and 48 h postexposure. Injuries ranged from scale loss to lacerated organs. Some injuries lessened in severity over 48 h while others increased. Using an analysis model fish index of trauma (FIT), we described the exposure-response functions of the fish to the blast exposure. The FIT metric quantifies fish trauma and recovery potential for blast exposed fish and is sensitive to the onset of nonlethal injury at low severity exposure levels.

10:45

2aABb11. Recovery from exposure to pile driving signals by Chinook salmon. Brandon M. Casper, Frazer M. Matthews (Dept. of Biology, Univ. of Maryland, College Park, MD 20742, bcasper@umd.edu), Michele B. Halvorsen (Battelle Marine Sci. Lab., Sequim, WA 98382), Thomas Carlson (Pacific Northwest Natl. Labs., Battelle, Portland, OR 97204), and Arthur N. Popper (Univ. of Maryland, College Park, MD 20742)

Juvenile Chinook salmon exposed to pile driving sounds showed a variety of barotrauma injuries immediately post-exposure, with number and severity of injuries increasing with increased cumulative sound exposure levels (SELcum). Important remaining questions pertain to the severity of these injuries over time post-exposure including whether (a) injuries are eventually mortal, (b) new injuries show up, or (c) there is recovery over time. Juvenile Chinook salmon were exposed to either high or low SELcum pile driving stimuli. Fish were assessed for barotrauma injuries immediately following exposure and at 2, 5, or 10 days post-exposure. No fish died during recovery and all fish exhibited normal feeding behavior within hours of exposure. The injuries observed at day zero were comparable to those found in the previous pile driving studies at the same SELcum. However, beginning with day 2, the frequency of appearance of injuries began to decrease, with fish at day 10 showing few remaining injuries. These results suggest that Chinook salmon exposed to pile driving stimuli have the potential to recover from barotrauma injuries, and that new injuries did not show up post-exposure. These are important findings that need to be considered when establishing exposure criteria for pile driving projects.

11:00

2aABb12. Mechanically-induced hair cell death using the zebrafish lateral line as a model system. Zhongmin Lu, Maxwell A. Frye, and Alexandra A. DeSmidt (Dept. of Biology, Univ. of Miami, 1301 Memorial Dr., Coral Gables, FL 33146)

Noise-induced hearing loss is a major type of hearing impairment that affects the quality of life for millions of Americans. The cause of noise-induced hearing loss is the death of sensory hair cells in the ear. The primary goal of the present study is to have a better understanding of mechanisms underlying noise-induced hearing and balance deficits. Mechanically-induced hair cell degeneration was investigated using the zebrafish (*Danio rerio*), which has become a model system for biomedical research because of a combination of powerful genetics, excellent embryology, and exceptional *in vivo* visualization in one organism. Zebrafish have the lateral line system containing superficial neuromasts in the head and trunk, and each neuromast has a cluster of hair cells that structurally and functionally resemble the hair cells in the ear. In this study, hair cells in single neuromasts of zebrafish larvae of 5 to 10 days post fertilization were stimulated by a piezoelectric actuator and the time course of degeneration of hair cells exposed to overstimulation was obtained using light microscopy. Microphonic potentials of neuromast hair cells were also recorded to show effects of intense mechanical stimulation on the function of hair cells. [Work supported by NIH/NIDCD R21DC009879 and Gabelli Fellowship.]

11:15

2aABb13. The effect of noise on behavior and acoustic communication in the blacktail shiner (Cyprinella venusta). Daniel E. Holt (1477 N. Donahue Dr., Unit # 2703, Auburn, AL 36830, holtdan@auburn.edu) and Carol E. Johnston (Auburn Univ., Auburn, AL 36849)

Despite their seemingly quiet underwater habitat, freshwater fishes are not sheltered from elevated noise levels. Anthropogenic noise along with natural noise sources increase noise levels in aquatic environments. Higher noise levels can result in elevated hearing thresholds and decrease the signal-to-noise ratio of acoustic signals. Because many fishes use acoustic signals during critical life history stages, it is important to determine whether elevated noise levels affect behavior or sound production during these stages. This study describes sound production and corresponding behaviors as well as hearing in the black tail shiner (*Cyprinella venusta*). An attempt was also made to determine the effect of elevated noise levels on acoustic communication and behaviors associated with reproduction and aggression. Behavioral interactions of *C. venusta* were recorded under noisy and quiet acoustic conditions. Temporal and spectral parameters of the calls as well as the type and number of behaviors performed by the dominant male of each trial were compared between the two conditions to determine if noise affected them. Hearing sensitivity and range were determined using the auditory brainstem response approach.

11:30

2aABb14. Role of environmental acoustic cues in the evolution of hearing specialization in cyprinidae. Carol E. Johnston and Daniel E. Holt (Fisheries, Auburn Univ., Auburn, AL 36849)

Numerous studies have documented hearing specializations in fishes, but virtually no work has focused on the adaptive significance of sensitive hearing to fishes. An exception is the finding that marine clupeid fishes have an adaptation for hearing the ultrasonic acoustic signals of their mammalian predators, an obvious advantage. The idea that hearing specializations in most fishes evolved as a predator detection mechanism has not been tested in freshwater fishes, however, and due to ecosystem differences may not explain the evo-

lution of such traits. Field observations suggested that cyprinids (Ostariophysi) congregate near areas of disturbed substrate to feed on dislodged aquatic insects and debris. This led to the exploration of whether the acoustic cues associated with substrate disturbance alone would attract cyprinds via a field playback experiment. These results may shed light on the detection of feeding cues as one of the potential selective pressures associated with the evolution of sensitive hearing in Ostariophysian fishes.

Contributed Paper

11:45

2aABb15. Testing mortality of fish larvae due to simulated offshore piling noise. Christ A.F. de Jong, Pieter J.G. van Beek, Michael A. Ainslie (TNO, Stieltjesweg 1, 2628 CK Delft, The Netherlands, christ.dejong@tno.nl), Loes J. Bolle, Olvin A. van Keeken, Cindy J.G. van Damme, Hendrik V. Winter, and Dick de Haan (IMARES, 1976 CP IJmuiden, The Netherlands)

Driven by the concern that impulsive noise produced by offshore pile driving may lead to mortality of fish larvae, a device was developed for testing the sensitivity of small fish and fish larvae to sound exposure. The device consists of a rigid-walled cylindrical chamber (110-mm diameter, 160-mm height), driven by an electrodynamical sound projector. Samples of up to 100 larvae can be exposed simultaneously to a homogeneously distributed sound pressure and particle velocity field, at a controllable static pressure up to 3 bars. Two configurations are available with either a dominant sound pressure or a dominant particle velocity exposure. Recorded piling noise can be reproduced in a controlled way, in the frequency range between 50 Hz and 1 kHz, at zero to peak pressure up to 40 kPa and single pulse sound exposure levels up to 187 dB re 1 μPa^2 s, or peak particle velocity up to 2.2 cm/s and integrated square particle velocity level 124 dB re 1 $(nm/s)^2$ s. Tests are carried out in which sole (*Solea solea*) larvae at different stages of development were exposed to various levels and durations of piling noise.

TUESDAY MORNING, 24 MAY 2011 GRAND BALLROOM A, 7:30 A.M. TO 12:00 NOON

Session 2aBA

Biomedical Acoustics and Physical Acoustics: High Intensity Focused Ultrasound (HIFU) Induced Thermal Lesion Imaging with Ultrasound

Lawrence A. Crum, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Vera A. Khokhlova, Cochair
Moscow State Univ., Acoustics Dept., 119992, Moscow, Russia

Chair's Introduction—7:30

Invited Papers

7:40

2aBA1. Acoustic neural network thermometry in monitoring of thermal therapies. K. Michael Sekins (Siemens Medical Solutions, Ultrasound Div., 22010 S.E. 51st St., Issaquah, WA 98029-7298)

The potential benefits of monitoring tissue temperatures and lesioning by "acoustic" methods are evident. Such methods are typically challenging, and limited to interrogating (acoustically) only one or two physical processes during dosing (e.g., sound speed and thermal expansion), which may narrow valid temperature ranges. Due to the complexity of tissue behavior and acoustic interactions during heating, a "gestalt" of acoustic parameters was enabled to be "learned" during dosing by a recurrent neural network (RNN). Using this method, real-time three-dimensional (3-D) acoustic thermometry was demonstrated and (from RNN coefficients) insights were gained into specific acoustic parameter contributions accompanying temperature changes during and after dosing. The RNN method is described for two applications. First, for radiofrequency ablation on excised bovine livers, coregistered *B*-mode and ARFI elasticity images were acquired. RNN thermometry images were then calculated from rf data, thermal doses calculated, and the necrosis region was then determined, and shown in good agreement with elasticity lesion images. Secondly, HIFU was used to ablate excised bovine liver regions, and the RNN temperatures in tissues were shown to reflect projected power depositions of the HIFU transducer. These results support RNN thermometry as potentially suitable for some thermal therapies. [This research sponsored by U.S. DARPA Contract No. HR0011-08-3-0004.]

8:00

2aBA2. Two-dimensional real-time ultrasound technique to control lesion size during high intesity focused ultrasound therapy. John Petruzzello, Ajay Anand, Shiwei Zhou, Shriram Sethuraman, and Jose Azevedo (Philips Res. North America, 345 Scarborough Rd., Briarcliff Manor, NY 10510)

The lack of an accurate low-cost technique for real-time monitoring of ablative therapies such as HIFU has limited its clinical adoption. Ultrasound based monitoring would enable a more widespread usage of HIFU therapy. We have developed an acoustic radiation force (ARF) based technique for controlling the lesion size and placement during HIFU therapy. A series of experiments were performed in excised bovine liver tissue to evaluate the proposed technique. Two different parameters were developed to perform real-time monitoring of lesions. The first parameter, normalized displacement difference (NDD), is defined as the difference between normalized displacement induced by ARF at the therapy endpoint and a reference determined from the data. This parameter was accurate in determining lesion sizes below 8 mm in diameter as independently determined by histological examination. The lesion dimensions estimated with this noninvasive approach matched histology to within ±2 mm. A second approach was based on the saturation of displacement as treatment progressed. This technique was accurate (>2 mm) for all lesions examined. Both the NDD and displacement saturation techniques were independent of treatment time and power. This study demonstrates potential for use of these techniques in real-time monitoring of HIFU therapy to determine therapeutic endpoint and improve treatment efficacy.

8:20

2aBA3. Characterization of high intensity focused ultrasound induced thermal lesions from changes in ultrasound backscatter. Chapelon Jean-Yves, Chenot Jeremy, Souchon Remi, and Melodelima David (INSERM, LabTAU U556, Univ. of Lyon, 151 cours Albert Thomas, 69424 Lyon Cedex03, France, jean-yves.chapelon@inserm.fr)

Although good results have already been obtained, notably in urology, ultrasound-guided HIFU devices lack a real time monitoring system different from conventional *B*-mode ultrasound imaging to check the efficacy of the treatment procedures. This presentation describes the development and assessment of several noninvasive procedures developed in our laboratory for making local measurements of attenuation variations, strain changes, and/or temperature increases during HIFU treatment procedures. The techniques to characterize lesions generated by HIFU are based on estimation of relative changes in tissue properties derived from backscattered rf data. Our recent work has demonstrated good agreements between ultrasound backscattering changes versus temperature increase and thermocouple measurements. In *ex vivo* liver samples, a linear relationship between changes in the radiofrequency signal and temperature was found for temperatures up to 90 °C. Using differential imaging, any issues related to the heterogeneity of the medium is eliminated, since only changes in attenuation, strain, and temperature of tissue are taken into account. These ultrasound techniques can be used during *in vivo* treatment procedures and they can be easily implemented in real time.

8:40

2aBA4. Real-time tumor ablation monitoring and mapping using harmonic motion imaging *ex vivo* and *in vivo*. Elisa Konofagou, Yi Hou, Caroline Maleke, Fabrice Marquet, and Jonathan Vappou (Dept. of Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave, New York, NY 10027)

Harmonic motion imaging (HMI) uses a focused ultrasound (FUS) beam to generate an oscillatory acoustic radiation force for an internal, noncontact palpation to internally estimate relative tissue hardness. HMI estimates and maps the tissue dynamic motion in response to the oscillatory force at the same frequency, and has been shown feasible in simulations, phantoms, *ex vivo* human and bovine tissues, as well as animals *in vivo*. Using an FUS beam, HMI can also be used in an ideal integration setting with thermal ablation using high-intensity focused ultrasound (HIFU), which leads to an alteration in the tumor stiffness. In this paper, the capability of HMI to localize and target the tumor, as well as monitor its subsequent ablation, is assessed. The findings presented demonstrate that HMI is capable of both detecting and characterizing the tumor as well as efficiently detecting the onset of ablation. More importantly, HMI is shown capable of distinguishing the tumor margins from that of the thermal lesions *in vivo* in order to assess the treatment success. HMI may thus constitute an integrated, real-time method for efficient HIFU monitoring. [This work was supported by NIH Grant No. R21EB008521.]

9:00

2aBA5. Near real time evaluation of cardiac radiofrequency ablation lesions with intracardiac echocardiography based acoustic radiation force impulse imaging. Patrick D. Wolf (Dept. of Biomedical Eng., Duke Univ., 136 Hudson Hall, Durham, NC 27708-0281, patrick.wolf@duke.edu), Stephanie A. Eyerly, David P. Bradway, Douglas M. Dumont (Duke Univ., Durham, NC 27708-0281), Tristram D. Bahnson (Duke Univ. Medical Ctr., Durham, NC 27708), Kathy R. Nightingale, and Gregg E. Trahey (Duke Univ., Durham, NC 27708-0281)

[Cardiac radiofrequency (rf) ablation is used to treat cardiac arrhythmias. Multiple rf lesions are created to form lines of electrical block to disrupt arrhythmic wavefronts. However, image based lesion evaluation is not performed and clinicians rely on changes in electrical propagation to evaluate lesion quality. A near real time lesion evaluation system was developed and tested *in vitro* and in canines using acoustic radiation force impulse imaging (ARFIi) combined with a catheter tracking system (CARTO, Biosense Webster). rf ablation was performed and delivery sites were indicated on a CARTO electroanatomic map. A CARTO spatially tracked SoundStar® catheter was then used to steer the imaging plan to transect lesions, and intracardiac echocardiography (ICE) based ARFIi (Siemens, S2000 Acuson) was used to evaluate lesion contiguity and transmurality. Imaging was gated to diastole when unablated tissue is soft; motion filtering was employed to reduce the effects of catheter and cardiac movement. ARFIi based evaluation of rf lesions demonstrated a resolution of less than 2mm *in vitro* and lesion assessment correlated with electrical block in animal studies ($\lambda=0.8$). Combining catheter tracking with ICE based ARFIi allows near real time evaluation of cardiac ablation lesions. [Work supported by Siemens Medical, Biosense Webster, and NIH Grant No. R01-EB-012484 and R37-HL096023.]

9:20

2aBA6. Real-time monitoring and control of thermally induced lesions using high intensity focused ultrasound. Dalong Liu, John R. Ballard, Choi Jeunghwan, John C. Bischof, and Emad S Ebbini (College of Sci. and Eng., Univ. of Minnesota, 200 Union St. SE, Minneapolis, MN 55455)

A real-time temperature/strain imaging system for monitoring HIFU-induced lesions was implemented in conjunction with a pulsed HIFU (pHIFU) phased array driver. The integrated system is capable of collecting two-dimensional (2D) strain/temperature fields at high frame rates (>200 fps) to capture intricate tissue motions and deformations in the vicinity of the pHIFU focus. In addition, the system is capable of refocusing and updating the pHIFU beam at rates up to 1000 times per second. These capabilities of the guidance and pHIFU control allow for precise lesion formation in the presence of tissue motion and deformation due to breathing and pulsation in the vicinity of large blood vessels. We present results from *in vivo* experiments in the hind limbs of Copenhagen rats (250 g average). The results demonstrate the capability of the strain imaging system to estimate the thermally induced strain (due to pHIFU) in the presence of high strains from breathing and vessel pulsation *in vivo*. In this paper, we describe the architecture and the signal processing algorithms used in the estimation of pHIFU-induced strains. The robustness of the method will be illustrated by describing the results from the application of pHIFU beams in close proximity to femoral artery of the rat model.

9:40—10:00 Break

10:00

2aBA7. Ultrasound-based targeting and monitoring of high intensity focused ultrasound fields. Francesco P. Curra and Neil Owen (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington)

A new method to address the challenging tasks of image-guidance, targeting, and treatment monitoring during HIFU treatments is presented. The approach, enabled by the use of a novel multi-layer PZT-PVDF array with broad receive bandwidth in conjunction with a programmable ultrasound engine, uses the passive-mode received echoes of the imaging array with a custom pixel-based beamforming for HIFU focal tracking and targeting, allowing real-time two-dimensional (2D) B-mode visualization of the HIFU beam. Temperature monitoring during treatment is based on acoustic nonlinear propagation theory and the physical relationship of sound speed and attenuation to frequency and temperature. The harmonics-rich received echoes are processed differentially, encoded into an RGB additive color channel, beamformed, and overlayed in color over regular B-mode images. Dynamic local temperature changes in the region of interest become visible as the 2D color image changes from frame to frame. Preliminary results on beam visualization and temperature estimations during HIFU exposure in ex-vivo muscle tissue will be presented. [Work supported by NIH NIDDK].

10:20

2aBA8. Imaging and monitoring non-cavitating focused ultrasound lesions using light and sound. Ronald A. Roy, Puxiang Lai, James R. McLaughlan, Andrew B. Draudt, Robin O. Cleveland (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215), and Todd W. Murray (Univ. of Colorado at Boulder, 427 UCB, Boulder, CO 80309)

Acoustically imaging non-cavitating HIFU lesions is challenging due to the relatively weak contrast between normal and necrosed tissues. However, thermal lesions posses optical scattering and absorption characteristics that can differ significantly from surrounding tissue. Acousto-optic (AO) sensing refers to the detection of phase modulated photons generated by the nonlinear interaction of diffuse laser light (532 and 1064 nm) and a focused ultrasound beam. AO sensing in excised chicken breast exposed to HIFU (1.1 MHz) is used to sense the onset and spatial extent of lesion formation in real time and to image an existing lesion after HIFU exposure. We also demonstrate that the AO signal can be used to provide real-time feedback in order to more effectively control the duration of the HIFU exposure. [Work supported by the Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

10:40

2aBA9. Heat diffusion constrained inversion of backscattered ultrasound data to image temperature rise during high intensity focused ultrasound therapy. Peter J. Kaczkowski, Gavriel Speyer, Andrew A. Brayman, Lawrence A. Crum (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pjk@uw.edu), and Ajay Anand (Philips Res., Briarcliff Manor, NY 10510)

Noninvasive ablative high intensity focused ultrasound (HIFU) therapy must be guided with precision, and monitored in real-time. Magnetic resonance imaging (MRI) can provide both high resolution tissue-specific images and temperature maps, but even the most recently developed MRI methods cannot do so in less than a few seconds. Ultrasonic imaging techniques using a sequence of rf frames to measure heating-induced apparent strain have been developed to produce heating maps, but the approach is challenging due to lack of sensitivity and substantial variability in tissue properties. To improve estimates of temperature rise, constraints based on heat diffusion modeling are imposed, thus minimizing the effects of noise and nonmonotonicity of the speed of sound with respect to temperature throughout the therapeutic range. Furthermore, noninvasive protocols for measuring relevant HIFU field and tissue properties in the region of interest enable patient-calibrated mapping of temperature rise during HIFU, at ultrasonic imaging frame rates. Further analysis of the heat-induced apparent strain leads to a modal decomposition of the strain, greatly reducing the computational load for use in real-time feedback and therapy control. Finally, a Rytov approximation applied to the problem leads to further improvement in computational efficiency and physical understanding. [Work supported by NIH grant No. CA109557.]

Contributed Papers

11:00

2aBA10. Phase propagation in ultrasonic backscatter monitoring of high intensity focused ultrasound therapy. Gavriel Speyer, Peter Kaczkowski, Andrew Brayman, and Lawrence Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Phase propagation using the Rytov method has recently been proposed as a means for modeling the time-of-flight changes induced by thermal therapy [Speyer *et al.*, J. Acoust. Am. **127**]. These results are extended to measurements from a linear array, under which the general problem of imaging material changes is cast. The linear array offers several design components, which can be exploited for therapy monitoring, including the apodization and probing frequency. Phase propagation models are shown to be consistent with many aspects of conventional modeling, linearizing material changes around the same operating points as have been proposed by other researchers, and providing time-of-flight changes linearly related to the temperature distribution under these conditions. Beyond expanding on model properties, experimental evidence is presented, which indicates that phase propagation modeling is significantly more consistent with backscattered ultrasound data than conventional ray approaches. [Work supported by NIH Grant No. 5R01CA109557.]

11:15

2aBA11. Passive mapping for imaging thermal lesions in tissue. Carl R. Jensen (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7DQ, United Kingdom), Gyöngy Miklós (Pázmány Péter Catholic Univ., Budapest 1083, Hungary), Robert Ritchie, and Constantin C. Coussios (Univ. of Oxford, Oxford OX3 7DQ, United Kingdom)

Passive acoustic mapping, a method of mapping sources of high-frequency ultrasound emissions during focused ultrasound treatment, has been tested in *ex vivo* ox liver. The method uses an array of ultrasound sensors placed coaxially with a focused therapeutic ultrasound transducer. The ultrasound emissions received during treatment are first filtered and then beamformed to produce reconstructions of either broadband or harmonic source intensity within the tissue during the application of focused ultrasound. At pressure amplitudes in excess of the inertial cavitation threshold, passive mapping of broadband emissions is found to provide a good indication of the location and size of the lesion, while mapping of harmonic emissions adequately describes the location and onset of boiling. The results for different treatment parameters across several liver samples will be presented. Alternative methods of postprocessing the passively received data for improved spatiotemperoral resolution and signal-to-noise ratio will also be introduced and compared. These methods are being developed to characterize cavitation activity in order to give estimates of cavitation energy, tissue heating, and temperature during therapeutic ultrasound exposures, to ultimately enable prediction of cell death during treatment.

11:30

2aBA12. Quantitative ultrasound assessment of thermal therapy in liver. Jeremy P. Kemmerer (Univ. of Illinois at Urbana-Champaign, 405 North Mathews Ave., Urbana, IL 61801, kemmere1@illinois.edu) and Michael L. Oelze (Univ. of Illinois at Urbana-Champaign)

High-intensity focused ultrasound (HIFU) is a promising modality for non-invasive therapy with application to cancer treatment. Non-invasive monitoring and assessment of HIFU remains a critical hurdle to widespread clinical acceptance of HIFU. Quantitative ultrasound (QUS) is a novel approach for characterizing tissue microstructure that shows promise for monitoring and assessment of HIFU therapy. The QUS parameters of effective scatterer diameter (ESD) and effective acoustic concentration (EAC) were examined differentially in rat liver specimens exposed to HIFU to form visible lesions. Increases in ESD of 15% and decreases in EAC of 3 dB were observed in post-exposure compared to pre-exposure scans, and statistical significance was observed in most cases ($p < 0.05$). These results were compared to rat liver samples heated in a water bath, in which the samples were heated to a therapeutic temperature (60 °C) for several minutes, and ultrasonic backscatter was measured at 37 °C. ESD was observed to decrease by an average of 15% in exposed compared to unexposed sample sections. [Work supported by NIH R01EB008992.]

11:45

2aBA13. The application of phase-shift nanoemulsion in high-intensity focused ultrasound-mediated heating and its potential in monitoring lesion formation. Peng Zhang and Tyrone Porter (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, pzhang@bu.edu)

Phase-shift nanoemulsion (PSNE) consists of dodecafluoropentane nanodroplets stabilized with phospholipid monolayer shell. These nanodroplets are pressure sensitive and can be vaporized into gas bubbles by a short ultrasound pulse ($t<50$ ms), provided the pressure exceeds a well-defined threshold. This property provides means of on-site and on-demand nucleation by applying PSNE into high-intensity focused ultrasound. In this study, the bubble-enhanced lesion formation with the presence of PNSE in albumin-containing gel phantom was investigated. The results demonstrated that the onset of bubble-related heating could be well controlled by choosing the vaporization time and the bubble location was confined at the HIFU focal region. By acoustically driving the vaporized bubbles, localized bubble-enhanced heating and a significant reduction of 71.9% in acoustic power were achieved. Furthermore, the vaporized bubbles can be monitored with *B*-mode ultrasound due to the huge difference between gas and tissue. By comparing the optical image of lesion and ultrasound image of bubble cloud, a good correlation in location and shape between lesion and bubble cloud was observed. The correlation suggests that the bubble-enhanced heating is localized at the region of bubble cloud. By monitoring cavitation activity, the spatial information of lesion formation could be provided during HIFU treatment.

TUESDAY MORNING, 24 MAY 2011 WILLOW B, 8:00 TO 11:35 A.M.

Session 2aNS

Noise, Committee on Standards, and Physical Acoustics: Aircraft Flyover Noise Measurements and Source Modeling

Richard L. McKinley, Chair
Air Force Research Laboratory, 2610 Seventh St., WPAFB, OH 45433-7901

Invited Papers

8:00

2aNS1. Improved prediction of community noise footprints from high performance military aircraft. Christopher M. Hobbs (Wyle Labs., 241 18th St. South, Ste. 701, Arlington, VA 22202, chris.hobbs@wyle.com), Russell W. Powers, Dennis K. McLaughlin (Penn. State Univ., University Park, PA 16802), Kenneth J. Plotkin (Wyle Labs., Arlington, VA 22202), and Philip J. Morris (Penn. State Univ., University Park, PA 16802)

Experimental research has shown how the addition of properly designed chevrons to model scale supersonic nozzles reduces the noise of the high speed jets issuing from such nozzles. It has also been shown that, to reasonable accuracy, this noise benefit scales between small- and moderate-scale nozzle geometries. The new advanced acoustic model (AAM) developed by Wyle Laboratories has the capability to use noise source data to produce dynamic acoustic footprints showing noise exposure in the vicinity of airfields: especially for landing and take-off operations. AAM uses information on aircraft engine jet exhaust noise sources for specific aircraft engine operating parameters and given flight conditions. This presentation describes the measurement and modeling of jet noise reduction using chevrons in both small and moderate scale nozzles. Then the AAM is used to demonstrate the expected change in acoustic footprint size for a full-scale jet aircraft, where it equipped with chevrons. The effects of the chevrons are described as a decibel correction to the noise source spheres used by the model. The AAM is run for a simulated trajectory of a typical operation for a jet aircraft with and without chevrons to show the effects on the acoustic footprint.

8:20

2aNS2. Development of fighter aircraft flight procedures for reduced noise impact. Anthony R. Pilon (Adv. Dev. Programs, Lockheed Martin Aeronautics Co., 1011 Lockheed Way, Palmdale, CA 93599, tony.pilon@lmco.com), Robert A. Psencik, Darrell W. Cribbs, and Jeanette M. Elliott (Lockheed Martin Aeronautics Co., Fort Worth, TX 76108)

Lockheed Martin engineers have recently developed a unique process to rapidly develop realistic and community-friendly fighter aircraft procedures for use near airports. The process integrates known aircraft performance (takeoff distance, rate of climb, etc.) with community noise estimation tools (Noisemap, INM), allowing rapid calculation of the changes in community noise impact due to proposed changes in flight procedures or the integration of new aircraft. Automated performance calculations capture the effects caused by

changes in aircraft weight, external loading, and airfield density altitude. Operational pilots from the airport under study validate each proposed procedure in a realistic flight simulator (with performance matching the actual aircraft). These "flights" ensure that each proposed profile is safe, realistic, and reasonable for that base. The process has been applied to contemporary fighter (F-16) operations, as well as those for aircraft entering service soon (F-35). When necessary, the newly developed process can be used to create flight procedures which reduce the aircraft noise impact in nearby communities. Initial studies have demonstrated that the community area significantly impacted by noise (i.e. that within the 65 DNL contour) can be significantly reduced through judicious control of power settings, climb angles, and other flight parameters, without modification of the aircraft.

8:40

2aNS3. How to improve the value of noise and performance data of fly-over measurements with high performance jets. Theo van Veen, Wim Lammen, and Johan Dijkhuizen (Air Traffic Environment and Policy Dept., Natl. Aerosp. Lab., Anthony Fokkerweg 2, 1059CM, Amsterdam, tavveen@nlr.nl)

The quality and value of noise measurements of high-performance jets depend on a complex set of factors. A thorough preparation and analysis phase is not always standard practice. Critical factors during preparation are, for example, the selection of the location, the aircraft, the communication with the pilot, the design of the flight profiles during the runs, the method and instrumentation for recording of the aircraft performance, and noise and meteorological conditions during the test. After the test, a thorough analysis of the data has to be performed to select the data with the required quality. Further processing of this raw recorded data will lead to noise levels in corrected LAmax, SEL, or other metrics. In this paper attention is paid to the critical factors that can be influenced prior to the test and after the test to maximize the quality of the data used for noise calculations. A good preparation of the test and an analysis of the quality of the measured data set are essential to guarantee the acquisition of a high quality data set after fly-over noise measurements. The paper contains critical factors of a noise test with aircraft types like the F 16, F 18, and F 35.

9:00

2aNS4. Microphone arrays for the measurement and modeling of vertical take-off and landing aircraft. Richard L. McKinley, Hilary L. Gallagher, and Robert C. McKinley (Air Force Res. Lab./711 HPW/RHCB, 2610 Seventh St., Wright-Patterson Air Force Base, OH 45433-7901, richard.mckinley@wpafb.af.mil)

Jet aircraft, which have vertical take-off and landing capabilities, presents special noise measurement and source modeling challenges. When the aircraft is high above the ground, the predominant source is the jet plume/plumes. When the aircraft is close to the ground, the jet impingement becomes a second major source. Aircraft height above ground, weight, wind, and power setting all are important variables to consider for measurement and source modeling. This paper will review full scale measurements and source modeling conducted on legacy aircraft such as the AV-8B Harrier and considerations for the measurement and source modeling of new aircraft with vertical capabilities.

9:20

2aNS5. Characterizing nonlinearity in jet aircraft flyover data. J. Micah Downing (Blue Ridge Res. and Consulting, 13 1/2 W. Walnut St., Asheville, NC 28801, micah.downing@blueridgeresearch.com) and Kent L. Gee (Brigham Young Univ., Provo, UT 84602, kentgee@byu.edu)

Various analysis techniques have been proposed as means of characterizing acoustical nonlinearities in high-thrust engine noise. These include probability distributions for the pressure and the time derivative of the pressure (i.e., the gradient), the skewness and kurtosis coefficients of the pressure and its time derivative, and ordinary and higher-order spectral measures. In this paper, a number of these analyzes are evaluated using acoustic data recorded during various military jet flyovers.

9:40

2aNS6. Simulation of rotary and fixed wing flyover noise for subjective assessments. Stephen A. Rizzi (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681-2199, stephen.a.rizzi@nasa.gov), Aric R. Aumann (Analytical Services and Mater., Inc., Hampton, VA 23666), Matthew P. Allen, Ricardo A. Burdisso (Virginia Tech, Blacksburg, VA 24061), and Kenneth J. Faller, II (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA 23681-2199)

Subjective assessments of noise from aircraft flight operations require time histories of acoustic pressure at listener positions. Synthesized sound has an advantage over recordings by allowing the examination of proposed aircraft, flight procedures, and other conditions or configurations for which recordings are unavailable. Previous work by the authors [J. Acoust. Soc. Am. **113**, 2245 (2003); **114**, 2340 (2003); **116** 2515 (2004)] focused on the development of a two-stage process for simulating flyover noise. The first stage entailed synthesizing the time histories at the flying source based on predictions of the source directivity. A second real-time rendering stage entailed propagation to a listener position on the ground, with binaural playback over headphones. More recently, loudspeaker based rendering was implemented for three-dimensional presentation in the NASA Langley Exterior Effects Room [J. Acoust. Soc. Am. **128**, 2482 (2010); POMA **9**, 015004 (2010)]. This paper discusses recent developments to the synthesis and rendering stages, including a new technique for sample-based synthesis of rotary wing sources, improvements to fan noise synthesis, and enhancements to ground plane simulation.

10:00

2aNS7. A simple-source model of military jet aircraft noise. Jessica Morgan, Kent L. Gee, Tracianne Neilsen, and Alan T. Wall (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, jessmorg2@byu.net)

Several inverse methods exist to analyze and characterize the turbulence-induced aeroacoustic sources from military jet aircraft. The current study's purpose is to explore an alternative approach, using a semi-empirical equivalent simple source model. The model originated as a single point source located above a hard plane such that the locations of destructive interference from its mirror image matched the measured interference nulls. This appears to provide useful information about the location of the dominant noise source region from the jet. The model has developed into a superposition of Rayleigh-distributed line arrays of uncorrelated and correlated monopoles. The model is tested on an extensive set of acoustic data taken on an F-22 Raptor. Although the model's line source characteristics are developed using data from only one measurement plane, it is able to accurately reproduce the radiation at other measurement planes. This equivalent line source matches the current prevailing theory that the sideline radiation is largely due to uncorrelated noise whereas the correlated sources dominate downstream. In addition, the semiempirical, simple-source model results corroborate the theory that the peak source location moves upstream with increasing frequency and lower engine conditions.

10:15—10:35 Break

10:35

2aNS8. Comparison of near-field military jet aircraft noise with similarity spectra. Tracianne B. Neilsen, Kent L. Gee, Alan T. Wall (Dept. of Phys., Brigham Young Univ., 283 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC 28801)

The jet mixing noise from high-performance military aircraft is broadband and, like other jets, appears to have two origins: the fine-scale turbulent structures (FSSs) close to the nozzle and the large-scale turbulent structures (LSSs) farther down the plume. The intense LSS noise dominates the radiation at inlet angles greater than 125 deg, and the more omnidirectional FSS noise is the dominant source of noise to the side and aft of the aircraft. An extensive set of measurements was taken of F-22 Raptor noise at different engine conditions in 2009. The F-22 noise is compared to the self-similarity spectra generally applied to describe the contributions of the LSS and FSS in jet noise [Tam and Zaman, AIAA J. **38**, 592–599 (2000)]. For large inlet angles, the shapes of the noise spectra fit the LSS self-similarity spectra at low frequencies, but the self-similarity spectra roll off more steeply for frequencies above 1 kHz with the engine at military power and afterburner. Closer to the sideline, the FSS self-similarity spectra do not match the shape of the data spectra very well at high engine power. Possible reasons for these discrepancies are investigated. [Work sponsored by the Air Force Research Laboratory.]

10:50

2aNS9. Multipole representations of directional sources and derivation of directional starters for parabolic equation algorithms. Sergey N. Vecherin, D. Keith Wilson (U.S. Army Engineer Res. Dev. Ctr., 72 Lyme Rd., Hanover, NH 03755, sergey.n.vecherin@usace.army.mil), and Vladimir E. Ostashev (NOAA/Earth System Res. Lab., Boulder, CO 80305)

Although many sources of outdoor sound (including aircraft, ground vehicles, machinery, and loud-speakers) are directional, most numerical methods for sound propagation are formulated for omnidirectional point sources. This paper describes a systematic approach to decomposing a directional source into its spherical harmonics and then representing each harmonic as a multipole (spatially compact collection of omnidirectional point sources). This effective source representation is useful for implementing directional sources in many numerical propagation methods. A new method is also derived for formulating directional starters in parabolic equation (PE) algorithms. The connections between multipole representations and directional starters are described with regard to the angular accuracy of the PE. It is shown that even horizontal multipoles can be appropriately represented with vertical starters, to a degree of accuracy consistent with the PE solution.

11:05

2aNS10. Source directivity in the parabolic equation method using an inverse Fourier transform technique. Joyce E. Rosenbaum, Anthony A. Atchley, and Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., University Park, PA 16802, jer262@psu.edu)

The parabolic equation (PE) method is accurate for prediction of low-frequency noise in the situation that the starting field does not emit significant energy at large elevation angles. The typical PE sound source is assumed to be omnidirectional and represented with a Gaussian starting field, defined at all heights of the grid. A method of representing a source with arbitrary directivity has been developed for the unique form of the PE starting field. By extending the defined vertical wave number spectrum of the arbitrary far-field directional function of the source to include evanescent wave numbers, the approach uses array theory and Fourier transform techniques to define the starting field. The approach recognizes the elevation angle limitation of the PE method and adheres to the accepted wavelength-dependent vertical grid point spacing, satisfying the fundamental requirements of the PE method. Results for horizontal and vertical dipole sources in the free field and above a ground surface are presented and compared with analytical solutions within the valid PE elevation angle range. [This research was funded by the Federal Aviation Administration (FAA) Western-Pacific Region through U.S. Department of Transportation Volpe National Transportation Systems Center.]

11:20

2aNS11. Three-dimensional finite-difference time-domain/ray-tracing hybrid modeling of low-amplitude sonic boom diffraction around large buildings. Sang I. Cho and Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, stc142 @psu.edu)

A hybrid method combining finite-difference time-domain (FDTD) and ray-tracing approaches is used for a 3-D numerical modeling of pressure loading on the exterior surface of large buildings due to an incoming low-amplitude sonic boom. The hybrid simulations model a field test conducted by NASA, where large building structures in Edwards Air Force Base were instrumented to measure their vibroacoustic responses to "low-booms" generated by F/A-18 fighter jets in a special dive maneuver. The low- and high-frequency contents of the low-booms are treated separately by the FDTD and ray-tracing parts of the hybrid model as the accuracy of each approach is restricted to its appropriate frequency limit. The simulation results agree well with the experiment for both single and multiple building cases. The building spiking effect, which describes the positive pressure spikes seen in the boom waveform recorded on the building side wall first impacted by the boom, is observed in each numerical simulation. [Work supported by NASA. The authors appreciate NASA making test data available for this work.]

Session 2aPA

Physical Acoustics: Infrasound

Roger M. Waxler, Chair
Univ. of Mississippi, NCPA, 1 Coliseum Dr., University, MS 38677

Contributed Papers

8:00

2aPA1. Array processing for direction of arrival of infrasound. R. Daniel Costley (U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS 39180, dan.costley@usace.army.mil), W. Garth Frazier (Univ. of Mississippi, University, Mississippi 38677), and Kevin Dillion (A Ducommun Co., Oxford, MS 38655)

Pipe arrays have conventionally been used to detect infrasound. The advantage of this approach is that the arrays have a large aperture over which wind noise can be averaged. In recent years, distributed arrays have been used for the same effect. In addition to reducing wind noise via spatial averaging, these arrays have the ability to process the individual signals from each sensor to estimate direction of arrival (DOA) of acoustic signals and, potentially, the direction of the wind flow. This is especially true for infrasound and low frequency acoustic sources of tactical interest, in the 1–100 Hz range. A frequency-wavenumber (F-K) processing technique has been applied to signals from rectangular infrasound arrays to estimate DOA. This approach will be discussed as well as the dependence of accuracy on signal-to-noise ratio.

8:15

2aPA2. Investigations of wind-noise reduction at the infrasound test bed in Trafelberg, Austria. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., P.O. Box 30, MS 06120D, State College, PA 16804) and Georgios Haralabus (Comprehensive Nuclear-Test-Ban Treaty Organization (IMS/ED/AM), Vienna, Austria)

The International Monitoring System (IMS) of the Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO) has a unique infrasound test site (I99AT) with four wind-noise-reduction pipe-array systems in close enough proximity to allow simultaneous measurements under similar environmental conditions. It was designed as a facility to study the effectiveness of various wind-noise-reduction systems so it is clear of trees, vegetation, or other wind breaks. In May 2010, four infrasonic microphones were deployed at the test bed. These reference microphones contributed additional single-point measurements, which could be taken individually or combined into a small array for comparison to the larger pipe-array systems. The test-bed wind-noise-reduction systems in conjunction with the additional sensors permitted observation of the relationship between sensor aperture and wind-noise-reduction effectiveness as a function of frequency. In particular, one 3-day period had varying wind conditions with very low anthropogenic noise. At the lowest frequencies, the turbulence scale sizes exceeded the largest pipe-array dimensions and all sensors converged to similar values of spectral density; as the frequency increased, the wind noise was suppressed first as a function of overall dimension, then as a function of number of inlets. [Work sponsored by the Comprehensive Nuclear-Test-Ban Treaty Organization.]

8:30

2aPA3. A portable calibration system for infrasound microphones. Chris Watson, Jay Helmericks, and Curt Szuberla (Geophysical Inst., Univ. of Alaska, 903 Koyukuk Dr., Fairbanks, AK 99775)

Infrasound signals have the characteristic that they propagate for long distances in the atmosphere. The far field recording of the signal is a convolution of the characteristics of the signal source, the atmosphere through which it propagates, and the infrasound microphones used to measure the signal. There is an emerging interest in knowing how well the signal source can be characterized using signals recorded in the far field. Characterization of the signal source requires the deconvolution of the recorded signal. Ensuring that the infrasound microphones are operating within manufacturer specifications is a standard way to simplify this process. To that end, Chaparral Physics is developing an economical calibration system that can be used to determine the calibration constants for infrasound microphones using an automated process controlled from a laptop. This paper will detail the mechanical theory associated with the calibration and the software used to collect and analyze the data.

8:45

2aPA4. Calibrating infrasound reference microphones. Thomas B. Gabrielson and Timothy M. Marston (Appl. Res. Lab., Penn State Univ., P.O. Box 30, MS 06120D, State College, PA 16804)

Low-frequency commercial measurement microphones can be used for infrasound measurement well below their nominal low-frequency roll-off as long as the magnitude and phase of the infrasonic response is known. Three methods are used at Penn State for laboratory measurement of this response: (1) an "absolute calibration" in a piston-driven chamber designed for accurate determination of internal pressure based on piston displacement, (2) a "comparison" calibration in the same chamber where the response is compared to the output of a dc-coupled piezoresistive pressure transducer, and (3) a "hydrostatic" calibration based on the pressure change associated with a sinusoidal variation in elevation. For the first technique, a full thermoviscous acoustic model for the chamber is used to bridge the adiabatic-to-isothermal transition and to convert piston volume velocity to pressure. The second technique depends on the calibration of the dc-coupled reference transducer. The third technique depends on the elevation-dependence of hydrostatic pressure. The first and second techniques are practical from 0.001 to 25 Hz; the third technique has a smaller practical frequency range—0.05 to 0.5 Hz—but permits an independent check on the other techniques. [Funded by the US Army Space and Missile Defense Command.]

9:00

2aPA5. *In-situ* calibration of infrasound array elements. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., P.O. Box 30, MS 06120D, State College, PA 16804)

A typical infrasound array element consists of an infrasonic pressure sensor connected to a multiple-pipe or porous-hose system for reduction of wind noise. While the frequency response of the sensor itself may be known, that response is modified by the wind-noise reduction system. One approach to measuring the frequency response of the complete system is to perform a comparison calibration using ambient noise and a co-located reference sensor with sufficiently low self-noise and well characterized frequency response. In the technique discussed here, the reference sensor is a virtual reference constructed by summing the outputs of two or three calibrated microphones; symmetric placement of the microphones places the phase center of the virtual reference at the geometric center of the pipe system. Proper combination of auto- and cross-spectral averages over a several-hour period produces an estimate of the response of the infrasound system relative to that of the virtual reference. Measured coherence and the consistency between the magnitude and the phase of the response provide quality checks on the process. This approach has been demonstrated at infrasound monitoring sites in Washington, Alaska, Manitoba, Austria, and Antarctica. [Funded by the US Army Space and Missile Defense Command.]

9:15

2aPA6. Wind noise minimization for single infrasonic sensors. Jeremy Webster and Richard Raspet (Natl. Ctr. for Physical Acoust., 1 Coliseum Dr, Univ. of Mississippi, University, MS 38677)

One of the main limitations in outdoor infrasound measurements in open areas is the noise generated by turbulent winds. Pipe arrays or arrays constructed using porous hoses are often used to average out the pressure fluctuations from the turbulence. These devices have a limited useful bandwidth, since they also average out the acoustic response at higher frequencies and distort received acoustic waveforms. We have investigated relatively small configurations to reduce wind noise based on our research into low-frequency wind noise over open outdoor surfaces. Such devices may be useful for temporary arrays for which waveform integrity is more important than wind noise reduction at very low frequencies. In this talk, we present results for infrasonic sensors placed bare on the surface, mounted underneath a flush metal foam flat surface, and under two types of hemispherical dome. For reference, these will be compared to and contrasted with wind noise measurements with barriers and pipe arrays previously reported in the literature [J. Acoust. Soc. Am. **114**, 1379–1386]

9:30

2aPA7. Modeling infrasound propagation using the Fourier pseudo-spectral method. Roger Waxler (Univ. of Mississippi, NCPA, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu) and Doru Velea (QinetiQ North America, Technol. Solutions Group, 11091 Sunset Hills Rd., Reston VA 20190)

As part of an undergoing effort to model infrasound propagation in the atmosphere, a pseudospectral time-domain method is employed to solve the linearized equations of thermo-viscous compressible gas dynamics in a two dimensional stratified atmosphere over a flat rigid ground surface. The system of equations is transformed from the spatial to the wavenumber domain using a spatial Fourier transform for the horizontal variables and a generalized trigonometric transform (to account for the ground surface) for the vertical variable. Time evolution is computed numerically in the wavenumber domain using a Runge–Kutta algorithm. In this initial work only viscous and thermal losses are taken into account. To handle the no-slip condition at the ground, acoustic boundary layer theory is used. Numerical results showing the effects of complex meteorology on the the propagation of an acoustic pulse will be presented. [Work supported by US Army SMDC.]

9:45

2aPA8. *In-situ* infrasonic calibrations at Windless Bight, Antarctica. Curt Szuberla, Dave Withoff, Dave Fee, Jay Helmericks (Geophysical Inst., Univ. of Alaska Fairbanks, 903 Koyukuk Dr., Fairbanks, AK 99775-7320), and Tom Gabrielson (Penn State Univ., University Park, PA 16802)

The array elements at the Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO) infrasound station I55US (Windless Bight, Antarctica) are covered with more than a meter of snowfall each year between maintenance visits. During these service trips, in the austral summer, the microphones and their attached wind reduction pipe arrays are brought back to the surface of the Ross Ice Shelf snow. During the November 2010 maintenance visit, an *in-situ* calibration reference microphone system was used in an attempt to characterize the effect of this compacted snow layer on the performance of the wind reducing elements associated with each microphone. In order to accomplish this, measurements were taken before and after the digging at certain elements. Additionally, one element was measured before and after a meter of fresh snow fell on the dug-up sensor and pipes. This presentation will address the basic technique, developed for the CTBTO arrays, our results, and broader potential applications.

10:00—10:15 Break

10:15

2aPA9. Simulation and analysis of infrasound generated by severe storms. David A. Schecter (NorthWest Res. Assoc., 4118 148th Ave. NE, Redmond, WA 98052, schecter@nwra.com) and Melville E. Nicholls (CIRES/Univ. of Colorado, Boulder, CO 80309)

Recent observations and longstanding theoretical considerations suggest that a tornado can have a detectable signature in the infrasound of a severe storm. To reliably distinguish the vortex signal from extraneous noise, it is important to advance current understanding of the different mechanisms that produce infrasound in atmospheric convection. Without detailed observations of the acoustic sources within a storm, numerical modeling may be the best method of investigation. This avenue of research is being explored with the Regional Atmospheric Modeling System customized to simulate aeroacoustics. By comparison to analytical results, it has been shown that the model adequately generates the infrasound of vortices and basic diabatic cloud processes. Given these encouraging results, the model is now being used to simulate the infrasound of various convective systems and individual storm features including a towering cumulonimbus (with multiple hydrometeor categories) and artificially forced tornadoes with realistic structure and fluctuations. This presentation will expound a convenient technique for analyzing the sources of infrasound in a complex storm simulation based on an acoustic wave equation (Lighthill analogy) for the perturbation Exner function. It will also provide some computational evidence that moderate-to-strong tornadoes can emit stronger 0.1–1 Hz infrasound than generic storms. [Work supported by NSF-ATM-0832320].

10:30

2aPA10. On the waveforms of infrasonic returns from the stratosphere. Roger Waxler, Phil Blom, Carrick L. Talmadge, and Jelle Assink (NCPA, Univ. of Mississippi, University, MS 38677, rwax@olemiss.edu)

The stratosphere is said to end at the stratopause, a local sound speed maximum at about 50 or 60 km altitude. This maximum is typically about 320–330 m/s and is not large enough by itself to cause sound produced on the ground to be refracted back to the ground. There is generally a strong wind jet at the stratopause, the stratospheric jet, which provides sufficient additional refraction to return sound to the ground downwind from a source. In the absence of other effects, stratospheric returns have a universal form. The closest point of return is a caustic of the wave field at which there is a single arrival. Further out the signal splits into two arrivals: a higher-angle "fast" arrival and a lower-angle "slow" arrival. The fast arrival preserves the phase of the source while the slow arrival undergoes a 90 deg phase shift. Deviations from this behavior come predominantly from two sources: wind jets in the lower 15 km of the atmosphere and variations within the stratospheric jet. Illustrations from both data and models will be shown.

10:45

2aPA11. On the sensitivity of infrasound to the upper atmosphere. Jelle D. Assink, Roger Waxler (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jdassink@olemiss.edu), and Doug Drob (Space Sci. Div. Naval Res. Lab., Washington, DC)

In comparison to the lower atmosphere where comprehensive global atmospheric specifications are commonplace, measuring the properties of the atmosphere above the stratosphere is an active area of scientific research. Here, we revisit the use of infrasound as a remote sensing technique for the upper atmosphere. Infrasonic signals from the Tungurahua volcano in Ecuador are used to investigate the behavior of the upper atmosphere. Depending on the atmospheric conditions, stratospheric, mesospheric, and thermospheric arrivals are observed during intervals of explosive volcanic activity. It is found that the travel times of the thermospheric arrivals exhibit a coherent variability with periods equal to those of the harmonics of the atmospheric tides. Theoretical predictions using atmospheric specifications show that the stratospheric arrivals are well predicted but that significant discrepancies exist for arrivals from higher altitudes. As such, these observations suggest a readily accessible means of passive atmospheric remote sensing that can be utilized in conjunction with other techniques as well as *a-priori* information to routinely measure and specify the state of the upper atmosphere.

11:00

2aPA12. A large scale portable infrasound network as upper atmospheric monitor. Jelle D. Assink, Roger Waxler, Carrick Talmadge, Phil Blom (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jdassink@olemiss.edu), and Doug Drob (Space Sci. Div. Naval Res. Lab., Washington, DC)

Analysis of data from this summer's large scale infrasound array deployment in the American West will be presented. The deployment consisted of an arc of arrays at a fixed range of 220 km and a northward line of arrays from 220 to 400 km. In this presentation particular attention will be given to data from the northward line. Tropospheric, stratospheric, and thermospheric arrivals were observed. The evolution with range of the arrivals' propagation parameters and waveforms will be discussed and compared with theoretical predictions that make use of atmospheric specifications. The data input in these specifications decreases with height; the specifications above the stratosphere are based on semi-empirical models rather than observations. Naturally, the largest deviations are found in the predictions of thermospheric arrivals at the various arrays. By reducing the misfit between predictions and observations, we work on updating upper atmospheric wind models using infrasound.

11:15

2aPA13. Utah Testing and Training Range 2010: Large scale infrasound sensor deployment to Utah and Idaho. Carrick L. Talmadge, Roger Waxler, Daniel E. Kleinert, Jr. (NCPA, Univ. of Mississippi, University, MS 38677), Sue Nava (ENSCO, Inc., Melbourne, FL 32940), Jelle Assink, and Hank Buchanan (Univ. of Mississippi, University, MS 38677)

A large scale infrasound deployment was undertaken during the summer of 2010 using large blasts (19.5-tons TNT, 8.8-tons TNT, and 1.9-tons TNT) that were taking place at the Utah Testing and Training Range west of Salt Lake City, UT. A total of four sensors were deployed in the near field and additional 114 infrasound sensors were placed in the "far field." Six arrays of five elements each were delayed along a "stratospheric arc" 220-km from the source from a bearing of 300° to 355° in 5^* increments. Additional six arrays (3 with three elements and 3 with six elements) were deployed in a northern "thermospheric" line, designed to separate the thermospheric line from the stratospheric one. Data will be reported on measurements from 24-explosions. The implications of these measurements on empirical models of atmospheric structure (e.g., G2S) will be discussed.

11:30

2aPA14. Auroral infrasound associated with ionospheric electron precipitation. Justin Oldham, Charles R. Wilson, John V. Olson, and David Fee (Geophysical Inst., Univ. of Alaksa Fairbanks, 903 Koyukuk Dr., Fairbanks, AK 99775)

Periods of persistent, high-trace velocity infrasound signals lasting several hours or more have been observed in the data from the CTBT/IMS I53 infrasound station in Fairbanks, AK. Early studies showed that the infrasound activity was associated with times of geomagnetic disturbance. The imaging riometer at the Poker Flat Research Range near Fairbanks collects cosmic noise absorption (CNA) data from an array of 16×16 beams that cover a region that is approximately 200×200 km^2 in the overhead ionosphere. Also available from the PFRR site are magnetometer data and all-sky camera videos showing auroral events when the sky is dark. The CNA event data give direct evidence of the precipitation of high energy electrons into the D-region of the lower ionosphere that produce both auroral displays captured in all-sky images and disturbances in the ground magnetometers. The simultaneous occurrence at I53US of high trace-velocity infrasonic waves, geomagnetic perturbations, and cosmic radio noise absorption provides verification that the observed infrasound is generated in the lower ionosphere. Examples of the high-trace velocity signals along with the CNA and magnetometer data will be shown.

11:45

2aPA15. Porosity effects in large wind screening structures. JohnPaul R. Abbott, Richard Raspet, and Jeremy Webster (Natl. Ctr. for Physical Acoust., Dept. of Phys. and Astr., Univ. of Mississippi, Coliseum Dr., University, MS 38677, jrabbott@olemiss.edu)

Recently completed research indicates that the dominant source of wind noise for infrasonic microphones placed on open ground is the interaction pressure of the average wind velocity profile and the vertical turbulence. The source region at infrasonic frequencies is quite large and therefore can only be reduced with relatively large structures. It is hypothesized that such a structure does not need to completely block the wind from the source region; it only has to reduce the wind velocity gradient above the microphone and perhaps modify the turbulence field. If so then a porous wind barrier or screen may be used. To investigate this hypothesis a 3 m high decagonal wind barrier with variable porosity has been constructed. Initial results indicate that reductions are strongest for intermediate porosities and for frequencies between 0.5 and 7 Hz. In addition, a near constant 3–5 dB reduction was also observed for frequencies below 0.5 Hz. The noise reduction results will be related to the measured wind profiles and turbulence spectra inside the enclosure. The enclosure is quite open and should have little effect on acoustic waveforms, unlike pipe arrays and porous hose arrays.

Session 2aPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Comparative Approaches to Peripheral Auditory Function

Christopher Bergevin, Chair
Columbia Univ., Otolaryngology, Head and Neck Surgery, 630 W. 168th St., New York, NY 10032

Chair's Introduction—8:00

Invited Papers

8:05

2aPP1. Lizard ears and simple solutions to auditory coding. Geoffrey Manley (Cochlear and Auditory Brainstem Physio. IBU, Faculty V. Carl von Ossietzky Univ. Oldenburg, 26111 Oldenburg, Germany)

The variety of lizard auditory organs provides an optimal substrate for examining structure-function relationships. What does an auditory organ need to perform the basic tasks in the coding of acoustic stimuli? The single-ossicle tympanic middle ears of lizards are connected through the head, providing for sensitive directional hearing with little neural processing. Lizards evolved hair cells whose frequency responses were determined by the morphological details of their stereovillar bundles and tectorial material, enabling a tonotopic arrangement of frequency sensitivity. Modern lizards achieve this with few hair-cell rows, showing that tonotopicity does not demand an extensive epithelium. Finally, different lizard families achieved dissimilar frequency selectivity through manipulations of anatomical features, especially of the tectorial membrane. There is a pay-off between the size of the auditory papilla and its ability to code different frequencies. Long papillae allow very local coupling of hair-cell bundles with very selective frequency tuning. Short papillae generally abandon tectorial coupling to allow for useful tuning but this results in poorer sensitivity and poorer tuning selectivity. Lizards demonstrate that sensitive and selective hearing organs need only a few hundred hair-cells organized along a straight axis. [work supported by the German Research Foundation DFG (MA 871-10.)]

8:30

2aPP2. The teachings of anurans: Mechanisms of auditory processing in the frog. Sebastiaan W. F. Meenderink (Dept. of Neurosci., Erasmus MC, P.O. Box 2040, NL-3000 CA Rotterdam, The Netherlands, swfmeenderink@yahoo.com)

Amphibians are one of the four classes of tetrapod vertebrates. The majority of amphibian species are anurans, better known as frogs and toads. Frog hair cells have served as a model for studying fundamental cellular processes including electrical tuning, forward and reverse transduction, and "cochlear" amplification. In most frogs, these processes act at much lower frequencies compared to the mammalian cochlea, and extrapolation of results to higher-frequency epithelia must be made with caution. The gross anatomy of the anuran inner ear is unique among vertebrates in that it possesses two distinct organs specialized in the detection of airborne sounds. Moreover, there is no analog of the basilar membrane in these organs; hair cells are situated directly over the stationary, cartilaginous labyrinth. These structural differences are reflected in how sounds propagate within the inner ear. Specifically, traveling waves seem absent in frog. Despite this, otoacoustic emissions in anurans and mammals share many characteristics, and their comparative study may help to elucidate universal mechanical properties of the inner ear. These examples, combined with numerous other types of experiments, have made the frog a useful substrate for the examination of cellular mechanisms, transduction, and macro-mechanical inner-ear properties underlying normal vertebrate hearing.

8:55

2aPP3. Why bother looking at a range of species? Glenis R. Long (Speech-Lang.-Hearing Sci. Program, Graduate Ctr. of the City, Univ. of New York, 365 Fifth Ave., New York, NY 10016)

Most anatomical and physiological researches are conducted on a small subset of mammals, and there are many pressures on auditory researchers to limit their research to a limited number of species with the assumption that this will permit us to explain normal and impaired hearing in humans. Otoacoustic emissions (OAEs) provide a particularly useful comparative tool because very similar measures can be efficiently obtained from a range of species. A full understanding of OAEs in the ear canal can be enhanced when simultaneous measurements are obtained both in the ear canal and within the cochlea. This paper will provide evidence from psychoacoustic, evoked potential and OAE research that comparative research is essential if we are to fully understand human auditory processing.

9:20

2aPP4. Otoacoustic emission delays as a probe to measure cochlear tuning: Comparative validations. Christopher Bergevin (Dept. of Otolaryngol./Head Neck Surgery, Columbia Univ., 630 West 168th St., PS 11-452, New York, NY 10032, cb2811@columbia.edu)

The ear is not only sensitive to sound but selective as well: Tonotopic tuning of the inner ear provides a means to resolve incoming spectral information. Measurements of frequency selectivity have traditionally relied upon either subjective psychophysical or objective (but invasive) physiological approaches. Sounds emitted from a healthy ear, known as otoacoustic emissions (OAEs), have been proposed to both objectively and non-invasively estimate peripheral auditory tuning. Despite diverse inner-ear morphological variation across animals, OAEs are a universal feature and correlate well to an animal's range of *active* hearing. Recent studies focusing on emission delays in response to a single *stimulus frequency* (SFOAEs), conducted systematically across species in a variety of classes (mammals, aves, reptiles, and amphibians), support predictions relating emissions and tuning. Longer SFOAE delays presumably reflect

the sharper tuning associated with resonant build-up time of the underlying auditory filters. Differences in tuning estimated from OAEs appear generally congruous with known anatomical and functional considerations: Larger sensory organs (i.e., more "filters") with smaller ranges of audition exhibit sharper tuning. Comparisons made both broadly (inter-class) and within phylogenetically-matched groups (intra-family) indicate that SFOAE delays in humans are longer than any other species so far examined, suggestive of exceptionally sharper tuning.

9:45—10:00 Break

10:00

2aPP5. Comparative cat studies: Are tigers auditory specialists? Edward J. Walsh (Developmental Auditory Physio. Lab., Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, edward.walsh@boystown.org), Douglas L. Armstrong (Henry Doorly Zoo, 3701 S. 10th St., Omaha, NE 68107), and JoAnn McGee (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131)

Although representatives of the 41 extant cat species inhabit nearly every biome on the planet and may have faced a highly diverse set of selection pressures during their evolution, relatively little effort has been made to compare commonly measured features of peripheral auditory function among species representing the family. Given their extensive geographic range, it is reasonable to suggest that auditory system adaptations may have occurred, leading to functional specialization in a subset of species. In that light, frequency-threshold curves and response latency-level and latency-frequency relationships will be compared in cats of widely varying body mass inhabiting a variety of habitats. The body mass of felids spans a range of more than two orders of magnitude, with small cats like the desert sandcat, *Felis margarita*, weighing as little as 2 kg and the Amur tiger, *Panthera tigris altaica*, weighing as much as 300 kg. While most cats studied thus far appear to satisfy the conditions necessary to be labeled auditory generalists, the tiger, and perhaps members of the *Panthera* genus generally, may be exceptions. We will consider the possibility that big cats are auditory generalists with regard to acoustic sensitivity, but exhibit a peripheral specialization affecting low-frequency neural latencies. [Funding provided by NSF Grant No. 0823417].

10:25

2aPP6. Interaural time difference processing in birds and alligators: Evolution of binaural circuits. Catherine E. Carr (Dept. of Biology, Univ. of Maryland, College Park, MD 20742, cecarr@umd.edu)

The auditory systems of birds and mammals use timing information from each ear to detect interaural time differences (ITDs). In birds, the circuits that encode ITDs are composed of delay lines and coincidence detectors. To determine if these circuits are evolutionarily conserved, we have compared the physiological and anatomical organizations of the auditory nuclei of birds and their sister group, the crocodilians. In both groups, precisely timed spikes in the first order nucleus magnocellularis encode the timing of sounds, and NM neurons project to neurons in the nucleus laminaris that detect interaural time differences. NL neurons act as coincidence detectors, and encode ITDs in both birds and crocodilians. In the crocodilians, however, the range of best ITDs represented in NL was larger than in birds, possibly because of the network of canals that connect the middle ear spaces. These interaural canals are also found in birds, and, for low frequency sounds, may in both groups provide a larger range of ITDs than predicted by actual head size.

Contributed Paper

10:50

2aPP7. Response properties and local connectivity of the cochlear nucleus of the big brown bat. Andrea M. Simmons (Dept. of CLPS, Brown Univ., Providence, RI 02912, andrea_simmons@brown.edu), Seth S. Horowitz, Jonathan R. Barchi, and James A. Simmons (Brown Univ., Providence, RI 02912)

We characterized responses of the cochlear nucleus (CN) in big brown bats using local field potentials and extracellular unit responses to tone bursts, forward and reversed FM sweeps, and pulse-echo pairs. Neurobiotin-filled pipettes were lowered through the inferior colliculus of anesthetized bats to depths from 3.2 to 4.3 mm based on stereotaxic coordinates. Response properties were correlated with local anatomical connectivity as shown by transport of neurobiotin from the recording site and immunohistochemical localization of GABA and of connexin 35/36 in alternate brain sections. In dorsal AVCN, response properties were similar to those seen in the nuclei of the lateral lemniscus. Recording sites in ventral AVCN near the insertion point of the auditory nerve showed unique characteristics, including very short latencies (1.5–3 ms) and oscillatory responses extending past the duration of the sound. Most sites were sharply tuned to low frequencies corresponding to the first harmonic of the FM sweep, and responded equally well to forward and reverse FM sweeps. Data indicate that early stages of auditory processing in the big brown bat are critical to echolocation hyperacuity and may be dependent on anatomical specializations of the AVCN. [Work supported by the NIH, ONR, and RI Space Grant (NASA).]

11:05—12:00 Panel Discussion

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics, and Noise: Advances in Vibroacoustic Treatments for Vehicles

Benjamin M. Shafer, Cochair
Serious Materials, 1250 Elko Dr., Sunnyvale, CA 94089-2213

Micah Shepherd, Cochair
Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Invited Papers

8:00

2aSA1. Aircraft cabin noise control: Challenges and opportunities. Krishna Viswanathan, Herb Hoffman, and Alex Lin (The Boeing Co., M/S 0R-JF, P.O. Box 3707, Seattle, WA 98124)

Increasingly stringent limits on community noise from aircraft operations have been imposed over the past three decades. However, no comparable limits have been in place for cabin noise levels until recently. The introduction of regulations by the European Union on the allowable levels for noise exposure inside the aircraft cabin for pilots, flight attendants, and passengers has brought this problem to the forefront. Several noise sources contribute to high noise levels inside the cabin; the dominance of any particular source varies depending on the location. In addition to the noise due to the turbulent boundary layer, engine vibrations, and equipment, the noise from the engines plays an important role. The nozzles of a dual-stream turbofan engine are operated invariably at super-critical pressure ratios at cruise, thereby producing shocks in the jet plume. Consequently, broadband shock-associated noise is generated in addition to the turbulent mixing noise component. The engine noise impinges on the fuselage and is transmitted into the interior of the aircraft cabin. Control strategies typically focus on appropriate designs for reducing source noise and optimizing acoustic treatment, as the added weight is undesirable. This talk provides an overview and concrete examples for noise control.

8:20

2aSA2. Cockpit door module for A380 and AXP A340 aircraft. Mike Dickerson, Jr. and Michael Dickerson, Sr. (MD Acoustics, 1107 Rambling Rd., Simi Valley, CA 93065)

The cockpit door module for the A380 and A340 aircraft was assessed for specification compliance. Per the manufactures specifications, the noise/vibration levels exceeded performance specifications at various frequencies during the testing phases. Each noise contributor in the overall system was identified evaluated and assessed. The real world practical application of active/passive noise control was utilized in a superposition system. Due to strict FAA requirements, multiple paths of mitigation measures were explored. Mitigation measures include product redesign, material design, absorptive material, and constrained layer damping technology. Noise measurements were conducted to assess the results of the migration efforts. The data and results of this assessment are presented and discussed.

8:40

2aSA3. Reconstruction of normal surface velocities on a baffled plate using Helmholtz equation least squares method. Logesh Kumar Natarajan and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detoit, MI 48202)

This paper provides comprehensive experimental validations of the Helmholtz equation least squares (HELS) method for reconstructing the vibro-acoustic responses of a highly non-spherical object such as a baffled square plate. This study highlights that a computationally simple HELS code based on the spherical wave functions can accurately and effectively reconstruct vibro-acoustic responses of a planar surface. The key to a successful reconstruction is to select an optimal origin of the coordinate system behind the planar surface, and a hybrid regularization scheme. Tests are conducted inside an anechoic chamber and plate is excited by a point force. The radiated acoustic pressures are measured using a planar array of microphones at very close distance to the surface. The reconstructed normal surface velocities are compared against the benchmark data measured by a laser vibrometer. The reconstruction effectiveness is analyzed by comparing the reconstructed operation deflection shapes at the natural frequencies against the theoretical natural modes and the measured resonance modes. The results revealed that the HELS codes, when coupled with a hybrid regularization procedure through a modified Tikhnov regularization and least squares method, can yield the very accurate reconstruction of the vibro-acoustic responses of a vibrating plate in both amplitudes and surface distributions.

9:00

2aSA4. Vibration isolation by frequency and temperature sensitive elastomers. Giora Rosenhouse (Faculty of Civil Engng., Technion, 89 Vagalil Str., Haifa 32684, Israel)

Damping is the most important factor in vibration isolation due to dissipating mechanical energy into heat, energy absorption under impact, imposing finite amplitude bound amplitudes at resonant frequencies, and mitigating vibration amplitudes in the vicinity of those frequencies while increasing the amplitudes in other frequency domains. Thus, many kinds of available elastomers of high damping ratio are in common use as antivibration materials. The choice of polymer depends on the specification (loads and environmental conditions) and the isolation purpose. Most of these commercial materials properties were tested and defined by specific viscoelastic models. Also, their properties depend on frequency, temperature, and time that can be measured following test regulations. This dependence leads to the dynamic response functions of single or multi-degree of freedom systems (SDF or MDF systems) that differ in shape from those of systems with independent (constant) parameters. A solution is developed here to response function of an SDF system with frequency and temperature dependent elasticity and loss factor that can be expanded for MDF systems analysis. Consequently, the sensitivity of parameters of typical elastomers leads to large changes in the response function that should be taken into account in designing elastic mountings and vibration isolation.

9:15

2aSA5. An integrated reception-plate inverse-force structure-borne noise source characterization test method. Kevin H. Lai (Boeing Commercial Airplanes, P.O. Box 3707, MC 0R-JF, Seattle, WA 98124-2207)

An integrated reception-plate inverse-force method for vibration power and force transmitted from a mounted vibration source measurement has been prototyped and evaluated. The tests developed data to aid in understanding the repeatability and reproducibility of the method, which is necessary as these methods are integrated into industrial noise control design applications. The reception plate has many advantages such as mounting structure mobility matching, simple source-receiver dynamics and inexpensive reproducible test setups. The inverse force method has the advantage estimating of the blocked force characterization *in situ*. Implementing the inverse method on a reception plate allows the use of a common set of accelerometers measurements. The accelerometers measurements are processed for plate flexural response to estimate input power using reception plate methods and transfer mobility measurements for the blocked force characterization method. The goal of this work is to development of an industrial engineering-tolerance cost-effective, vibration source characterization method for the specification of acceptable source levels of equipment.

9:30

2aSA6. On the measurement of angular dependent sound transmission through airborne, supercritical plates. Matthew D. Shaw and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mdshaw16@gmail.com)

Angular dependence of sound transmission through airborne plates, at frequencies above their critical frequencies (supercritical), is discussed. Experimental measurements of this angular dependence have been conducted using phased array methods. Theory governing the sound transmission through supercritical plates is already well developed; however, experimental results are known to depart from theory near grazing incidence. The details of the measurement apparatus and issues involved in making this type of measurement will be discussed. The results of the measurements will be compared to theory, focusing on the departure of experiments from theory.

9:45—10:00 Break

10:00

2aSA7. Analytical modeling of the acoustic field during a Direct Field Acoustic Test. Jerry W. Rouse, Eric C. Stasiunas, and Mikhail Mesh (Sandia Natl. Labs., P. O. Box 5800, 87185-0346, jwrouse@sandia.gov)

The acoustic field generated during a Direct Field Acoustic Test (DFAT) has been analytically modeled in two space dimensions using a properly phased distribution of propagating plane waves. Both the pure-tone and broadband acoustic field were qualitatively and quantitatively compared to a diffuse acoustic field. The modeling indicates significant non-uniformity of sound pressure level for an empty (no test article) DFAT, specifically a center peak and concentric maxima/minima rings. This spatial variation is due to the equivalent phase among all propagating plane waves at each frequency. Predicted spatial variation is shown to agree well with experimental measurements. The excitation of a simply supported slender beam immersed within the acoustic fields was also analytically modeled. Results indicate that mid-span response is dependent on location and orientation of the beam relative to the center of the DFAT acoustic field. For a diffuse acoustic field, due to its spatial uniformity, mid-span response sensitivity to location and orientation is nonexistent. Extension of the modeling to three space dimensions and numerical methods is underway.

10:15

2aSA8. Experiences in performing a high-intensity, direct-field acoustic test on a contamination sensitive system. Eric C. Stasiunas, Troy J. Skousen, and Vit Babuska (Structural Dynam. Res. Dept., Sandia Natl. Labs., P.O. Box 5800, MS-0557, Albuquerque, NM 87185)

A direct-field acoustic test (DFAT) was performed on a Sandia system in order to verify survival due to an acoustic environment of 146.7 dB. The DFAT technique—performed by surrounding a test article with a wall of speakers and controlling the acoustic input with a closed-loop control system—was chosen as the test method in order to meet a critical schedule. In choosing this test method, other challenges became apparent, such as how to obtain the high-intensity acoustic levels and what occurs to that environment inside the bagged frame constructed to maintain a contamination-free system. In addition, the vast amounts of data measured during a single test necessitated a way for the test director to quickly visualize the acoustic environment, saving time and provide insight for input adjustments if necessary. Finally, even though the specified acoustic environment was successfully obtained, the results illustrated some drawbacks of the current DFAT method. This paper will detail the DFAT setup used to obtain the test specification, the effects of the contamination frame on the acoustic environment, the quick-look data program created for visual analysis of the acoustic field, and ideas for performing more diffuse DFAT tests in the future.

10:30

2aSA9. Harvesting ambient acoustic energy using acoustic resonators. Bin Li and Jeong Ho You (Dept. of Mech. Eng., Southern Methodist Univ., P.O. Box 750337, Dallas, TX 75275-0337)

There is currently a high demand for the development of powering systems to supply electricity to small wireless electronic devices. This has led to various energy harvesting concepts aimed to harvest potential energy sources, such as solar, thermal, wind, and vibration/wave energies. Harvesting acoustic energy has been rarely investigated because of its lower energy density compared with other resources. Even though a single acoustic wave has the lowest energy density among the above candidates, sound is abundant in our everyday life and is currently wasted. Therefore, harvesting sound energy would be a good alternative to already existing other energy harvesters. In this work, a computational model for harvesting acoustic energy using a Helmholtz resonator will be presented. To convert acoustic en-

ergy to electricity, piezoelectric wires are placed inside a Helmholtz resonator. When an external sound excites the resonator at acoustic resonant frequencies, the amplified acoustic pressure is developed and generates the vibration motion of the piezoelectric wires resulting in electricity generation. Fundamental mechanisms coupling the electrical power generation by acoustic resonant behavior of Helmholtz resonators will be presented. The effects of geometries of Helmholtz resonators and piezoelectric wires on electricity generation have been studied numerically and will be presented.

10:45

2aSA10. Radiation loss in Pelamis-like ocean wave energy conversion devices. Amadou G. Thiam and Allan D. Pierce (Dept. Mech. Eng., Boston Univ., 110 Cummington St., MA 02215, thiam@bu.edu)

At the previous ASA meeting, Peter Rogers suggested that the loss by re-radiation imposes limitations on the energy that can be extracted from incoming ocean waves. In an attempt to answer this question, the authors consider the canonical problem of an infinitely long flexible circular cylinder floating on the surface of an infinitely deep ocean. The cylinder is assumed to be externally forced into heaving oscillations at fixed frequency, and the phasing of the oscillations corresponds to a traveling wave with a fixed wavelength, which is not necessarily the wavelength of the radiated water waves. The entrained mass and the energy radiated per unit length of the cylinder is derived in the low frequency limit by the method of matched asymptotic expansions. The derived analytical results are used in the approximate modification of the multi-degree-of-freedom model of a finite-length multi-segmented cylinder vibrating under the influence of an incoming ocean wave. The influence of radiation losses are significant if the overall length of the device is much greater than the wavelength of the incoming water wave.

11:00

2aSA11. Sound reduction of a vibrating plate via dimpling design. Nabeel Alshabatat, Kyle Myers, and Koorosh Naghshineh (Dept. of Mech. Eng., Western Michigan Univ., 1903 West Michigan Ave., Kalamazoo, MI 49008)

The vibroacoustic performance of plates can be altered by different methods known as structural dynamic modification methods. One of these methods is the shape modification of plate surface. In this study, a method for reducing the sound radiation of vibrating plates is presented. The method based on creating dimples on the surface of a plate. The dimple can be defined as a local modification in the surface of a structure, which has a spherical shape. The dimples change the local stiffness of the plate without changing its mass. Also, they alter the mode shapes so as to reduce sound power, i.e., the radiated sound at some modes is inefficient due to low net volume velocity. The design method couples finite element method, for predicting plate vibration response, with optimization technique based on a genetic algorithm to minimize sound power radiation. The sound radiation is minimized either at a single frequency, or over a broad frequency band. The results show that dimples forming on the plate can achieve effective reductions in radiated sound power.

11:15

2aSA12. A new approach for material characterization of anisotropic plates using ultrasonic guided waves. Jalil Jamali (Shoushtar Campus, Islamic Azad Univ., Shoushtar, Iran, ja_ja032@yahoo.com)

Nondestructive testing methods play an important role in physical characterization of materials and in assessment of their quality and serviceability in structures. Ultrasonic NDT methods are effective instruments for evaluation of elastic modulus, stiffness, and other essential parameters which are vital for analysis and design of structures. In this paper, a novel method for material characterization of anisotropic plates using ultrasonic guided wave is presented. For this purpose, the ultrasonic guided wave propagation in anisotropic plate structures is modeled and the parameters affecting the symmetric and antisymmetric modes are studied. To investigate reliability and efficacy of the proposed approach, the theoretical and experimental results are compared for a thin carbon fiber reinforced plate.

TUESDAY MORNING, 24 MAY 2011 METROPOLITAN B, 8:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Speech Production (Poster Session)

Bryan W. Gick, Chair
Univ. of British Columbia, Dept. of Linguistics, 2613 West Mall, Vancouver, VC, V6T 1Z4, Canada

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

2aSC1. An acoustic description of Maku vowels. Rebekka Puderbaugh and Benjamin V. Tucker (Dept. of Linguist., Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, puderbau@ualberta.ca)

Maku is believed to have originated in Venezuela and migrated into Brazil in the 19th century though there are no longer believed to be any living speakers. Field recordings of a single adult male speaker made in 1953 and 1965 by Migliazza [1978] are used in the current project for an acoustic analysis of the phonemic vowel inventory of Maku. Early impressions of the vowel space indicate five to seven vowels: /i, ɨ, u, e, o, a/ as well as the areally unique /y/, with possible length and nasal/oral vowel distinctions. All vowels and their surrounding context in the original recordings were identified and coded. Vowel duration and values for the first three formants at 25%, 50%, and 75% through the vowel were extracted. Vowel categories were modeled from formant values using conditional inference trees [Hothorn *et al.* (2006)]. These models showed the likelihood of five vowel categories on the basis of the formant measures as well as the length of the vowel segment. Additional analyzes focus on consonantal context, word position, stress and pitch. The acoustic data are also used to reconstruct the vowel space from the bottom up without the influence of investigators' perceptual judgment.

2aSC2. Effects of morpheme type on Deg Xinag ejectives. S Hargus (Dept. of Linguist., Univ. of Wash., Box 354340, Seattle, WA 98195-4340, sharon@u.washington.edu)

Stem-initially in Deg Xinag (Athabaskan), there is a three-way contrast between voiceless unaspirated, voiceless aspirated and ejective stops, and affricates at a variety of places of articulation, including alveolar with lateral release. In prefixes, there is a single lateral ejective affricate, /tɬ'/, historically from *h-s-ɬ-, [Leer (2000)]. In a previous acoustic study [Hargus (2008)], the prefixal lateral affricate had relatively few of the characteristics of ejectives in stems (VOT, f0 perturbation, jitter perturbation, rise time, and normalized fricative energy measures). However, it was not clear whether the reduced number of ejective characteristics of the prefix was a reflection of its evolution from non-ejective *h-s-ɬ- or an expected effect of morpheme type (presumably via concomittant differences in stress). In a follow-up study, the ejective characteristics of a prefixal retroflex ejective affricate (first person plural subject) are compared to those of retroflex ejective affricates in stems. Retroflex ejectives in stems have significantly longer VOT, slower rise time, lesser normalized fricative energy, and/or differences in normalized f0 relative to the prefix. This suggests that the variability of the prefixal lateral ejective affricate is an expected effect of morpheme type, and not a stage in the evolution to ejective. [(Work supported by NSF.)]

2aSC3. Vowel length in Luganda. Irene Vogel (Linguist. and Cognit. Sci., 46 E Delaware Ave., Newark, DE 19716) and Laura Spinu (Concordia Univ., 1455 de Maisonneuve West, Montreal H3G 1M8, Canada)

Luganda, a language of Uganda, is like other Bantu languages in exhibiting a contrast between short and long vowels. In many positions, however, this contrast is neutralized, for example, before NC clusters, where it is reported that only long vowels appear. This paper investigates the extent to which such long vowels, which are also nasalized, are similar in duration to contrastively long vowels. Based on recordings of four native speakers of Luganda living in the US (two female), we compare the durations of vowels in the following contexts: (i) vowels before NT and ND clusters (N = nasal, T = voiceless stop, D = voiced stop); (ii) short vowels before T, D, N; and (iii) long vowels before T, D, N. Analysis is in progress and will also examine the acoustic properties of different vowels. Preliminary results show that the overall duration patterns of Luganda are consistent with those reported for several other Bantu languages. That is, vowels are indeed lengthened before NT/ND clusters; however, their average duration is somewhat shorter than that of contrastive long vowels (i.e., V: 149 ms; V before NC: 298 ms; V: 392 ms). Implications for the implementation of the vowel length contrast are discussed.

2aSC4. Unsupervised learning of linguistic tone systems from field-recorded audio. Joshua S. Hou (Kiha Software, 505 5th Ave. S., Ste. 310, Seattle, WA 98104)

This study presents a system that uses unsupervised learning algorithms on field recorded audio data of a tone language and induces the underlying tone system from the data. Most of the previous work on tone and machine learning has focused on supervised learning of tone. This requires large amounts of annotated data, which exists only for major languages. Unsupervised learning opens up the possibility of automatic tone recognition without the large amounts of training data needed for supervised learning. A number of studies have been done on unsupervised learning of tone. However, these studies used learning algorithms which assume prior knowledge of the number of tonal categories in the data. This study presents a proof-of-concept system that uses a clustering algorithm that does not require prior knowledge of the number of tonal categories and describes the experiments that demonstrate the efficacy of these methods on audio data with field recording quality.

2aSC5. Aspiration and tone. Yuwen Lai and Michi Chen (Dept. of Foreign Lang. and Lit., 1001 Univ. Rd., Hsinchu, Taiwan 300, Republic of China, yuwen.lai@gmail.com)

The perturbation effect of prevocalic aspiration on fundamental frequency (F0) was investigated in Mandarin. Previous research has shown conflicting results regarding the effect of aspiration on F0. The present study aims to develop a rigorous experimental design to test possible modulating factors which gave rise to the controversy. Forty minimal pairs contrasting in prevocalic aspiration across three places of articulation from four Mandarin tones were recorded in a carrier sentence. F0 of the following vowel was measured at the onset and every 10% of the tonal contour. Effect of independent variables including aspiration (aspirated and unaspirated prevocalic stops), gender (male and female), and speaking rate (fast and slow) on F0 was evaluated. Preliminary results from 20 native Mandarin speakers (10 females, 10 males) reveal significant main effects of aspiration (higher F0 after aspirated stops), gender (higher F0 in female), and speaking rate (higher F0 in fast mode). More interestingly, interactions of aspiration and gender as well as aspiration and speaking rate are also significant. The effect of aspiration and its interaction with intrinsic F0 of speakers and speaking rate are discussed.

2aSC6. Tonal variations in Anong. Ela Thurgood (Dept. of English, CS4 Chico, 400 W. First St., Chico, CA 95929)

This study presents an acoustic analysis of Anong tones. It has three aims: (1) to present some detailed documentation of instrumental-acoustic data on citation tones, (2) to compare the acoustic features of the citation tones with the same tones used in disyllabic utterances, and (3) to analyze tones in disyllabic utterances. Anong citation tones are characterized by three features. First, the tonal space is small. Second, onset consonant type correlates with the pitch height of the following vowel. Third, non-modal phonation is not an acoustic cue to any of the five tones. Neither is it a contrastive property of vowels. In disyllabic utterances, the F0 shapes and heights on the first-syllable tones can be readily related to the F0 shapes and heights in the citation forms. The F0 value of the second-syllable tones depends on the offset value of the tone in the first syllable. However, disyllabic forms with the prefix /a31/ follow a different pattern, one in which the first-syllable tone is raised. This pattern correlates with vowel laryngealization and vowel duration.

2aSC7. Vowel devoicing in Mandarin Chinese spoken in Taiwan. Setsuko Shirai (School of Business, Aoyama Gakuin Univ., 4-4-25 Shibuya, Shibuy-aku, Tokyo)

The results of the research on vowel devoicing in Mandarin Chinese spoken in Taiwan are presented. In this research, 30 Taiwanese who have lived in Taipei area at least 10 years are served as subjects. They read the sentences written in traditional Chinese, whose final syllables consist of voiceless fricatives or aspirated affricates and high vowels. The results agree with the previous study conducted by Shirai *et al.* The vowels carry tone 3 (rising and falling tone) tend to undergo devoicing. In addition, some final vowels with frication in high frequency, also known as syllabic sibilants, are partially devoiced. The devoicing of syllabic sibilants is discussed. Furthermore, vowels in penult syllables undergo devoicing. Whether devoicing in Mandarin Chinese is delted or replaced is also discussed.

2aSC8. Airflow and duration patterns in Tunisian Arabic fricatives. Ian Maddieson and Rachid Saghrouni (Dept. of Linguist., Univ of New Mexico, Albuquerque NM 87131-1196)

Airflow and duration patterns in Tunisian Arabic fricatives were examined in data from six male speakers. TA is conventionally described as having eight fricative phonemes, five voiceless (labio-dental, alveolar, uvular, pharyngeal and "glottal") and three voiced (dental, alveolar, and pharyngeal). These segments occur as singletons and geminates. Tokens of both types were collected in word-medial position following an initial /Ca/ sequence, where C is a single stop and /a/ a short low back vowel. Duration of this vowel preceding singleton and geminate fricatives was compared with its duration preceding a medial fricative + consonant (/FC/) cluster, where a syllable boundary is assumed to occur between /F/ and /C/ (e.g., /kafara, kaffara, kafra/). Vowel duration before the geminate is similar to that observed before the cluster, arguing for a syllable boundary within the geminate. However, airflow patterns do not suggest that any "re-articulation" occurs during the geminate constriction [unlike in French pseudo-geminates, Smith (1996)]. Airflow shows the expected elevation at the transitions between vowel and fricative and a single low trough during the constriction, except for /h/. Because there is no supraglottal constriction, /h/ has peak airflow near the center of its duration. Thus /h/ does not pattern with other "gutturals."

2aSC9. Vowel density differences among regional dialects of American English. Robert A. Fox and Ewa Jacewicz (SPA Labs, Dept. of Speech and Hearing Sci., Ohio State, 1070 Carmack Rd., Columbus, OH 43210)

Over the past decade there has been considerable interest in differences in the size and shape of vowel spaces across different ages and genders of speakers and across different regional dialects of English. Vowel spaces provide information about the distribution of vowels in the F1 by F2 plane in an individual speaker or group of speakers. Recent studies have investigated which vowels should be used to establish the spatial boundaries and what criteria should be used in drawing the boundaries. However, the weakness of all such approaches is that they do not take into account the number of vowels produced in different portions of the vowel space (the occurrence of a single vowel in a peripheral location is sufficient to enlarge the vowel space). This study proposes to expand and refine the concept of vowel space by creating vowel density maps across different regional dialects of American English. These maps represent a three-dimensional overlay of relative vowel frequencies onto the F1 × F2 plane (the borders of which represent the vowel space). Dialectal differences will be described in terms of differences in the intersection of vowel space areas and differences in vowel density within different portions of the intersected spaces.

2aSC10. Coarticulation and consonant cluster production in Iwaidja. J. M. Fletcher (School of Lang. and Linguist., Univ. of Melbourne, Parkville, Victoria 3010, Australia, janetf@unimelb.edu.au), A. R. Butcher (Flinders Univ., South Australia 5001, Australia), D. Loakes (Univ. of Melbourne, Parkville, Victoria 3010, Australia), and H. Stoakes (Flinders Univ., South Australia 5001, Australia)

Articulatory analyzes of consonant cluster production have provided a rich source of information for modeling the production of speech over the years. It has been suggested that Australian indigenous languages show remarkable stability in heterorganic C1C2 sequences where C1 is coronal and C2 is noncoronal as in /nk/ sequences. In other words they show little or no anticipatory place of articulation assimilation. In this paper, the focus is on consonant cluster production in Iwaidja, an endangered Australian indigenous language spoken in the Northern Territory, Australia. Like most Australian languages, Iwaidja has a rich set of oral stop, lateral, and nasal place of articulation contrasts. Results of an acoustic phonetic and electropalatographic study show that assimilation is resisted in various ways in order to maintain place contrasts in certain sequences. Strategies include gestural delay and associated lengthening of initial consonants in C1C2 sequences. Nevertheless there is also clear evidence of a range of other effects including blending in situations of gestural conflict. Implications for models of coarticulatory processes are presented in the light of prevailing speech production models and compared to earlier findings based on analyzes of two other Australian languages: Warlpiri and Arrernte.

2aSC11. An acoustical study of American English diphthongs. Byunggon Yang (English Education Dept., Pusan Natl. Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

English diphthongs are usually produced with more than one vocal tract shape. This study attempts to collect acoustical data of English diphthongs published by Hillenbrand *et al.* (1995) online and to examine acoustic features of the diphthongs for phoneticians and English teachers. 63 American males and females were chosen after excluding those subjects with different target vowels or ambiguous formant tracks. The author used PRAAT to obtain the acoustical data systematically at 11 equidistant timepoints over the diphthongal segment. Results show that the formant trajectories of the diphthongs produced by the American males and females appeared quite similar. When the female formant values were uniformly normalized to those of the males, almost a perfect collapse occurred. Second, the diphthongal movements on the vowel space appeared not linear due to the coarticulatory gesture for the following consonant. Third, the average duration of the diphthongs produced by the females was 1.156 times longer than that of the males while the pitch ratio between the two groups turned out to be 1.746 with a similar contour over measurement points. The author concludes that English diphthongs produced by various groups can be compared systematically when the acoustical values are obtained at proportional timepoints.

2aSC12. Toronto English vowels: Deriving the vowel space from conversational data. Robert Hagiwara (Dept. Linguist., Univ of Manitoba, Winnipeg, R3T 5V5, MB Canada)

This presentation develops an overview of the Toronto English vowel space using data derived from the unscripted, casual speech typical of sociolinguistic interviews, rather than the more "formal" avenue of laboratory speech. Drawing on a corpus of sociolinguistic interviews collected by Walker & Hoffman ("language contact, linguistic variation and ethnic identity in Toronto English" SSHRC SRG 410-2008-2048) and using multiple-point formant measurements of stressed vowels in "clean" environments (e .g., avoiding coda nasals and liquids), this study develops views of the vowel space that illustrate modal or average vowel centers, within-category scatter, between-category dispersion, and VISC. In refining this kind of vowel data collection and analysis, this research provides a baseline for further studies, such as to describe vowel differences in "ethnically" marked varieties of English, to characterize adjustments made in specific phonological contexts, to identify previously undescribed phonetic variation in Canadian English, and to compare similar data from other varieties of English.

2aSC13. An acoustic description of Utah English vowels. Robert D. Sykes (Dept. of Linguist., Univ. of Washington, C104 Padelford Hall, Box 354360, Seattle, WA 98195, rsykes@u.washington.edu)

Labov [**30**, (1991)] described Western English as a dialect where neither the Northern Cities Shift nor the Southern Shift operate. The vowel/æ/ is resistant to raising and /a/ and /ɔ/ are merged as a single low-back phoneme. Labov *et al.* [**37**, (2006)] (ANAE) predicted differential fronting of /uw/ and /ow/, a lack of Canadian Raising or glide-deletion, and a merger or near-merger of /i/ and /ɪ/ before tautosyllabic /l/. Di Paolo and Faber [(1990)] argued that the vowel system of Utah fits the Southern pattern. This would predict a reversal of /i/ and /ɪ/ in phonetic space and the deletion or weakening of the glide in the diphthong /aɪ/. This study tests this prediction in the speech of natives of Salt Lake City, UT. Sociolinguistic interviews of seven young adults were analyzed acoustically. Results show that the Utah vowel system is more complicated than either ANAE or Di Paolo and Faber suggested. The data suggest differential fronting of /uw/ and /ow/, an approximation of /i/ and /ɪ/, and an approximation of /a/ and /ɔ/, consistent with Western patterns. However, data also suggest the glide of /aɪ/ is weakened before voiced segments and in open syllables, consistent with the Southern pattern.

2aSC14. Vowel reduction and merger in Pacific Northwest English. Alicia B. Wassink (Dept. of Linguist., Univ. of Washington, Box 354340, A210 Padelford Hall, Seattle, WA 98195-4340, wassink@uw.edu)

The assumption stands in acoustic phonetics that vowel reduction (including undershoot and centralization of short vowels) is greater in casual than controlled speech. However, few studies have investigated the relative magnitude of language-general (phonetic) effects relative to dialect-specific (sociolinguistic) ones. This paper investigates reduction in vowels undergoing sound-change. Pacific Northwestern English (PNWE) /æ/, /ɛ/ are rising to the spectral location of /ey/, a "merger by approximation" [W. Labov, Princ. Ling. Change, Blackwell (1994)]. Sociolinguistic literature guided selection of two stable/changing pairs (in which one member is stable, the other involved in the merger): /iy ey/, /ɪɛ/. The normalized corpus (n=1500) was balanced for the following place and voicing and included three conditions (wordlist, reading, and casual). Speakers were partitioned into groups (merged/unmerged). Wassink's [J. Acoust. Soc. Am. **119**, 2334–2350 (2006)] spectral overlap assessment metric was used to detect differences between volumes of vowel ellipsoids (in F1×F2×duration space) and test for style-related spectral shift. Euclidean distances (measured between 20% and 80% timepoints) differentiated vowel trajectories recorded in different conditions. Merged and unmerged speakers' reduction strategies differed. Merged: expected reduction patterns in both short vowels (including raised /ɛ/). /y/ trajectories show greater gliding in casual speech. Unmerged: /ɛ/ does not shift position or trajectory as task changes. [Research supported by National Science Foundation BCS#-0643374.]

2aSC15. Ultrasound analysis of the Czech phoneme /ř/. Phil Howson, Ekaterina Komova (Dept. of Ling., Univ. of BC, E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada, mystblades454@hotmail.com), and Bryan Gick (Univ. of BC, Vancouver, BC V6T 1Z1, Canada)

Previous studies comparing the articulation of the contrasting Czech trills /ř/ and /r/ have described /ř/ as being produced with the laminal portion of the tongue at a more anterior location with a narrower channel [Ladefoged and Maddieson, *The Sounds of the World's Languages* Blackwell, Oxford (1996)] and using a tighter constriction than /r/ [Dankovičová, JIPA **27**, 77–80 (1999)]. These observations correspond to the IPA symbol [r] with a raising diacritic, which replaced the former symbol corresponding to a voiced strident apico-alveolar trill. The current project uses *B*-mode and *M*-mode ultrasound imaging and acoustics to test these descriptions of /ř/. Preliminary results based on a single participant revealed nearly identical constriction locations for the two trills, with the /ř/ constriction vibrating a larger tongue surface area. Overall, the tongue surface is lower and more forward in the mouth for /ř/ than for /r/, making [r] with a lowering diacritic a more appropriate choice for representing this sound. Spectrographic evidence also reveals important temporal properties for /ř/. Results for additional participants will be reported. [Work supported by the NSERC.]

2aSC16. The palatoglossal sphincter in French uvular /r/. Naomi C. Francis, Anna Klenin, Ezra Mizrahi, Denise C. Tom, and Bryan Gick (Dept. of Linguist., Buchanan Bldg., E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada)

Speech production models have typically defined lingual constrictions in terms of tongue motion relative to unmoving opposing surfaces, such as the palate or teeth. While such a model may apply well to hard structures such as the palate, it is not clear that this appropriately characterizes constrictions involving soft structures such as the velum and uvula. Anatomically, the uvula and velum constitute the upper surface of a muscular sphincter including the palatoglossus muscles laterally and the tongue below. The present study tests whether uvular constrictions may be better described as actions of this palatoglossal sphincter, analogous to the lips, velopharyngeal port, or epilaryngeal tube. We tested this proposal by measuring the response of the velum/uvula during production of French uvular /r/ in an x-ray film [Munhall *et al.*, J. Acoust. Soc. Am., **98**(2), 1222–1224 (1995)]. Preliminary findings indicate that, as seen with other sphincters, speakers exhibit a variety of different mechanisms for forming constrictions, but all show active involvement of the palatoglossal arch, in addition to the tongue, in producing the uvular constriction. Modifications to speech production models to accommodate these findings will be discussed. [Work supported by the NSERC.]

2aSC17. Suprasegmental features of Chinese-accented English. Bin Li and Lan Shuai (Dept. of Chinese, Translation Linguist., City Univ. of Hong Kong)

This study compares read speech by Cantonese and Mandarin learners of English with those by native speakers of English. The utterances produced by learners of English are all rated as accented. We aim to identify measurable prosodic differences leading to such perceptual judgment. The prosodic correlates we examine include both syllable duration and pitch values. Our preliminary results reveal in generally mean syllable durations of Cantonese and Mandarins learners are very similar, and much longer than that of the native speakers of English. It is also found that all learners' production exhibits much greater variations in duration than native speakers. Examination on duration of stressed and unstressed syllables reveals more differences across L1 groups. With regard to pitch correlates, a much wider pitch range is found in the production of native speakers of English than in those of both Mandarin and Cantonese learners of English. Of the Chinese learners, Mandarin group employ a wider pitch range than Cantonese. Moreover, Cantonese learners' speech is found dominated by a rather "flat" overall pitch contour. We propose that the prosodic deviations in Chinese learners' English from a native speaker can be accounted for from the perspective of rhythmic and tone system in Chinese.

2aSC18. Acoustic correlates of perceptual ratings of foreign-accented English. Elizabeth McCullough (Dept. of Linguist., The Ohio State Univ., 225 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, eam@ling.ohio-state.edu)

What acoustic cues contribute to the perception of foreign accent? While many previous studies have explored this question for a single foreign accent, few have simultaneously considered several foreign accents in a single language. In this perception study, ten American English listeners rated 360 stop-vowel sequences extracted from English words produced by L1 American English, L1 Hindi, L1 Korean, and L1 Mandarin talkers on a continuous scale of degree of foreign accent. Despite this minimal input, listeners robustly rated Korean- and Mandarin-accented productions as having a higher degree of foreign accent than native productions, and Hindi-accented productions as having a higher degree of foreign accent than all other productions. Acoustic cues of the stimuli were measured, and stepwise linear regression models revealed that vowel quality and VOT contributed significantly to the ratings. These cues collectively accounted for 26% of the variance in the ratings of all stimuli and for 38% of the variance in the ratings of stimuli beginning with voiceless stops. The results suggest that vowel quality and VOT contribute substantially to the perception of foreign accent in stop-vowel sequences, and perhaps in larger units of speech.

2aSC19. Time to pull out the stops: Spirantization in Pacific Northwestern English. John M. Riebold (5131 26th Ave. NE, Seattle, WA 98105, riebold@uw.edu)

Anecdotal evidence suggests that English speakers in North America are spirantizing stops. Spirantization is a consonant lenition process active in many languages, but has traditionally been assumed not to be active in English, except as a lexical or affixal process. This study is an investigation into the spirantization of stops using recordings of eight age/sex matched dyads ($n=16$) from a small corpus of Pacific Northwestern English speech. Representative monologues were extracted for each speaker and analyzed acoustically. Alveolars and voiceless aspirated stops were excluded from analysis as alveolar stops have a large number of allophones, and aspiration may block spirantization. The resulting tokens were rated for the absence of a stop burst, the presence of frication noise, and the presence of formant structure. Scores for each factor were then used to create a coarse-grained spirantization score for each token, with lack of a release burst, and presence of frication or formants indicating incomplete closure. Preliminary results from four speakers (88 tokens total) show that spirantization is indeed occurring: 44% of the tokens were spirantized. Additionally, it appears that velars are much more likely to spirantize than bilabials: 77% of velars spirantized compared to 26% of bilabials.

2aSC20. Does merger happen in the three-way contrast of Korean stops? Jianjing Kuang (Dept. of Linguist., Phonet. Lab., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095-1543, kuangjj@gmail.com) and Mira Oh (Chonnam Natl. Univ., Korea)

Most previous studies of Korean stops have focused on limited acoustic cues, e.g., VOT and F0; few have considered voice quality and if any, only H1-H2 was mostly studied. Therefore, the question of the realization of three-way stop contrast (lenis, aspirated, and tense) in Korean is not settled. Given that the contrast is undergoing sound change (VOT of lenis is increasing), there arises the question whether other acoustic cues also change over time to keep the three-way contrast. Therefore, this study comprehensively investigates various acoustic measures of Korean stops from ten young Seoul speakers (five females; five males). Other than the distinctive low F0 after lenis stops and non-distinctive VOT between lenis and aspirated stops, our preliminary results show the the following. First, females and males exhibit significantly different acoustic realizations of three types of stops. Second, compared to the other two types, lenis stops are special in the way that they have larger variations of values in various acoustic measures (VOT, F0, $H1^*$-$H2^*$, $H1^*$-$A1^*$, $H1^*$-$A3^*$, etc.), which may account for the conflicting results in previous studies. Third, in the young generation, although VOT of lenis stops is merging toward aspirated stops, the vowel phonation after lenis stops is moving toward that after tense stops. This study suggests that the three-way stop contrast is maintained but with different acoustic realizations.

2aSC21. The theory of adaptive dispersion and acoustic-phonetic properties of cross-language lexical-tone systems. Jennifer A. Alexander (Dept. of Linguist., Simon Fraser Univ., 8888 Univ. Dr., Burnaby, BC V5A 1S6, Canada)

Lexical-tone languages use fundamental frequency (F0) to convey word-meaning. Nearly half of all languages use lexical tone [Maddieson (2008)], yet those systems are under-studied. To increase our understanding of speech-sound inventory organization, I extend to tone-systems a model of vowel-system organization, the theory of adaptive dispersion (TAD) [Liljencrants and Lindblom (1972)]. This is a cross-language investigation of whether and how tone-inventory size affects acoustic tone-space size. Five languages with different-sized tone-inventories were compared: Cantonese (six tones), Thai (five tones), Mandarin (four tones), Yoruba (three tones), and Igbo (two tones). Six native speakers (three female) of each language produced 18 CV syllables in isolation, with each of his/her language's tones, six times. Tonal F0 (semitones) was measured at three equidistant points across the vowel. Each language's tone-space was defined in two ways: (1) the F0 difference between its highest and lowest tones and (2) the configuration of tones in a 2-D space defined by onset F0 x offglide F0. Following the TAD, I predicted that languages with larger tone-inventories would have larger tone-spaces; this was not supported by (1). However, the dispersion of tones in (2) supports the TAD hypothesis that sound-categories will be well-dispersed across the space and highly contrastive.

2aSC22. The vowel tango: Rethinking vowel-inherent spectral change. Catherine L. Rogers, Merete M. Glasbrenner, Teresa M. DeMasi, and Michelle Bianchi (Univ. of South Florida, Comm. Sci. & Dis., 4202 E. Fowler Ave., Tampa, FL 33620)

Vowel-inherent spectral change (VISC) refers to vowel-intrinsic formant movement across a vowel steady state. VISC has been shown to (1) be consistent across talkers within a given dialect, (2) vary regularly across vowels within a dialect, (3) vary regularly across dialects, and (4) be necessary for peak vowel-identification accuracy. Hence, VISC has become accepted as a phonetic feature of monophthong vowels of North American English. VISC is typically portrayed using averages across tokens and talkers, highlighting regularity but potentially masking individual differences. To understand vowel production by second-language learners, we were particularly interested in such individual variation. In analyzing individual differences for neighboring target vowels, we found no single time point at which all sets of target vowel tokens were well distinguished from one another. However, looking across three time points, all native-speaker vowel sets were well distinguished from each possible neighbor set at some time point. Thus, VISC can be seen as the steps in a sort of dance, as each vowel moves to avoid overlapping with another, ultimately causing overlap with another and then more movement. This perspective is compatible with models of efficient coding and stochastic and/or exemplar based models of speech production and perception. [NIH-NIDCD #1R03DC005561-01A1.]

2aSC23. Effects of talker experience on perceived clarity and acoustic features of clear versus conversational speech. Sarah Hargus Ferguson, Taylor Denise Widener, and Tanner Mackey (Dept. of Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

Studies of the 41-talker Ferguson Clear Speech Database [Ferguson (2004)] have found no relationship between talkers' self-reported experience communicating with individuals with hearing loss and the magnitude of the clear speech effect for vowels presented in noise. That is, talkers with no experience talking to people with hearing loss showed just as much clear speech vowel intelligibility benefit as talkers with extensive experience. The present study explored whether talker experience effects might emerge with more meaningful materials. Clear and conversational sentences produced by eight talkers (four with extensive experience and four with no experience) were presented in quiet to young listeners with normal hearing, who rated the clarity of each sentence on a seven-point scale. Several acoustic measures were also performed on these sentences. While the average size of the clarity difference between clear and conversational speech was similar for the two groups, talker experience effects were found for some of the acoustic measures. However, considerable variability was observed among the talkers, particularly within the "no experience" group. Relationships between acoustic measures, clarity ratings, and talker experience will be discussed. The present data will also be compared to those from an earlier study using a different set of eight talkers.

2aSC24. Acoustic landmark analysis of speaker differences in producing clear versus conversational speech. Sarah Hamilton, Sandra Combs, Shruti Balvalli (Dept. of Comm. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH 45267-0379, hamilsm@mail.uc.edu), Joel MacAuslan (S.T.A.R. Corp., Bedford, MA 01730), Jean Krause (Univ. of South Florida, Tampa, FL 33620-8100), and Suzanne E. Boyce (Univ. of Cincinnati, Cincinnati, OH 45267-0379)

In previous work, we have shown that landmark analysis can be used to detect acoustic changes in clear versus conversational (typical and plain) speech that are associated with changes in intelligibility to listeners. However, these results derive from studies using a small number of speakers [Bradlow and Bent, (2002); Smiljanic and Bradlow (2008)]. The degree to which landmark analysis can be used to predict intelligibility for any particular speaker or sample of speech is not known. In this paper, we present results from a study of listener intelligibility for clear and conversational speech by four speakers with widely differing scores on parameters of acoustic change as determined by landmark analysis. The data consist of intelligibility scores (# content words correct) for a set of BKB sentences produced by these four speakers and presented to four groups of listeners in quiet and in noise. We compare these data with data from studies by Krause & Braida [2002, 2004]. Results will be discussed in terms of the distribution of acoustic changes used by individual speakers to produce clear versus conversational speech, and the relationship of these changes to listener intelligibility.

2aSC25. Motor-induced suppression of auditory neural responses to pitch-shifted voice feedback. Roozbeh Behroozmand and Charles Larson (Speech Physio. Lab., Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL 60201)

Surviving in an alien world is dependent upon the ability to distinguish between sensory inputs arising from self-actions and those of the others. In the auditory system, the motor-driven predictions about expected sensory feedback (efference copies) are proposed to suppress sensory neural responses to self-voice feedback that is predicted by the efference copies. In the present study, event-related potentials were recorded in response to five different magnitudes (0, 50, 100, 200, and 400 cents) of pitch shift stimuli at voice onset during active vocal production and passive listening to the playback. Results indicated that the suppression of N1 component during vocal production was largest for normal voice feedback (PSS: 0 cents) and became smaller as the magnitude of pitch shift stimuli increased. For the largest tested PSS magnitude (400 cents), the N1 suppression was completely eliminated. One possible explanation for this effect is that the brain utilizes the motor predictions (efference copies) to suppress the auditory feedback of self-generated vocalizations. The reduction of suppression for 50, 100, and 200 cents pitch shifts in voice feedback and its elimination for 400 cents stimuli provide supporting evidence for the idea that motor-driven predictions provide mechanisms that lead to distinctly different sensory neural processing of self versus non-self vocalizations.

2aSC26. Re-examining the effects of perturbation magnitude, direction, and duration on the pitch-shift reflex. Patricia J. Allen and Theresa A. Burnett (Dept. of Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, pja@indiana.edu)

Comparing average voice fundamental frequency ($F0$) responses to pitch-shifted auditory feedback has revealed effects of some stimulus parameters, but not of others. We hypothesize that some stimulus effects on pitch-shift reflex characteristics are spurious, the result of differences in the incidence of the pitch-shift reflex rather than its inherent characteristics. Using a template correlation analysis technique that allows the pitch-shift reflex to be identified in individual trials, the effects of stimulus magnitude, direction, and duration were re-examined. We presented 30 healthy adults with auditory feedback pitch shifts during sustained phonation of the vowel /u/. Pitch shifts were all combinations of magnitude (50, 300, and 600 cents), Direction (up and down) and duration (100 and 500 ms) for a total of 12 conditions. Results indicate significant effects of pitch perturbation magnitude and direction on the incidence of the pitch-shift reflex. Implications for our understanding of voice *F*0 responses to changes in auditory feedback are discussed.

2aSC27. Statistical relationships in distinctive feature models and acoustic-phonetic properties of English consonants. Kenneth de Jong (Dept of Linguist., Indiana Univ., 322 Memorial Hall, Bloomington, IN 47405-7005, kdejong@indiana.edu), Noah H. Silbert (Univ. of Maryland), Kirsten T. Regier, and Aaron Albin (Indiana Univ.)

The relationship between segmental contrasts, often modeled as being composed of distinctive features, and actual acoustic-phonetic properties is complex and many-to-many. Contrasts may be cued by a multiple acoustic-phonetic properties, and acoustic-phonetic properties often cue multiple contrasts. This paper presents a hierarchical multivariate statistical model of the relationship between a suite of 11 acoustic measurements and various target feature systems. Measurements are taken from 10 repetitions of 16 English consonants by 20 native speakers of English in both onset and coda position in nonsense monosyllables. Target feature systems range from models with little generalization across the various segments to ones that fully cross distinctive features to specify all of the segments. The statistical model enables analysis of within-speaker and between-speaker sources of variability in consonant production, and constitutes a principled statistical method for comparing these different distinctive feature models. Also, the role of the statistical model as a baseline for work on relations between different segments in perceptual categorization will be outlined.

2aSC28. Three dimensional digital articulator labeled atlas. Emi Z. Murano (Dept. of Otolaryngol., Head and Neck Surgery, 550 N. Broadway, Baltimore, MD 21205, emurano1@jhmi.edu), Yuanming Suo, Fangxu Xing, Snehashis Roy, John Bogovic (Johns Hopkins Univ., Baltimore, MD 21218), Maureen Stone (Univ. of Maryland, Baltimore, MD 21201), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD 21218)

There is an increased use of both dynamic and structural magnetic resonance imaging (MRI) methods on studies in normal and disordered speeches. However, due to the anatomical variation among subjects, it has been a challenge to obtain quantitative results from a series of subjects or patients. To compare inter- and intrasubjects' MRI data a 3-D digital articulator labeled atlas is presented. Experts, head and neck surgeons and speech language pathologists, manually segmented the tongue, soft and hard palate, lips, pharyngeal walls, and larynx from three sets of MRI of healthy volunteers. Then, 30 naïve raters segmented the same articulators with a multiple-structure semiautomated approach for rapid delineation. Concept, application, and comparison with multiple raters and fully manual delineation are provided. [Work supported by NIH-NIDCD Grant No. R00-DC009279.]

2aSC29. Subphonemic planning across syllable and word boundaries. Donald Derrick and Bryan Gick (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC, V6T 1Z4, Canada, dderrick@interchange.ubc.ca)

Previous research demonstrated that there are no fixed motor programs or tasks in speech, and there is evidence for subphonemic planning in speech within a word across up to two phoneme boundaries [Derrick and Gick (Submitted)]. Because this evidence is word-internal, it could be suggested that speakers simply memorize many motor programs for each word and draw on them as needed. We demonstrate that speech planning extends across three phonemes, two syllables and a morpheme boundary. Participants produce more up-flaps the first flap of words such as "editor" and "auditor", which often end with the tongue tip up, versus alveolar taps in "edify" and "audify," which end with the tongue tip down. We also demonstrate planning across word boundaries. Participants also produce more up-flaps in "edit" and "audit" if these words are followed by "a," which has a double flap sequence, than if the words are followed by "the," which does not. Planning across morpheme and word boundaries would ultimately require memorization of an infinite number of motor programs or tasks.

2aSC30. Feed-forward control of phonetic gestures in consonant–vowel syllables: Evidence from responses to auditory startle. Chenhao Chiu (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, chenhao@interchange.ubc.ca), Andrew Stevenson, Dana Maslovat, Romeo Chua (Univ. of British Columbia, 210-6081 University Blvd., Vancouver, BC V6T 1Z1, Canada), Bryan Gick (Univ. of British Columbia, 2613 West Mall, Vancouver, BC, V6T 1Z4,Canada), and Ian Franks (Univ. of British Columbia, 210-6081 University Blvd., Vancouver, BC V6T 1Z1, Canada)

Speech production like other limb movements relies on both feed-forward and feedback mechanisms. Use of a startling auditory stimulus (>90 dB) has been shown to trigger fast, accurate feed-forward performances in upper limb movements prior to access to feedback information [Valls-Solé *et al.* (1999), J. Physiol. **516**: 931–938; Carlsen *et al.* (2004), J. Mot. Behav. **36**: 253–264]. This startle paradigm is applied to test whether pre-programed, feed-forward speech production differs in phonetic detail from production with access to feedback. The experiment examined the production of the CV syllable [ba], starting with the mouth either open or closed. This speech production was triggered either by a control stimulus (82 dB) or by a startling stimulus (124 dB). Results from ten participants showed that lip compression occurred for both starting conditions (mouth open and mouth closed), and also indicated that the timing relationships of the articulators were stable across control trials and startle trials. The acoustics of syllables, formant values, and maximum amplitudes, were consistent across control trials and startle trials, suggesting that the production of pre-programmed gestural configurations in CV syllables may be executed under exclusively feed-forward control.

2aSC31. Word duration and segment deletion as measures of reduction in a corpus of spontaneous speech. Philip Dilts, R. Harald Baayen, and Benjamin V. Tucker (Dept. of Linguist., Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, pdilts@ualberta.ca)

The present study explores phonetic reduction in the Buckeye Corpus [Pitt *et al.*, Speech. Commun. **45**, 89–95 (2005)] following up on the work of Johnson [Proceedings of the 1st Session 10th International Symposiam (2004), pp.], who counted the number of segments and syllables deleted from each word in the subset of the corpus that was available at the time. The first experiment presented here provides updated rates of segment deletion (over 25% of words) and syllable deletion (over 6% of words) as measures of reduction rates in the entire completed corpus. The second experiment investigates reduction in the duration of words as a function of several linguistic factors, including frequency, conditional probability, and rate of speech. The data are modeled using multiple mixed-effect linear regression to predict both the duration of each word and the reduction from the median duration of each word. Age, gender, and interviewer gender were not found to affect word duration. Citation length and current rate of speech were both strong predictors of word duration. Frequency and predictability from context also correlate strongly with word duration. Verbs were found to be significantly shorter in duration than other content words with the same citation length.

2aSC32. Are apical trills associated with raised fundamental frequency? Julia E. Trippe, Susan Guion-Anderson (Dept. of Linguist., Univ. of Oregon, Eugene, OR 97403-1290, trippe@uoregon.edu), and Ratree Wayland (Univ. of Florida, Gainesville, FL 32611-5454)

Previous work has suggested that apical trill production may condition a raised F0, affecting the vowel after onset consonants. This effect has been suggested as the impetus for the reanalysis of a falling-rising tone replacing an onset trill [r] in Khmer [Wayland and Guion, Mon-Khmer Studies **35**, 55–82 (2005)]. However, it has not been established whether this increase in F0 is a natural correlate of apical trill production. If so, it should be found in other languages. A raised F0 has been found for trill production in Thai, but preliminary data examining Finnish and Spanish onset trills found no such pattern. Some speakers produced higher F0 after [r] and others did not. The questions of whether there may be aerodynamic constraints associated with trill production that affect F0 or whether language-specific phonetic knowledge, perhaps unconditioned by articulatory constraints, is the source of the F0 patterns are discussed. As part of this discussion, the variation in trills, from multiple cycles to approximate productions, is considered. From a usage-based phonology and sound change perspective, it is questioned how [r] consistently increases F0 to the point of conditioning tonogenesis in one language, but seemingly has no effect on F0 in other languages.

2aSC33. Compression of the acoustic space in intoxicated speech. Abby Kaplan (Linguist. Dept., Univ. of Utah, 255 S. Central Campus Dr., Rm. 2300, Salt Lake City, UT 84112, abby.kaplan@utah.edu)

Researchers at the phonetics-phonology interface frequently invoke "articulatory effort" to explain the behavior of various speech sounds: less effortful sounds are preferred. However, there is scant empirical basis for most claims about which sounds are the effortful ones. This experiment attempts to observe effort reduction in the laboratory by comparing the speech of sober and intoxicated subjects; there is reason to believe that subjects produce more "easy" sounds when intoxicated than when sober. Subjects produced 264 target words containing intervocalic stops, nasal-stop sequences, and stressed vowels. Acoustic analysis of these productions reveals that subjects' acoustic space was somewhat smaller in the intoxicated condition than in the sober condition for several measures including stop voicing and intensity; however, no such contraction occurred for vowels. These results have two implications. First, compression of the acoustic space suggests that speech in the intoxicated condition may indeed have been less effortful. Second, the results call into question traditional hypotheses about effort reduction: intervocalic stops moved toward the middle of the VOT continuum, rather than becoming more voiced; vowels, contrary to expectation, did not become more centralized. This study highlights the need for further experimental work on the role of articulatory effort in sound patterns.

2aSC34. Diphthong formant transitions in four speaking tasks. Christina Kuo and Gary Weismer (Dept. of Communicative Disord. and Waisman Ctr., Univ. of Wisconsin-Madison, 1975 Willow Dr., Madison, WI 53706, kuo2@wisc.edu)

Diphthongs have been demonstrated to have longer durations and to involve more extensive articulatory gestures, reflected as larger frequency changes, when compared to monophthongs [Lehiste, I., and Peterson, G. E., J. Acoust. Soc. Am. **33**, 268–277 (1961); Holbrook, A. and Fairbanks, G., J. Speech Hear. Res. **5**, 38–58 (1962)]. However, limited work has documented the nature of diphthong variability. This study describes and examines diphthong formant transitions in the speech production of ten healthy male native speakers of American English, with Wisconsin dialects, to identify and evaluate the pattern, if any, of diphthong variability in four speaking tasks: conversational speech, story reading, sentence in citation form, sentence in clear speech. The study is motivated by the phenomenon of vowel reduction [Lindblom, J. Acoust. Soc. Am. **35**, 1773–1781 (1963)], which has been documented widely for monophthongs, as well as speaking rate-induced changes in diphthong formant transitions [Gay, T., J. Acoust. Soc. Am. **44**, 1570–1573 (1968)]. Five diphthongs /eɪ/, /ɑɪ/, /ɑʊ/, /oʊ/, and /ɔɪ/ in American English are included, and formant transition measures, including transition duration, transition extent, and derived slope (i.e., transition extent /duration) are used for analysis. Findings are discussed within the framework of the acoustic theories of speech production.

2aSC35. Multipulse articulatory modeling in the Wisconsin x-ray microbeam speech production database. Athanasios Katsamanis, Erik Bresch, Louis Goldstein, and Shrikanth Narayanan (Univ. of Southern California, Electrical Eng., 3740 McClintock, Rm. 400, Los Angeles, CA 90089)

Multipulse modeling of articulatory movements can provide a flexible and intuitive representation of the dynamic behavior of speech articulators. Such a representation can prove especially useful for the exploitation of the significant amounts of rich articulatory data that are collected via vocal tract imaging techniques such as real-time MRI or x-ray microbeam. Original multipulse LPC articulatory modeling studies only focused on a limited set of articulations by a single speaker as these were imaged using x-rays. In this work, application of the multipulse modeling framework on the Wisconsin x-ray microbeam speech production database is investigated. This database contains articulatory data, i.e., trajectories of points on the tongue, jaw, and lips during multiple articulations as well as hard palate tracings, from 57 speakers. The original measurements are converted to constriction degree measurements and it is assumed that the corresponding trajectories can be modeled as the output of a low-order linear system when it is excited by a sequence of pulses of variable amplitudes. Pulse locations are determined using optimal matching pursuit and all pulse amplitudes are reoptimized every time a new pulse is introduced. Model similarities and differences are presented both across multiple utterances by the same speaker and across speakers.

2aSC36. Decomposition of vowel and consonant contributions to the time-varying vocal tract shape. Brad H. Story and Kate Bunton (Dept. Speech, Lang., and Hear. Sci., Univ. of Arizona, Tucson, AZ 85721)

An important aspect of speech production is the temporal overlap of the different vocal tract movements that produce vowels and consonants. In previous work a method was proposed for separating the time-varying vocal tract shape of a talker into vowel transitions and consonant superposition functions [Story, J. Acoust. Soc. Am. **126** (2009)]. The purpose of this study was to apply the method to a wider variety of consonant and vowel contexts than were previously reported. The specific aims were to analyze x-ray microbeam data produced by one male speaker for three stop consonants [p,t,k] embedded in six vowel contexts [ia, ai, oae, aeo, ii, aa] and reconstruct them as separate consonant and vowel components. When superimposed, these were shown to be reasonable approximations of the original VCVs, as assessed qualitatively by visual inspection and quantitatively by calculating rms error and correlation coefficients. The information extracted through these analyzes can be used as input to a speech production model to simulate the original utterances. [Research supported by NIH R01-DC04789.]

2aSC37. The labial viseme reconsidered: Evidence from production and perception. Connor Mayer, Jennifer Abel, Adriano Barbosa, Alexis Black, and Eric Vatikiotis-Bateson (Commun. Dynam. Lab, Univ. of British Columbia, 6368 Stores Rd., Vancouver, BC V6T 1Z4, Canada, connorm @interchange.ubc.ca)

Previous studies have demonstrated that the labial stops /p,b,m/ may be impossible to discriminate visually, leading to their designation as a single viseme. This perceptual limitation has engendered the belief that there are no visible differences in the production of /p,b,m/, with consequences for research in machine recognition, where production differences below the level of the viseme have been largely ignored. Kinematic studies using high-speed cine, however, have previously documented systematic differences in the production of labial consonants. This study examines the degree to which visual /p,b,m/ are discriminable in production and perception. Two experiments—one designed to measure kinematic orofacial movement using optical flow analysis and one designed to test perceiver discrimination of /p,b,m/—were used to establish the absence/presence of systematic visual differences in bilabial productions, and to replicate the previous perception findings. Results from the optical flow analysis indicate systematic kinematic differences for the peak velocity of the lower face, which corresponds to the release of the bilabial consonant. Significant differences were found in velocity between /m/ and /b/, and between /p/ and /b/, but not between /p/ and /m/. Preliminary results from the perception task suggest discrimination of bilabial consonants is at chance level.

Session 2aUWa

Underwater Acoustics and Acoustical Oceanography: Propagation, Modeling, and Inversion I

Tarun K. Chandrayadula, Chair
George Mason Univ., Electrical and Computer Engineering Dept., 4400 University Ave., Fairfax, VA 22030

Contributed Papers

8:00

2aUWa1. Implementing wave propagation on desktop graphics processing units: Split-step Fourier parabolic equation modeling example. Paul Hursky and Michael B. Porter (HLS Res. Inc., 3366 North Torrey Pines Court, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

The quest for raw computing power has shifted from increasing processor clock speeds to increasing the number of processing cores. Currently, mainstream CPUs can be purchased in dual-slot quad-core and hex-core configurations. On the other hand, graphic cards provide hundreds of processing cores. Although there have been various implementations of scientific applications on graphics hardware, including underwater acoustic modeling, widespread use of this technology has been hampered by the often extraordinary effort needed to program this hardware, especially if the application architecture did not match the canonical graphics pipeline for gaming. In the last few years, the major graphics board manufacturers have stepped away from designing hardware specialized for particular new graphic special effects and made a concerted effort to provide general-purpose computing capabilities, of the sort that can be exploited for scientific computing. For example, Nvidia's CUDA environment currently provides many building blocks for scientific computing, such as (subsets of) BLAS, LAPACK, and FFTs. We will present our experiences implementing the split-step Fourier parabolic equation in the CUDA environment, showing how we have achieved a 10 times speedup relative to a multi-core CPU implementation, with a modest investment in programming effort. We will also provide some guidelines on how various applications of interest to the underwater acoustics community might benefit from use of this technology.

8:15

2aUWa2. Finite element modeling of longitudinally invariant shallow water waveguides. Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Finite element models of shallow water waveguides are fully customizable to include effects such as range dependent bathymetry, sound speed profiles, and interface roughness. However, until recently, these solutions have been confined to 2-D environments. While these models provide benchmarks for other solution methods such as coupled modes or parabolic equations, they cannot be compared with data. Fully 3-D models of shallow water waveguides are still beyond current computational capability, but longitudinally invariant models may bridge the gap. These solutions are often known as 2.5D models. In this scheme, the waveguide is reduced to a series of 2-D finite element domains, the solutions to which comprise the Green's function in the wavenumber-frequency domain of the 2.5D space. A cosine transform is then used to convert these into a frequency-domain solution for a longitudinally invariant space. In this study, the wavenumber-frequency Green's function will be analyzed in order to develop more efficient computational methods for shallow water waveguides. Solutions for both flat and rough interface shallow water waveguides in 2.5D will be presented and compared with a wavenumber integration approach. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

8:30

2aUWa3. Comparison of two and three spatial dimensional solutions of a parabolic approximation of the wave equation at ocean-basin scales in the presence of internal waves: 200–250 Hz. John Spiesberger (Dept. of Earth and Environ. Sci., Univ. of Pennsylvania, 240 S 33rd St., Philadelphia, PA 19104-6316, johnsr@sas.upenn.edu)

Numerical solutions are given for a parabolic approximation to the acoustic wave equation at 200 and 250 Hz in two and three spatial dimensions to determine if azimuthal coupling in the horizontal coordinate significantly affects horizontal correlation in the presence of internal gravity waves in the sea. Coupling is a small effect at distances of 4000 km or less. This implies that accurate solutions are possible using computations from uncoupled vertical slices. Shapes of horizontal correlation are not far from shapes given by several theories. Estimates of horizontal correlation at 4000 km and 200 and 250 Hz are about 0.4 and 0.3 km, respectively. [Work supported by the Office of Naval Research Contract Nos. N00014-06-C-0031 and N00014-10-C-0480 and by a grant of computer time from the DOD High Performance Computing Modernization Program at the Naval Oceanographic Office.]

8:45

2aUWa4. Three dimensional parabolic equation modeling of acoustic intensity fluctuations due to internal wave phenomena. Georges A. Dossot, James H. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882), Kevin B. Smith (Naval Postgrad. School, Monterey, CA 93943), Mohsen Badiey (Univ. of Delaware, Newark, DE 19716), and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

During the Shallow Water 2006 (SW06) experiment, a J-15 acoustic source deployed from the Research Vessel Sharp transmitted broadband (50–450 Hz) chirp signals 15 km away from a vertical line array. The array was intentionally positioned near the shelf-break front and in an area where internal waves are known to occur. During the same time an internal wave, labeled event 44, passed through the sound field such that the internal wave front was near parallel to the acoustic transmission path. Measured data show substantial intensity fluctuations that vary over time and space due to complex multimode and multipath (both two and three dimensional) interference patterns. This presentation compares three dimensional modeling results using the experimental geometry, acoustic signal parameters, and a simulated oceanographic environment based on environmental moorings and ship-born sensors to mimic the measured internal wave event. A modified version of the three dimensional Monterey–Miami parabolic equation (MMPE) code, which incorporates a user-defined sound speed field, is used. Measured and modeled intensity fluctuations are compared during dominating horizontal regimes such as refraction, focusing, and defocusing. Modal-dependent time-arrival analysis during the different horizontal regimes is examined. A sensitivity study of different sea-bottom properties is explored. [Work sponsored by the Office of Naval Research.]

9:00

2aUWa5. Nonlinear internal wave parameter influences on energy propagation using radiative transport theory. Kara G. McMahon (Dept. Math. Sci., Rensselaer Poly. Inst., 110 8th St., Troy, NY 12180, mcmahk3@rpi.edu), James F. Lynch, Ying-Tsong Lin (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Ning Xiang, and William L. Siegmann (Rensselaer Poly. Inst., Troy, NY 12180)

When two trains of nonlinear internal waves roughly align, their lead waves form effective boundaries of an acoustic duct in which energy is trapped. This type of propagation may be considered a scattering process resulting from the broken internal wave fronts between the lead waves. A traditional approach uses adiabatic normal modes and sound speed perturbations to calculate energy propagation along horizontal rays. An alternate is a radiative transport method in which acoustic vertical modes carry energy and a two-dimensional (2D) transport equation describes horizontal propagation. This model has parameters that are related to waveguide properties and are found from physical internal wave features. One parameter incorporates the significant curvature observed in many internal wave fronts. The two solution approaches are compared and contrasted. Data from the Shallow Water '06 Experiment are used to specify internal wave parameters and to compare with model calculations. [Work supported by the ONR.]

9:15

2aUWa6. On the interaction between acoustic waves and an approaching solitary wave front in waveguides. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716)

Existence of temperature or density solitary wave fronts in waveguides can change the modal behavior of acoustic wave propagation. When a sound signal travels in horizontal distances, certain source-receiver geometries with respect to the front of a soliton cause multipath intensity interference fringes that are related to the generation of rays in the horizontal plane. Intensity fringes registered at the receiver during these events are due to the interaction of various vertical modes and horizontal rays. The fringe patterns and time-frequency structure, before and after a solitary wave crosses an acoustic track, are shown. Analysis of these fringe structures is discussed using normal mode and PE models. [Work supported by the ONR].

9:30

2aUWa7. Azimuthal variability in the "Quantifying, Predicting, and Exploiting Uncertainty" experiment: Can we reliably model the measurements? James F. Lynch, Arthur E. Newhall, Ying-Tsong Lin, Timothy F. Duda (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Chris Emerson, and Philip Abbot (OASIS, Inc., Lexington, MA 02421)

Using an acoustic source mounted on an autonomous underwater vehicle, measurements were made of the azimuthal variability of transmission loss (TL) in the East China Sea just northeast of Taiwan. These measurements showed substantial variability in the peak level TL (5–10 dB) and somewhat less in the pulse integrated TL. Angular peak widths were distributed between 0 and 35 deg. The variability in the acoustic field is caused primarily by oceanographic variability (and its interaction with bathymetry) and is influenced by fine scale to large scale ocean processes. We examine here whether or not we can adequately model such variability at present, and if not, what improvements in ocean and acoustic models will be needed to do so.

9:45

2aUWa8. A measurement system for shear speed using interface wave dispersion. James H. Miller, Gopu R. Potty, and Jeannette M. Greene (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882, miller @egr.uri.edu)

Our recent work has highlighted the effect of shear on the dispersion of acoustic normal modes. Recently, Pierce *et al.* showed the effect of shear on compressional wave attenuation and its frequency dependence. Potty and Miller [(2010)] showed that sediment shear speed can significantly impact compressional modal arrival times near the Airy phase. We have observed that neglecting shear in compressional wave speed inversions results in a bias in our long range sediment tomography inversions. All of these factors emphasize the importance of estimating shear wave speeds in semiconsolidated shallow water sediments. One of the most promising approaches to estimate shear speed is to invert the shear speed profile using the dispersion of seismoacoustic interface waves (Scholte waves) that travel along the sediment-water boundary. The propagation speed and attenuation of the Scholte wave are closely related to shear-wave speed and attenuation over a depth of one to two wavelengths into the seabed. A shear measurement system being developed at the University of Rhode Island based on this concept will be presented. Test data collected on land will be shown and preliminary analysis techniques will be discussed. [Work supported by the Office of Naval Research.]

10:00

2aUWa9. Calculation of wave propagation with elastic boundary conditions. Cathy Ann Clark (Naval Undersea Warfare Ctr., Code 1513, 1176 Howell St., Newport, RI 02841, cathy.clark@navy.mil)

Calculations of compressional and shear wave transmissions and reflections through sediment layers are utilized to derive a single bottom reflection coefficient by formulating an infinite sum of matrices and expressing the result as a convergent series. The expression of bottom loss as a single coefficient eliminates the need to search for sediment mode eigenvalues, resulting in an improvement in efficiency. When implemented in a normal mode propagation model, the bottom loss prediction is shown to successfully reproduce resonance effects due to shear wave conversion in various types of sediments. Comparisons of boundary calculations to other published results are presented and propagation predictions are compared to a number of measured data sets.

10:15—10:45 Break

10:45

2aUWa10. Adaptive beamforming techniques for bottom-loss estimation using marine ambient noise. Lanfranco Muzi and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201)

Previous studies in underwater acoustics have shown that conventional beamforming of ocean noise with a vertical line array of hydrophones can be used to estimate the bottom loss (BL) as a function of frequency and grazing angle. The BL estimate is obtained by comparing the output power of beams steered toward the sea surface with that of beams reflected off the seabed. In practice, the finite aperture of arrays causes the beamwidths to be frequency dependent, and wider beams decrease the resolution of the BL estimate, blurring the interference patterns caused by seabed layering, and other distinct features such as the critical angle transition. This effect increases as frequency decreases and eventually sets the low-frequency limit on BL estimation for this technique. To minimize these effects, high-resolution adaptive beamforming techniques were applied to both measured and simulated ambient noise data. The results in this presentation show that some ABF techniques (e.g., the white-noise-gain-constraint beamformer) are better suited for this particular application, whereas others can sometimes have surprisingly poor performance. [Work sponsored by the Office of Naval Research.]

11:00

2aUWa11. Estimation of geophysical parameters from ambient noise correlation. James Traer and Peter Gerstoft (UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093)

The passive fathometer correlates the ambient noise recorded by elements of an array to infer the depth of seabed reflecting layers. This includes two distinct processes: the cross-correlation of two elements separated by a finite distance and the autocorrelation of a single element. A model is presented for the correlation of ambient noise in the presence of reflecting boundaries with one sensor, a pair of sensors, and an array of sensors. The results obtained from experimental data from pressure and particle-velocity (vector) sensors are presented. The presence and location of reflecting boundaries can be inferred when the amplitude of the incident noise field

varies smoothly across all directions of incidence. In the presence of discontinuities in the incident noise field, spurious peaks appear in the noise correlation which can be attenuated relative to the reflection peaks with use of adaptive array processing (minimum variance distortionless response). The reflection coefficient of the reflecting boundary affects the height of the reflection peaks in the noise correlation and in the ocean this can be used to infer seabed properties such as density and sound speed.

11:15

2aUWa12. Bayesian inversion of seabed scattering data. Gavin Steininger, Stan E. Dosso (School of Earth and Ocean Sci., Bob Wright Ctr. A405, Univ. of Victoria, P.O. Box 3065, STN CSC, Victoria, BC V8W 3V6, Canada), Charles W. Holland (Penn State Univ., PA 16802), and Jan Dettmer (Univ. of Victoria, Victoria, BC V8W 3V6, Canada)

Reverberation modeling and sonar performance predictions in shallow water require good estimates of seabed scattering and reflection as well as an understanding of scattering processes in a particular region. This talk considers the ability to resolve scattering parameters (e.g., scattering strength and roughness) and geoacoustic parameters (layer thicknesses, sound speed, density, and attenuation) using Bayesian inversion and a forward model based on first-order perturbation scattering theory and multilayer reflection coefficients. Results are considered in terms of marginal posterior probability distributions, which quantify the effective data information content to resolve scattering/geoacoustic parameters. Inversions are applied to synthetic data and to direct-path scattering measurements from shallow-water test beds. These measurements probe the seabed on an intermediate spatial scale (patch-size radius of ~500 m for both reflection and scattering), which reduces the effects of ocean variability (associated with sound speed profile, seabed, and biologics) and uncertainty relative to long-range reverberation measurements. [This work is supported by the ONR.]

11:30

2aUWa13. Bayesian ambient noise inversion for geoacoustic uncertainty estimation. Jorge E. Quijano, Stan Dosso, Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada, jorgeq@pdx.edu), Martin Siderius, and Lisa M. Zurk (Portland State Univ., Portland, OR 97201)

The noise produced by wind-driven breaking waves in shallow water provides a method for probing the seabed, and drifting vertical arrays have been deployed for remote sensing of geoacoustic parameters by estimating the frequency- and angle-dependent reflection coefficients. In addition, techniques such as spectral factorization allow obtaining the impulse response of the multilayered seabed environment. This impulse response carries information of the sediment acoustic properties that can be extracted and passed as prior information to a Bayesian framework for the estimation of geoacoustic parameters and its corresponding uncertainties, which ultimately determine the resolution of the method. The Bayesian formulation estimates a joint posterior probability density function, from which marginal density functions, moments, and covariances between geoacoustic parameters of interest can be quantified. In this work, Bayesian inversion based on Markov-chain Monte Carlo sampling is applied to simulated ambient noise data for the estimation of layer thicknesses, compressional sound speed, density, and sediment attenuation, and the resolution of the method is explored as a function of qualities of the data such as array design and wind speed. The approach is applied to experimental data collected near Sicily.

11:45

2aUWa14. An under-ice Arctic geophysical exploration sonar system concept to resolve international territorial claims. Juan I. Arvelo, Jr. (Appl. Phys. Lab., The Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099)

A consequence of the shrinking polar ice cap is the increased interest in commercial shipping. Therefore, several nations are surveying the continental slopes and Arctic basin to better define the political boundaries. Under Article 76 of the United Nations Convention on the Law of the Sea (UNCLOS), the boundary between territorial and international waters is defined where the sub-bottom basement extends l km from the seafloor. Therefore, icebreakers and air guns are currently implemented for deep sub-bottom profiling. However, there still remains older and thicker ice that leads to inaccessible areas for geophysical surveys from surface ships. Submarines can easily navigate below the Arctic ice, but a safer sound source must be identified to avoid air guns or explosives. An analytic formulation was developed to predict the maximum basement detection depth of an under-ice sound source. This formulation was implemented to compare candidate underwater transducers and impulsive sound sources. Results from this study will be presented with discussions on additional constraints affecting the selection process. A workable submarine system for defining territorial boundaries would also help increase our understanding of the Arctic environment and facilitate exploration for potential natural resources. [Funding provided via UAF sub-award under NOAA Grant No. NA09NOS4000262.]

Session 2aUWb

Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Measurement, Characterization, and Mitigation of Underwater Anthropogenic Noise I

Peter H. Dahl, Cochair
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George V. Frisk, Cochair
Florida Atlantic Univ., Dept. of Ocean Engineering, 101 N. Beach Rd., Dania Beach, FL 33004-3023

Lisa M. Zurk, Cochair
Portland State Univ., Electrical and Computer Engineering Dept., 1900 S.W. Fourth Ave., Portland, OR 97207

Chair's Introduction—8:25

Invited Papers

8:30

2aUWb1. Anthropogenic noise and effects on marine life: A survey of literature and metadata. Christine Erbe (JASCO Appl. Sci., Brisbane Technol. Park, 1 Clunies Ross Ct., Eight Mile Plains, Queensland 4113, Australia) and Ed Urban (Univ. of Delaware, Newark, DE 19716)

The Scientific Committee on Oceanic Research and Partnership for Observation of the Global Oceans have initiated an effort to compile and review the literature on underwater noise with focus on the past 20 years. Literature on anthropogenic noise, ambient noise, and sounds made by marine organisms will be surveyed. Metadata concerning sound profiles of underwater noise sources will also be collected. Results will be made publicly available and links to the original literature and data will be provided where possible. The goal is to provide a baseline of information that can be used as a foundation for new cooperative international research on sound in the ocean as part of an International Quiet Ocean Experiment. An initial synthesis of the literature and any trends in research directions and gaps in research will be presented.

8:55

2aUWb2. Acoustic radiation during marine pile driving. Per G. Reinhall (Dept. of Mech. Engr., Univ. of Washington, MS 352600, Seattle, WA 98195, reinhall@uw.edu) and Peter H. Dahl (Univ. of Washington, MS 355640, Seattle, WA 98195)

Pile driving in water produces extremely high sound levels in the surrounding under water environment. Sound levels as high as 220 dB re 1 μPa are not uncommon 10 m away from a steel pile as it is driven into the sediment with an impact hammer. The primary source of underwater sound originating from pile driving is associated with compression of the pile. The pile is struck and the Poisson effect produces a radial displacement motion in the pile that will propagate downward at a computed speed comparable to but less than the longitudinal wave speed in steel. It is shown, using both finite element analysis and modeling based on the parabolic wave equation, that this radial motion of the pile is responsible for the ensuing high underwater sound pressures. It is also shown that the radial motion of the pile is transmitted into the water, either directly from the pile or indirectly via the bottom sediment that is in contact with the pile. A dominant feature of the resulting sound field is an axisymmetric Mach cone with apex traveling along with the pile deformation wave front.

9:20

2aUWb3. Comparative performance of attenuation treatments designed to mitigate underwater noise from pile driving. Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Peak sound pressure levels generated by large cast-in-shell steel (CISS) piles being driven by impact hammers into a sedimentary bottom can exceed 180 dB *re* 1 μa out to a 1-km range. Most solutions for mitigation of sound radiation from piles involve getting air around them. Three treatments are theoretically evaluated: a bubble curtain, a bare steel sleeve, and an "air-wrapped" steel sleeve. Attenuation by bubble curtains is a function of thickness and void fraction; however, to achieve high attenuation at low frequencies, bubbles must large, which complicates the design. A bare steel sleeve provides little attenuation underwater at low frequencies. From 550–800 Hz, frequencies containing the most acoustic energy for radiation from 36- and 48-in. diameter CISS piles, it is only 1 dB.

Adding an air wrap on the inside and/or outside walls of the sleeve decouples it from the water so it can act more like a rigid body to provide higher transmission loss. Along with the air-water interfaces, the air-wrapped sleeve results in 35–40 dB attenuation. So how can air be included in an underwater structure to provide sound attenuation? A correlation between this analysis and effectiveness of current field techniques will be presented.

Contributed Papers

9:45

2aUWb4. Observations and parabolic wave modeling of underwater pile driving impact noise. Peter H. Dahl (Dept. of Mech. Eng. and Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105) and Per G. Reinhall (Univ. of Washington, Seattle, WA)

Pile driving in water produces extremely high sound levels in both surrounding air and underwater environments. In a companion work [Reinhall and Dahl] it is shown using finite element simulation that for underwater case the primary sound signal originates from a compression wave traveling down the pile at a speed in excess of Mach 3. In this work, we present measurements pile driving impact noise made from a marine construction site in Puget Sound using a vertical line array (VLA) positioned at ranges 8–15 m from full-scale impact pile driving. The measurements are modeled using the parabolic wave equation approach for which synthetic time series are generated (bandwidth 50–2050 Hz). The simulation is achieved by way of a phased array of point sources, representing one source traveling down the pile at supersonic speed. Pile end reflections are included and the process is repeated with both an up- and down-traveling time-delayed sources. With the field computed in this manner, excellent agreement is achieved between model and observations of peak pressure level, and the compression wave speed is also confirmed by way of arrival angle estimation using the VLA. Implications on transmission loss are also discussed. [Work supported by the Washington State Department of Transportation.]

10:00—10:15 Break

10:15

2aUWb5. Acoustic transmission loss in industrial pile driving. Mark L. Stockham (Dept. of Mech. Eng., Univ. of Washington, Stevens Way, Box 352600, Seattle, WA 98195, stockham@u.washington.edu), Peter H. Dahl (Univ. of Washington, Seattle, WA 98195), and Per G. Reinhall (Dept. of Mech. Engr., Univ. of Washington, Seattle, WA 98195)

Industrial pile driving is a source of high-level underwater noise and new understanding of its effects on the behavior and health of marine mammals and fish has motivated numerous regulations intended to limit these effects. Of primary importance is to identify a suitable transmission loss model to predict where, given a certain source level, the noise produced by the pile driving reaches regulatory thresholds. In November 2009, data were collected from a marine construction site in the Puget Sound. Measurements at two ranges (8 and 12 m) from the pile being driven were taken using a nine hydrophone vertical line array (VLA). Concurrently, at a range of approximately 120 m, there was also a single hydrophone at a depth of 5 m (sensitive to frequencies greater than 10 kHz). By comparing the levels at the VLA to the more distant hydrophone across a number of pile strikes (each forming a identifiable short- and far-range pair), the transmission loss can be estimated. These results are in turn modeled using an approach based on the parabolic wave equation. [Work was funded by the Washington Department of Transportation.]

10:30

2aUWb6. Measurement of the underwater noise levels generated from marine piling associated with the installation of offshore wind turbines. Pete D. Theobald, Stephen P. Robinson (Natl. Physical Lab., Hampton Rd., Teddington TW11 0LW, United Kingdom), Michael A. Ainslie, Christ A. F. de Jong (Dept. TNO Acoust., 2628 CK Delft, The Netherlands), and Paul A. Lepper (Loughborough Univ., Leicestershire LE11 3TU, United Kingdom)

Marine piling is the most commonly used method for the installation of offshore wind turbines in the shallow coastal waters in the UK and consists of steel mono-piles being driven into the seabed using powerful hydraulic hammers. This is a source of impulsive sound, of potentially high level, that can travel a considerable distance in the water column and has the potential for impact on marine life. This presentation describes methodologies developed for measurement of marine piling and for the estimation of the energy source level. Measurements are presented for piles of typically 5 m in diameter driven by hammers with typical strike energies of 1000 kJ. Data were recorded as a function of range from the source using vessel-deployed hydrophones, and using fixed acoustic buoys that recorded the entire piling sequence, including soft start. The methodology of measurement is described along with the method of estimation of the energy source level. Limitations and knowledge gaps are discussed.

10:45

2aUWb7. Pile driving noise: Source level and sound generation mechanisms. Christ A. F. de Jong (TNO, Stieltjesweg 1, 2628 CK Delft, The Netherlands, christ.dejong@tno.nl), Mario Zampolli, Michael A. Ainslie, Erwin W. Jansen, Laurent Fillinger, Fred M. Middeldorp (TNO, 2628 CK Delft, The Netherlands), Richard A. Hazelwood (R&V Hazelwood Assoc., Guildford GU2 8UT, United Kingdom), Stephen P. Robinson (NPL, Teddington TW11 0LW, United Kingdom), and Bob Jung (IHC Hydrohammer, Kinderdijk, The Netherlands)

Offshore percussive pile driving generates loud impulsive sounds in the marine environment, sometimes characterized in terms of their "source level," a measure of acoustic pressure in the far field of its source. Special difficulties with the definition of source level for this application are described. Environmental assessment of future building projects requires a proper understanding of the sound generation mechanisms and their dependence on the various parameters (e.g., hammer type, stroke energy, pile construction, and sediment properties). Progress is reported on a combined numerical and experimental approach to modeling the sound generation and propagation through the medium. The acoustic response of the pile and its near field in fluid and sediment are calculated with a finite element model. The far field is obtained by coupling the acoustic field near the pile, from the finite element model, to a horizontal wavenumber integration model. Measurements carried out on a test pile in shallow water at IHC Hydrohammer in Kinderdijk are presented. Synchronous recording of hammer sensor data, pile vibrations, sound pressure, and sediment surface vibrations, close to the pile and at larger distance from the pile, enable a better understanding of the mechanisms and make it possible to check the model results.

11:00

2aUWb8. Standards for measurement and reporting of underwater sound: Application to the source level of trailing suction hopper dredgers. Christ A. F. de Jong, Michael A. Ainslie, Erwin W. Jansen (TNO, Stieltjesweg 1, 2628 CK Delft, The Netherlands, christ.dejong@tno.nl), and Benoit A. J. Quesson (TNO, Den Haag, The Netherlands)

Increasing concern exists for possible effects of low frequency anthropogenic sound in the sea with particular emphasis on shipping, seismic surveys, off-shore construction, and underwater explosives. The need to make like with like comparisons between different sounds, sounds measured in different places, or sounds measured by different monitoring equipment is being addressed by means of a trans-national collaboration between national institutes from The Netherlands, United Kingdom, and Germany with the objective to develop a common measurement and reporting standard. Initial progress is reported with the emphasis on developing a common terminology, including the definition of "source level," for which no widely accepted definition exists. A method to implement this definition has been developed and is applied to the measurement of seven trailer suction hopper dredgers undergoing various activities involving dredging, rainbowing, sand dumping and pumping ashore, and transit between sites. In the frequency range 30 Hz to 8 kHz, the measured third octave band source levels of the dipole formed by the ship source and its surface image, averaged over three angles between 15 and 45 deg from the horizontal (chosen for consistency with ANSI standard S12.64-2009), range approximately between 133 dB and 185 dB *re* 1 $\mu Pa^2\ m^2$.

11:15

2aUWb9. Effect of dredging, traffic, wind, and fish on ambient noise close to the Port of Rotterdam. Michael A. Ainslie, Christ A. F. de Jong (TNO, Stieltjesweg 1, 2628 CK Delft, The Netherlands), Jeroen Dreschler, Groen Wim (TNO, 2509 JG The Hague, The Netherlands), and Paul A. van Walree (FFI, 3191 Horten, Norway)

Concern exists for possible effects of anthropogenic sound in the sea. Measurements are presented from monitoring of underwater noise associated with dredging activities close to the Port of Rotterdam [http://www.maasvlakte2.com/en/index/]. Measured spectra in the approximate frequency range 30 Hz to 50 kHz are presented in two 1-week average snapshots, one before construction began (September 2008) and the other during construction (September–October 2009). Correlations are described of third-octave band noise level with wind speed on the one hand and with a parameter representing the density of nearby shipping on the other. The correlations are consistent with noise spectra dominated by wind noise at high frequency and shipping traffic noise at low frequency; at intermediate frequencies (roughly in the range 2 to 20 kHz), the sound probably originates from nearby ships with sea surface scattering playing a role in damping the sound in the presence of wind-generated breaking waves and near-surface bubbles. In 2009, a strong correlation between noise level and time of day was observed in the third-octave bands centred at 2.5 and 3.15 kHz with night-time levels exceeding day-time levels by about 10 dB. This day-night difference is attributed to a fish absorption line, possibly due to sprat.

11:30

2aUWb10. Underwater radiated noise measurements of a noise-reduced research vessel: Comparison between a U.S. Navy noise range and a simple hydrophone mooring. Alex De Robertis (Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Seattle, WA 98115, alex.derobertis@noaa.gov), Christopher D. Wilson (Alaska Fisheries Sci. Ctr., Seattle, WA 98115), and Peter H. Dahl (Univ. of Washington, Seattle, WA 98195)

[A feasibility study was undertaken to characterize underwater radiated noise for a new class of noise-reduced fisheries research vessels using a field-deployable hydrophone system. Recent studies have demonstrated that vessel-radiated noise can impact the behavior of fish, and that periodic monitoring of survey-vessel radiated noise is desirable to characterize potential biases in fish abundance estimates. Vessel radiated noise is traditionally measured at naval ranges, but lower-cost options are desirable. Beam aspect measurements of a noise-reduced vessel made at a U.S. Navy noise range are compared to those made using an experimental mooring equipped with commercially available instrumentation. Hydrophone depths and distance-to-the-vessel were comparable for the mooring and those used at the Southeast Alaska Acoustic Measurement Facility (SEAFAC). SEAFAC and mooring measurements were taken within a day of one another. Data processing was consistent with the recent American national standard for measurement of underwater sound from ships (ANSI/ASA S12.64-2009/Part 1). The measurements from the experimental mooring were precise and comparable to those made at SEAFAC. This suggests that reliable measurements suitable for monitoring the underwater radiated noise of vessels with low source levels can be made in the field.

11:45

2aUWb11. Mitigation of low-frequency underwater sound using large encapsulated bubbles and freely-rising bubble clouds. Kevin M. Lee, Kevin T. Hinojosa, Mark S. Wochner (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029), Theodore F. Argo, IV, and Preston S. Wilson (The Univ. of Texas at Austin, Austin, TX 78712-0292)

Low-frequency anthropogenic noise may affect marine life, motivating the need to minimize its potential impact. Bubbles cause significant dispersion and attenuation of underwater sound at frequencies near the individual bubble resonance and can potentially be used to abate this noise. Such effects have been reported for large encapsulated bubbles with resonance frequencies below 100 Hz, and significant attenuation due to bubble resonance phenomena and acoustic impedance mismatch was observed in a tank experiment [J. Acoust. Soc. Am. **127**, 2015 (2010); **128**, 2279 (2010)]. Both of these mechanisms were found to significantly reduce down-range radiated acoustic pressure, as much as 40 dB, at low frequencies (60 to 1000 Hz) in a series of lake experiments where a sound source was surrounded by an array of tethered resonant toroidal air bubbles, a cloud of freely-rising sub-resonant bubbles, and various combinations of the two. Hydrophones were placed at various depths and ranges to determine the effect of the bubbles on the radiated field. The effects of void fraction and bubble size variation on the spectrum of the radiated sound were also investigated. [Work supported by Shell.]

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC1, Noise,
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machines,
ISO/TC 108/SC 6, Vibration and shock generating systems,
and
IEC/TC 29, Electroacoustics

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

D. J. Evans, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108 Mechanical vibration, shock and condition monitoring, and ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

W. C. Foiles, Co-Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
BP America, 501 Westlake Park Boulevard, Houston TX 77079

R. Taddeo, Co-Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

D. D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, NV 89120

D. J. Vendittis, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
701 Northeast Harbour Terrace, Boca Raton, FL 33431

R. Taddeo, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

C. Peterson, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 6 Vibration and shock generating systems
200 Dixie Ave., Kalamazoo, MI 49001

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Drive, Stop 8221, Gaithersburg, MD 20899–8221

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3 and S12, which are scheduled to take place in the following sequence:

Tuesday, May 24. 2011	10:30 a.m. — 12:00 noon	ASC S2, Mechanical Vibration & Shock
Tuesday, May 24, 2011	1:45 p.m. — 3:15 p.m.	ASC S1, Acoustics
Tuesday, May 24, 2011	3:30 p.m. — 5:00 p.m.	ASC S12, Noise
Wednesday, May 25, 2011	9:00 a.m. — 10:30 a.m.	ASC S3, Bioacoustics
Wednesday, May 25, 2011	10:45 a.m. — 12:00 noon	ASC S3/SC 1, Animal Bioacoustics

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SC1** Noise	ASC S12
D.J. Evans, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
W.C. Foiles, Co-Chair R. Taddeo, Co-chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair R. Taddeo, Vice Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machines	ASC S2
C. Peterson, Chair	**ISO/TC 108/SC6** Vibration and shock generating systems	ASC S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

TUESDAY MORNING, 24 MAY 2011 QUEEN ANNE, 10:30 A.M. TO 12:00 NOON

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

A. T. Herfat, Chair, ASC S2

Emerson Climate Technologies, Inc., 1675 West Campbell Road, PO Box 669, Sidney, OH 45365-0669

C. F. Gaumond, Vice Chair, ASC S2

Naval Research Laboratory, Code 7142, 4555 Overlook Avenue SW, Washington DC 20375-5320

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and its five subcommittees, take note — that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 24 May 2011.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

Session 2pAAa

Architectural Acoustics: 25th Anniversary of Newman Fund Awards–Recipient and Participating School Updates

William J. Cavanaugh, Cochair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Rd., Sudbury, MA 01776

Michelle C. Vigeant, Cochair
Univ. of Hartford, Mechanical Engineering, 200 Bloomfield Ave., West Hartford, CT 06117-1599

Chair's Introduction—1:00

Invited Papers

1:05

2pAAa1. A short history: The Newman student awards program. William J. Cavanaugh (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776) and Michelle C. Vigeant (Univ. of Hartford, Mech. Eng., 200 Bloomfield Ave., West Hartford, CT 06117-1599)

This paper traces the Newman Award Fund programs from the first student medals awarded in 1986 to 13 recent awards in "for merit in the study of Architectural Acoustics" bringing the total to 225 medals awarded to students at over 55 schools of architecture, architectural engineering, and music to date. Any degree granting institution offering a basic course in architectural acoustics and opportunities for students to apply their knowledge in student theses, senior design projects, or research may nominate one graduating student for the Newman Medal from that school each year. In addition, every 2 or 3 years the Fund awards Theodore Schultz Grants to teachers and researchers in architectural acoustics for merit in the development of course materials, innovative course programs, or new texts. A third leg of the Fund program is the Wenger Prizes awarded to student teams for meritorious design in competitions sponsored each year by the ASA Technical Committee on Architectural Acoustics. This paper brings the perspectives and experiences of a charter member of the Newman Fund Advisory Committee as well as those of a more recent member of the Committee, who is a past Newman medalist in her own right. For Newman Fund updates visit www.newmanfund.org.

1:20

2pAAa2. Following hallowed footsteps and new paths. Carl J. Rosenberg (Acentech, 33 Moulton St., Cambridge, MA 02138 crosenberg@acentech.com)

Robert Bradford Newman taught architectural acoustics for over 30 years at Harvard and M.I.T. and was a frequent lecturer at numerous other schools. He left an enduring legacy of engaging pedagogy, professionalism, diligence, and wit. He did this in a manner that is remembered still today by former students. This paper acknowledges ways that the legacy has been fostered in on-going courses at M.I.T. (by the author who succeeded Newman at M.I.T.) and elsewhere. A broader view is also addressed with regard to the relationship of architectural acoustics education to developments in building technologies, new trends in engaging students in acoustics, and possible realignment of the process of making architects aware of their professional responsibilities to the aural environment.

1:35

2pAAa3. The Newman medal and architectural acoustics at the University of Florida. Gary W. Siebein (Univ. of Florida School of Architecture/Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607) and Bertram Y. Kinzey, Jr. (Univ. of Florida, Blacksburg, VA 24060)

The architectural acoustics program at the University of Florida was heavily influenced by the Newman medal program as well as the life and work of Robert B. Newman. Bertram Y. Kinzey, Jr., the founder of the program, had the pleasure of attending one of Bob Newman's seminars and brought much of Bob's enthusiasm for the subject to all of his students. Four principles that organize the program were influenced by the life and work of Bob Newman. The program is based in architecture as the fundamental discipline from which to approach architectural acoustics. The creative application of acoustical design principles in buildings is the focus of study. The exploration of acoustics in actual buildings becomes a thrust for research in the field. The program teaches future architects about acoustics in the built environment. A fifth principle that introduces a strong research component is necessary in a modern university. We are extremely grateful for the recognition for excellence in the study of architectural acoustics that we have been able to extend to our students as a result of the Newman Award program.

1:50

2pAAa4. The Newman Student Award Fund, a major contributor to architectural acoustics education. Robert C. Coffeen (School of Architecture, Design and Planning, Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045, coffeen@ku.edu)

The creation of the Newman Student Award Fund in honor of Bob Newman by Mary Newman and some of Bob Newman's friends and colleagues has been of considerable help in interesting and rewarding students who intend to pursue a career in professional acoustical consulting. The Fund provides the Robert Bradford Newman Student Medal for Merit in Architectural Acoustics and the Schultz Grant for educators in architectural acoustics, and it is a sponsor of the Annual Acoustical Society Student Design Competition. From the architecture and the architectural engineering programs at the University of Kansas and during the past 14 years, over 30 KU students plus several students from architectural engineering at Kansas State University have been guided into acoustical consulting or closely related fields. KU students have received at least one of the five awards from the student design competition since its beginning in 1994, and a number of Newman medals have been awarded. As a member of the Newman Student Award Fund Advisory Committee, it is a privilege to assist with the continuation of this excellent organization.

2:05

2pAAa5. Influence of Robert B. Newman on education. M. David Egan (Clemson Univ., Clemson, SC 29631)

Professor R. B. Newman's 4.43 course at MIT followed the scholarly legacy of W. C. Sabine, V. O. Knudsen, and C. P. Boner, but featured irreverent observations, succinct admonitions, case histories of BBN projects, and field trips (which stressed importance of sensory observations to understand principles of physics). In the late 1960s, the presenter, a former student of Newman, developed teaching aids at Tulane University which evolved into architectural acoustics, now in print for its 41st year. In the first 7 years of Newman Award Fund, the number of participating schools grew due to face-to-face communication with faculty and school administrators. The presenter and W. J. Cavanaugh facilitated the initial growth primarily at Clemson University (6 medalists), University of Florida (6), Rhode Island School of Design (5), and Oklahoma State University (4). Examples will be cited to show how Newman Award enhanced status of acoustics educators in designer-dominated academic cultures.

2:20

2pAAa6. Acousticatalyst. Daniel Butko (College of Architecture, The Univ. of Oklahoma, 504 W. Main, Norman, OK 73069)

Architectural acoustics, generally the result of meticulous calculations and implementation, can occasionally be unintentional byproducts of material choices and spatial relationships. Such an example of fortuitous acoustics exists in downtown Tulsa, OK, commonly known as the center of the universe. This acoustical anomaly has entertained all walks of life for many years, bringing laughter and astonishment to numerous unexpected or doubtful inhabitants. Regardless of someone's occupation, education, or preconceived thoughts, the acoustical results within this space are noticeable and intriguing. Numerous studies have been performed attempting to decipher the somewhat mysterious amplification of sound as sources radiate outward from the center of the circle. Research has proven both fundamental and advanced acoustical lessons, but the acoustical element as a "found object" in design sparks common interest to investigate further. While visiting Tulsa for an unrelated studio project, a group of University of Oklahoma design students became intrigued to visit the location due solely to the premise of something extraordinary. Those students are now suddenly interested in architectural acoustics. This paper focuses on this type of acoustical phenomenon as a catalyst to spark creative thought and interest in designers that previously placed little or no importance on acoustical design.

2:35

2pAAa7. Wenger's history with Newman Fund Award (student design competition). Ronald R. Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060)

A unique historical connection exists between Bob Newman and the Wenger Corporation. Before founding the Wenger Corporation in 1946, Harry Wenger was a dynamic band director with a talent for inventing solutions to improve music education. This passion included an appreciation of the fundamental importance of acoustics for rehearsal and performance. He wrote "In all too many cases, the [musicians'] fine preparations are seriously impaired by the poor acoustics of the stage or area from which the group is expected to perform." To remedy that problem, Wenger began developing portable acoustical shells, engaging Bob Newman as a consultant during the 1960s. Newman visited Wenger's Minnesota headquarters on a number of occasions, sharing animated discussions with Harry and Harry's son, Jerry. The same passion for music and acoustics central to the Wenger Corporation's founder continues today. The relationship with Bob Newman has evolved to encompass well-known names in acoustics such as Ron McKay, Larry Kierkegaard, Bill Cavanaugh, Chris Jaffe, and BBN's successor firm, Acentech. The Wenger Foundation, funded by the Wenger Corporation, has served as ongoing award sponsor for the Student Design competition. Supporting the efforts of students working in music and acoustics is a logical, natural extension of Wenger's passion.

2:50—3:05 Break

3:05

2pAAa8. An international perspective on the work of the Newman fund. Stephen Dance (Dept. of Urban Eng., London South Bank Univ., Borough Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk)

Since 2007 London South Bank University has been involved with the Newman Fund through their Masters programe in Environmental and Architectural Acoustics. This link has lead to closer relations between London South Bank University and the Acoustical Society of America starting with a Schultz Grant award, followed by attendance at the fourth Concert Hall Research Group Summer Institute, and recently in Dr. Dance becoming the first international member of Newman committee. Each year the best dissertation has been submitted to the Newman committee, and fortunately nearly every year one of our acoustic students has been awarded the Newman medal. The Acoustics Group has used the Newman fund to encourage the award-winning students to go on to our Ph.D. research programe. The most effective inducement has been found to be sending the prospective Ph.D. student to an international conference, where a research paper is presented based on their Masters dissertation. Case studies of the awarding winning projects are briefly outlined, emphasizing the link between the Masters dissertation and the subsequent student selected Ph.D. topic.

3:20

2pAAa9. The Newman medal: A recipient's recollections after 20 years. Gary Madaras (Bldg. Momentum Group, 4849 S. Austin Ave., Chicago, IL 60638, gmadaras@bmgsc.com)

Gary Madaras received the Newman medal 20 years ago (1991) for his Master of Architecture thesis work at Kent State University, OH. It was awarded at a pivotal time, as his academic and professional direction was changing from designing the visual environment to designing the aural environment. Dr. Madaras reflects on the circumstances that contributed to this lifelong redirection and on the studies for which he received the Newman medal. An overview of his career in architectural acoustics since receiving the Newman medal is followed by a heartfelt thank you to those that made, and still make, the program possible two decades later.

3:35

2pAAa10. Beyond architectural schools. Anthony K. Hoover (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., 325 Westlake Village, CA 91362)

The legacy of Bob Newman and Ted Schultz has benefitted architects, acousticians, and many others. This paper will summarize some of the author's efforts to carry on their pedagogical spirit. The architectural acoustics class at Berklee College of Music began in 1987, with nearly 3500 total students over the years. A number of those students have been Newman Awardees, and some students have become teachers and even Newman Award mentors. The text "An Appreciation of Acoustics", with funding support from the Schultz Grant, was written primarily for the Berklee class, and has been used elsewhere. The ASA College of Fellows recently began the Outreach to Undergraduate Students program, which has effectively introduced acoustics as a potential vocation to students in various schools and curricula.

3:50

2pAAa11. University of Hartford's Newman Fund Award recipients and the history of its undergraduate acoustical engineering programs. Robert D. Celmer and Michelle C. Vigeant (Acoust. Program and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117)

The University of Hartford has two ABET-accredited undergraduate engineering programs in the area of acoustics: (1) the Bachelor of Science in Mechanical Engineering with Acoustics Concentration and (2) the Bachelor of Science in Engineering with a major in Acoustical Engineering & Music. Both programs encompass the same engineering vibrations and acoustics courses as well as the same acoustics projects sequence. Alumni of both undergraduate programs have successfully obtained positions in consulting (architectural and environmental), audio product and A/V design, musical instrument design, hearing- and psychoacoustic-related design, noise control, as well as pursued graduate degrees. Projects that have received Newman Fund awards have included investigations of auralization accuracy, just noticeable difference of clarity index (C80), and the relationship between room absorption and late lateral sound level (GLL). Alumni updates on four Newman Medalists will be presented.

4:05

2pAAa12. From Portugal to Florida, and the Newman award. Antonio Carvalho (Lab. of Acoust., Dept. of Civil Eng., Univ. of Porto, 4200-465 Porto, Portugal, carvalho@fe.up.pt)

I was 31 years-old when, in 1991, I left the University of Porto to study architectural acoustics at the University of Florida with Professor Gary W. Siebein. In 1994, I finished my Ph.D., I got the Robert Bradford Newman Award, and my career changed. In fact, that 1994 would be the "big bang" of my future life. Personally, academically, and scientifically, my Universe blew up and began expanding. What I learned and what I taught in the 17 years after that (and what the architectural acoustics changed in Portugal and Southern Europe) is the subject of this paper.

4:20

2pAAa13. A review of the Robert Bradford Newman Student Award Fund's impact on the University of Nebraska Acoustics Group. Lauren M. Ronsse and Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska-Lincoln, Omaha, NE 68182-0816, lwang4 @unl.edu)

The Robert Bradford Newman Student Award Fund has significantly impacted the acoustics group within the Architectural Engineering Program at the University of Nebraska-Lincoln over the past 10 years since the group was founded. Many of our students have received the Newman Student Medals. Students have also participated in the annual Student Design Competition, co-sponsored by the Newman Fund and the ASA Technical Committee on Architectural Acoustics. A Schultz Grant, disbursed by the Newman Fund, was awarded to our group in the year 2000 through which the webpage "concerthalls.org" was developed. This paper will review the positive influence that these three initiatives sponsored by the Newman Fund have had on the Nebraska Acoustics Group.

4:35 Reception

Session 2pAAb

Architectural Acoustics: Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture II: Acoustics as a Factor of Ergonomics

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

William J. Cavanaugh, Cochair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Rd., Sudbury, MA 01776

Chair's Introduction—4:40

Invited Papers

4:45

2pAAb1. Acoustics as a factor of ergonomics: Communication behavior and workload of pupils and teachers in highly absorbent classrooms. Markus Oberdoerster (Saint-Gobain Ecophon GmbH, Taschenmacher Str. 8, 23556 Luebeck, Germany, markus.oberdoerster@ecophon.de) and Gerhart Tiesler (ISF Bremen, 28199 Bremen, Germany)

This lecture refers to an interdisciplinary research project carried out from 2000 to 2006 by the Bremen University, Germany. A mixed team of acousticians, occupational and medical scientists, and pedagogues investigated the kind of work and communication behavior in classrooms in two elementary schools. Using a database of 175 examined lessons an analysis is made of how different kinds of work (frontal lessons versus differentiated lessons) affect the basic and working sound level in the classroom. Parameters are discussed, which can describe classroom acoustics appropriately. Also discussed are how altered room characteristics (e.g., increased absorption, shortened reverberation time, and improved speech intelligibility) affect the sound level in the context of each kind of work. A methodical examination of the database allows not only an assessment of mean values but also of the detailed teaching phases, as characterized by certain pedagogical factors. The results provide the basis for discussion of stress and work demands of teachers: Based on recordings of teacher's heart rate the effects of noise level on the workload of the teachers as a stress reaction and a factor of fatigue are analyzed.

TUESDAY AFTERNOON, 24 MAY 2011 ISSAQUAH, 1:25 TO 4:45 P.M.

Session 2pABa

Animal Bioacoustics: Memorial Session in Honor of Ronald Schusterman and David Kastak II

Patrick W. Moore, Cochair
National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106

Robert Gisiner, Cochair
OPNAV N45, Navy Energy and Environmental Readiness Div., Arlington, VA 22202

Roger M. Gentry, Cochair
ProScience Consulting LLC, 22331 Mt. Ephraim Rd., Dickerson, MD 20842

Chair's Introduction—1:25

Invited Papers

1:30

2pABa1. The nature and nurture of seeing with sound: The role of learning in biosonar. Robert Gisiner (Navy Energy and Environ. Readiness Div. (OPNAV N45), Arlington, VA 22202, bob.gisiner@navy.mil) and Colleen Reichmuth (Univ. of California, Santa Cruz, CA 95064)

Echolocating dolphins achieve performances in detecting and classifying components of their environment that rival the performances we typically associate with vision. Many researchers have referred to dolphins "seeing with sound" or forming internal "images" of the world via acoustics, implying a presumed isomorphy of sensory inputs and a common representation at higher levels of processing in the brain. But some aspects of acoustics do not translate into visual equivalents (hollow objects and objects of different materials) and some visual aspects of an object do not translate into acoustic equivalents (color and brightness). So, how is this cross-

modal sensory translation achieved? Is it hard-wired into the anatomy of the brain, learned through association, or is it some combination of the two? Experimental tests of nature versus nurture in cross-modal sensory performance are reviewed, as first explored by Schusterman and Kastak. Further studies are suggested to reveal the respective roles of neuroanatomy and associative learning in the formation of a dolphin's perceptual and conceptual world.

1:50

2pABa2. The dolphin's mental representation during echolocation: Ron Schusterman and the email debate between the "seeing through sound" and "associative learning" hypotheses. Brian K. Branstetter (Natl. Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmfoundation.org) and Jason Mulsow (U.S. Navy Marine Mammal Program, San Diego, CA 92152)

[Evidence suggests that detection, discrimination, and recognition abilities of dolphin echolocation are related to perceived differences in time, frequency, and amplitude information from received echoes. Although an acoustic analysis has proven successful for explaining simple experimental paradigms, no similar analysis has successfully accounted for the findings from a series of cross-modal matching experiments. In cross-modal "identity" matching, the dolphin is first presented with a sample object to either vision or echolocation only. The dolphin must then select a matching object from a number of nonmatching alternative objects presented to the opposite sensory modality. The "seeing through sound" hypothesis claims that immediate (i.e., first-trial) cross-modal matching is evidence that dolphins perceive object shape through echolocation while an alternative hypothesis states that successful performance is the result of the animal's "associative learning" history. Evidence against, or in support, of both alternative hypotheses is critically examined and the dolphin's mental representation during echolocation is discussed.

2:10

2pABa3. Auditory scene analysis in the echolocating dolphin. Patrick Moore (Natl. Marine Mammal Foundation, 2240 Shelter Is. Dr., San Diego, CA 92106) and James J. Finneran (US Navy Marine Mammal Program, SSC Pacific, 53560 Hull St., San Diego, CA 92152-5001)

Auditory scene analysis (ASA) refers to an animal's ability to organize acoustic information in order to construct an understanding of its environment. In most mammals, vision is the primary sensory system and audition plays a secondary role; however, in the echolocating dolphin the reverse is likely true. One underlying tenant of ASA is *stream* analysis. This is the ability of an animal to integrate acoustic information over time and depends on short term memory for acoustic events. Moss and Surlykke [J. Acoust. Soc. Am. **110**, 2207 (2001)] demonstrated that the bat echolocation perceptual system possesses the minimum requirements for stream analysis. This experiment replicates the bat study and tests the dolphins ability to assemble information about changing echo delay by discriminating between phantom targets of varying delays. Phantom targets are presented using a Phantom echo generator system. Emitted signals are received via contact melon hydrophone and delayed echoes transmitted directly to the dolphin lower jaw. Will the dolphin perform as well as the bat?

2:30

2pABa4. Critical bandwidths in echolocating porpoises, dolphins and whales. Paul Nachtigall (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734)

Odontocete cetaceans may differ from most mammals in their response to noise. A close look at the published data [Popov *et al.* (2006)] of the critical bandwidths of two species of porpoises, the harbor porpoise (Phocaena phocoena) and the finless porpoise (Neophocaena phocaenoides), shows constant bandwidth critical bands in the high frequency area where echolocation signals are processed. A further look at harbor porpoise critical band data [Kastelein *et al.* (2009)] can be interpreted to show a mixture of constant Q bandwidths at lower frequencies and constant bandwidth data at higher frequencies. Data from Lemonds *et al.* [1997] indicate that the bottlenosed dolphin shows typical mammalian constant Q filters in lower frequency whistle areas but shifts to constant bandwidths in the areas of high frequency where echolocation discrimination processing is assumed to occur. Recent work (Kloepper *et al.*) has shown substantial loss in echolocation discrimination performance in the false killer whale with the loss of high frequency hearing. General high frequency hearing loss due to noise in the environment may particularly affect the echolocation processing capabilities of odontocetes and thus the foraging capabilities and fitness of odontocete echolocators. [Work funded by the Office of Naval Research.]

2:50

2pABa5. Dolphin response time in vocal reporting of echolocation targets. Ridgway Sam (Natl. Marine Mammal Foundation, 2410 Shelter Island Blvd., San Diego, CA 92106), Wesley Elsberry, Diane Blackwood (Florida Fish and Wildlife Conservation Commission, 100 SE 8th Ave., St. Petersburg, FL 33701), T. Kamolnick, Mark Todd, Don Carder (Natl. Marine Mammal Foundation, San Diego, CA 92106), and Ted Cranford (San Diego State Univ., San Diego, CA 92182-4614)

To study response time in echolocation, dolphins were trained to wear opaque suction cups over their eyes and to station on an underwater apparatus behind and acoustically opaque door. This put the dolphins in a known position and orientation. When the door opened, the dolphin produced clicks to identify the presence or absence of targets. Dolphin S emitted a whistle if the target was a 7.5 cm water filled sphere, she made a pulse burst if the target was a rock, and she remained quiet if there were no target present. Dolphin B whistled for the sphere but remained quiet for rock and for no target. Thus, S had to choose between three different responses, whistle, pulse burst, or remain quiet. B had to choose between two different responses, whistle or remain quiet. S gave correct vocal responses averaging 114 ms after her last echolocation click (range 18 ms before and 219 ms after the last click). Average response for B was 21 ms before her last echolocation click (range 250 ms before and 95 ms after the last click in the train). More often than not, B began her whistle response before her echolocation train ended.

3:10—3:30 Break

3:30

2pABa6. Biosonar beam formation in the bottlenose dolphin: Evidence for focusing in stages. Ted W. Cranford (San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-3422), Petr Krysl, and Vanessa Trijoulet (Univ. of California, San Diego, La Jolla, CA 92093-0085)

The idea that a dolphin biosonar beam is focused in a series of stages was put forth by Dr. Norris more than 50 years ago. Development of our finite element modeling (FEM) tools allowed us to test this hypothesis. We constructed an FEM model of a bottlenose dolphin (Tursiops truncatus) head from CT scans and tissue property measurements, and simulated the formation of the sonar beam. The sound transmission system within the dolphin forehead contains several tissue elements. The model allowed us to tease apart the functional contributions that these structures make to the formation of an echolocation beam. The simulations showed that the direction of the beam was consistent with prior biosonar investigations and illustrated that the narrowing of the sound transmission beam increases with various levels of refinement in structural (tissue) complexity. It appears as if each additional structure such as the melon, the air spaces, source location, and configuration adds to the effect of narrowing the beam, and their combined contribution is significant. [Work supported by the Chief of Naval Operations (Grant No. CNO45).]

3:50

2pABa7. Dolphin echolocation signals measured at extreme off-axis angles: Insights to sound propagation in the head. Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744), Patrick Moore, Brian Branstetter (Natl. Marine Mammal Foundation, San Diego, CA), and James Finneran (U.S. Navy Marine Mammal Program, San Diego, CA)

Echolocation signals radiated along the beam axis of an Atlantic bottlenose dolphin resemble single transient-like oscillations. As the azimuth of the measuring hydrophones in the horizontal plane progressively increases with respect to the beam axis the signals become progressively distorted. At approximately $\pm$ 45 deg, the signals begin to divide into two components with the time difference between the components increasing with increasing angles out $\pm$ 90 deg. The time difference between the two pulses measured by the hydrophone on the right side of the dolphin's head is on the average approximately 10 μs larger than the time differences observed by the hydrophone on the left side of the dolphin's head. The center frequency of the first pulse is generally lower by 33 to 47 kHz than the center frequency of the second pulse. When considering the relative locations of the two phonic lips, the data suggest that the signals are being produced by one of the phonic lips and the second pulse resulting from a reflection within the head of the animal. The data also indicate that the process of generating echolocation signals is a complex one and the exact mechanisms are not yet known.

4:10

2pABa8. Harmonic beamforming: Categorical perception segregates targets from clutter in bat sonar. James A. Simmons (Dept. of Neurosci., Brown Univ., Box G-LN, Providence, RI 02912 james_simmons@brown.edu), Mary E. Bates (Brown Univ., Providence, RI 02912), and Tengiz V. Zorikov (Inst. of Cybernetics, 0186 Tbilisi, Georgia)

Beaming for FM biosonar transmissions of big brown bats is broad and frequency-dependent (70 deg at 25 kHz to 30 deg at 80 kHz). In flight, sound reception is directional and frequency-dependent, too, being oriented to the front and centered on-axis. A broadcast beam defined by the harmonic ratio of FM2 to FM1 depicts the frontal zone for flat-spectrum ensonification surrounded by increasingly lowpass ensonification for objects located off to the sides and farther away. The width and flat front of the central lobe of this harmonic-ratio beam predict the bat's acoustic behavior during flights in obstacle arrays of different densities [Petrites *et al.*, JCP-A (2009)] and the spatial unmasking of target detection by relocating clutter to progressively larger horizontal separations from the target [Sumer *et al.*, JCP-A (2009)]. Amplitude-latency trading for responses to FM2 relative to FM1 in lowpass echoes causes echo-delay acuity to defocus, which also prevents masking from clutter. Only focused delay images can mask other focused images, which effectively excludes clutter echoes from causing masking. [Work supported by the ONR and the NIMH.]

Contributed Paper

4:30

2pABa9. Changes in spectrotemporal features of echolocation signals in multiple bat assemblages. Mary E. Bates (Dept. of Cognit., Linguistic, and Psychol. Sci., Brown Univ., Providence, RI 02912), Jeffrey M. Knowles, Jonathan R. Barchi, James A. Simmons (Brown Univ., Providence, RI 02912), Emyo Fujioka, Yu Watanabe, Yuto Furusawa, Shizuko Hiryu, and Hiroshi Riquimaroux (Doshisha Univ., Kyotanabe, Japan)

Echolocating bats face potential acoustical interference when flying and foraging near echolocating conspecifics. Quantifying changes in the signal structure of echolocation emissions between bats flying alone and within a larger group has proven difficult. Here, we use two new methodologies, an onboard radio telemetry microphone and a multiple microphone array to record the sounds of big brown bats (Eptesicus fuscus) flying alone and in small groups both in the laboratory and in the field. In a laboratory flight room, bats changed the ending frequency of their first harmonic when the conspecific with which they were paired emitted calls at a similar frequency. In the field, ending frequency and sweep shape were more variable among groups of bats flying together than among single bats compared to one another. The presence of other nearby bats did not have any effect on the timing of emissions or the interpulse intervals [Work supported by NIH, NSF, ONR, and JSPS.]

Session 2pABb

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Fish Bioacoustics II

Allison Coffin, Cochair
Univ. of Washington, Dept. of Otolaryngology-HNS, Seattle, WA 98195

David G. Zeddies, Cochair
Marine Acoustics, Inc., 4100 Faifrax Dr., Arlington, VA 22003

Invited Papers

1:15

2pABb1. Auditory evoked potential audiometry in fish. Friedrich Ladich (Dept. of Behavioural Biology, Univ. of Vienna, Althanstrasse 14, 1090 Vienna, Austria, friedrich.ladich@univie.ac.at)

A recent survey lists more than 100 papers utilizing the auditory evoked potential (AEP) recording technique for studying hearing in fish. More than 90% of these AEP-studies were published after Kenyon *et al.* introduced a non-invasive electrophysiological approach in 1998 allowing rapid evaluation of hearing and repeated testing of animals. Applying and further developing the AEP-technique enabled the investigation of a wide range of scientific questions. First, it was possible to describe and compare basic hearing abilities in a large number of species. Subsequently, the ontogenetic development of hearing as well as the influence of various accessory hearing structures (Weberian ossicles and swimbladder) were studied. The technique was also successfully utilized to analyze the temporal resolution ability of the auditory system. The AEP-technique is suitable for studying threshold shifts after exposure to (TTS) and in the presence of (masking) various noise types (white, ambient, and anthropogenic). Comparison of AEP-audiograms with sound spectra, along with the analysis of AEPs in response to conspecific sounds, enabled us to assess the ability to communicate acoustically in general, and during ontogeny and in the presence of noise in particular. Finally, various factors potentially influencing hearing such as temperature, albinism, and cave dwelling were investigated. [Work supported by the FWF.]

1:30

2pABb2. Behavioral and evoked potential audiograms for goldfish. Richard Fay (Dept. Psych., Loyola Univ., 6525 N. Sheridan Rd, Chicago, IL 60626)

Twelve recently determined evoked potential (EP) audiograms were compared to six behavioral (BEH) audiograms for the goldfish. There are no theories possible on the relationships between these two very different measurements of auditory response; nevertheless many workers desire to know how they compare so that the EP audiograms can possibly "stand in" for the BEH measures. The data from the many audiograms indicate that they are only very roughly similar in terms of sensitivity (variation ranging from 45 to 60 dB) and best frequency (variation ranging from 300 to 1500 Hz). This variation among audiograms, within and between measurement methods used, is discomforting. There are no trends indicating that this variation is due to a single factor such as near field acoustics (all audiograms were determined in the near field of the sound source), threshold definition, ambient noise levels, conditioning method, or type of sound source. The averaged audiograms differ in best frequency (EP=525 Hz; BEH=725 Hz) and best sensitivity (EP=72 dB; BEH=63 dB). Thus, the EP method results in audiograms that are roughly comparable to the BEH audiograms, except at frequencies higher than the best frequency where the EP method appears to overestimate sensitivity.

Contributed Paper

1:45

2pABb3. The effects of sex and reproductive condition on auditory evoked responses in the round goby, *Neogobius melanostomus*. Jeffrey N. Zeyl and Dennis M. Higgs (Dept. of Biology, Univ. of Windsor, 401 Sunset Ave., Windsor, Ontanio N9B 3P4, Canada zeylj@uwindsor.ca)

Effectively receiving and responding to mating signals is critically important in acoustically communicating vertebrate species, particularly for individuals that are reproductively mature. Behavioral responses to mating and advertisement signals in these individuals are often more intense than in their non-reproductive counterparts, which might be explained in part by changes in the response properties of the auditory system. The current study examined the potential for auditory plasticity of hearing sensitivity in relation to reproductive state in the round goby. This fish breeds in high density aggregations, with males guarding nests and producing low frequency (150–180 Hz dominant frequency) vocalizations that both males and females can localize. Auditory sensitivities in both sexes were assessed in reproductive and non-reproductive individuals using auditory evoked responses to tone pips that ranged from 100 to 600 Hz. Females were more sensitive than males across their hearing range. When assessed based on reproductive status, reproductive females were more sensitive than non-reproductive females and males at 100 Hz. Thresholds in males were similar between reproductive and non-reproductive individuals. These results suggest that hearing sensitivity in the round goby is sexually dimorphic at selected frequencies and is enhanced with reproductive maturity in females. [Work supported by NSERC.]

Invited Paper

2:00

2pABb4. Seasonal plasticity of saccular sensitivity in the type II sneaker-male plainfin midshipman fish (*Porichthys notatus*). Elizabeth A. Whitchurch and Joseph A. Sisneros (UW Box 351525, Seattle, WA 98195-1525, lwhitchu@uw.edu)

Acoustic communication is essential to the reproductive success of the nocturnally breeding plainfin midshipman fish (*Porichthys notatus*). During the summer breeding season, type I singing-males excavate nests in the Pacific intertidal zone and vocalize a multi-harmonic "advertisement hum" to attract potential mates. Summer reproductive females are more sensitive to the dominant frequencies

that comprise the hum than winter nonreproductive females. Type II or "sneaker" males employ alternative mating strategies that do not require nest construction or hum production but instead involve satellite and/or sneak spawning strategies. These strategies require nocturnal nest selection and localization. Because sneaker-males must rely on their auditory system to find the nests of singing males just as females must, we hypothesized that the saccular sensitivity of the reproductive sneaker-males would be most like that of reproductive females. Here we measured the evoked saccular potentials of reproductive and nonreproductive sneaker-males to tones of various sound pressure levels, and determined saccular thresholds for frequencies from 65 to 505 Hz. As predicted, thresholds measured in reproductive sneaker-males very closely matched those reported for reproductive females. Future studies will investigate the effects of gonadal steroids on hearing in sneaker-male midshipman fish. [Work supported by NIH NIDCD Grant No. 2T32DC005361-06.]

Contributed Paper

2:15

2pABb5. A new connection: Enhanced hearing ability in the New Zealand bigeye, Pempheris adspersa. C.A. Radford, P. Caiger, S. Ghazali (Dept. of Marine Sci., Univ. of Auckland, P.O. Box 349, Warkworth 0941, New Zealand), and D.M. Higgs (Univ. of Windsor, Windsor, ON N9B 3P4, Canada)

Recently the bigeye, *Pempheris adspersa*, has been found to be one of the few sound producing fish found in the temperate marine waters of New Zealand. An initial crude morphological examination of this species found a special connection, analogous to that of the well-defined laterophysic connection, between the otic capsule and the lateral line recess. The aim of the present study was to characterize the hearing ability of the bigeye using auditory evoked potentials, understand the role of this special connection, and describe the morphology of the hearing structures using microCT and MRI techniques. The novel connection consists of a robust ligament directly connecting the otic capsule to the lateral recess, with the swim bladder running directly posterior to the ligament. *Pempheris adspersa* could hear in the frequency range 100–1000 Hz, with greatest sensitivity between 100–800 Hz. Cutting the connecting ligament resulted in a 15–20 dB threshold shift, with a further 5–10 dB shift after swim bladder puncture. This connection represents a completely novel hearing specialization and shows that the lateral line is directly connected to the otic capsule, enhancing the hearing ability of this species especially at low frequencies (100–200 Hz).

Invited Papers

2:30

2pABb6. Directional and frequency saccular sensitivity of the little skate, Raja erinacea. Joseph A. Sisneros (Dept. of Psych., Univ. of Washington, Seattle, WA 98195) and Richard R. Fay (Loyola Univ. Chicago, Chicago, IL 60660)

Although a number of previous behavioral studies have demonstrated that elasmobranch fishes can detect and are attracted to low frequency sounds, few physiological studies have characterized the auditory response properties of the elasmobranch inner ear to such low frequency sounds. In this study, we examined the directional and frequency responses of the inner ear saccule in the little skate, Raja erinacea to low frequency stimuli. Evoked microphonic potentials were recorded from the middle region of the saccule while sound was generated using a shaker table designed to mimic the particle motion vector component of sound. 8 test frequencies (50, 64, 84, 100, 140, 185, 243, and 303 Hz) and 11 directions were used to characterize the displacement sensitivity, frequency response and directional response properties of the skate saccule. Saccular potentials were evoked and measured at twice the stimulus frequency *in vivo* using a wave analyzer while stimuli were generated via the shaker table system. The right and left saccules appeared to have an omnidirectional response based on initial measurements, and the frequency response of the skate saccule had lowest displacement thresholds from 100 to 185 Hz.

2:45

2pABb7. Mysterious magno. Peggy L. Edds-Walton, Solymar Rivera-Matos, and Richard R. Fay (Marine Biological Lab., Woods Hole, MA 02543)

Fish have three otolithic endorgans that may be involved in auditory processing: saccule, lagena, and utricle. One or more of those endorgans may serve a vestibular (tilt) function. In general, vestibular inputs are distributed more ventrally than auditory inputs in the medulla of teleost fishes, but there are zones of overlap with the more dorsal auditory sites. The octaval nucleus magnocellularis (Magno) receives overlapping input from all three endorgans, but Magno is not part of the ascending auditory circuit in the oyster toadfish, Opsanus tau, and the sensory role of this nucleus is unknown. Extracellular recordings were conducted in Magno and neurobiotin was injected at successful recording sites to confirm location and label local cell types. Sinusoidal particle motion stimuli (50–303 Hz) were presented at multiple stimulus levels to assess relative sensitivity. The data indicate that some cells in Magno respond to auditory frequencies at biologically relevant levels. While these results are suggestive, many attempts to record auditory responses in the divisions of Magno were negative, indicating that only a subset of cells is auditory.

3:00—3:15 Break

3:15

2pABb8. Vibration of otolithlike scatterers due to low frequency harmonic wave excitation in water. Carl R. Schilt (Bigleaf Sci. Services, P.O. Box 225, North Bonneville, WA 98639), Ted W. Cranford (San Diego State Univ., San Diego, CA 92182-3422), Petr Krysl (Univ. of California, San Diego, La Jolla, CA 92093-0085), and Anthony D. Hawkins (Loughine Ltd., Aberdeen AB12 5YT)

Otoliths may be approximated as hard objects moving in an acoustic medium in response to arriving sounds. We modeled how parameters of sound such as frequency and direction could be encoded from the motion of the otoliths. Prior numerical models predicted harmonic oscillation (rocking) of scatterers suspended in an acoustic medium when exposed to planar harmonic waves. Because of the

potential for the angular oscillation to produce additional information about the sound source, the simulated rocking motions were studied using an additional numerical model. The number of scatterers, their spatial arrangements, shapes, and the characteristics of the incident sound waves were varied. The results were analyzed and will be presented. [Work supported by the Office Naval Research.]

Contributed Paper

3:30

2pABb9. Experimental investigation of the Krysl–Cranford–Schilt model for the otolith vibration of a teleost fish. Gwendolyn V. Rodgers and Peter H. Rogers (School of Mech. Eng., Georgia Tech, Atlanta, GA 30332)

The Krysl–Cranford–Schilt model [Schilt, *et al.*, J. Acoust. Soc. Am. **127**, 1754 (2010)] for the response of a teleost otoith to sound makes some unexpected predictions, which, if true, would have significant implications for our understanding of the mechanisms of hearing in fish. It has been believed for over 60 years that the otoliths function like the inertial mass of an accelerometer, which, in the absence of attachments to the skull (or a swimbladder), would vibrate back and forth in the direction the acoustic particle velocity but with a smaller amplitude. The KCS model, however, predicts that the otolith motion is highly dependent on both the frequency and direction of the incident sound wave and involves significant rocking motion as well as translation. If this is true, then the particle-motion-vector models for directional hearing in fish, which have held sway for over 40 years are incorrect. The validity of this model is experimentally tested in a direct and simple manner. An otolith is suspended in a sphere of low-shear-modulus tissue phantom in a large acoustic test tank. The motion of the acoustically excited otolith is measured using an ultrasonic vibrometer.

Invited Papers

3:45

2pABb10. A shear wave elastography system for cetacean tissues and its potential application in fish bioacoustics. Michael D. Gray, James S. Martin, and Peter H. Rogers (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA, 30332-0405)

A system is being developed to determine the *in vivo*, low-frequency elastic properties of cetacean head soft tissues. The system consists of two ultrasound-based components: a shear wave generation source and a vibrometer that monitors the induced tissue motions. Processing of the tissue motion data yields estimates of the frequency dependent shear speed and loss as a function of tissue depth. This work builds upon developments by other researchers in the area of radiation force elastography, with the addition of novel design features intended to enable the examination of tissue volumes that are relatively thick and/or obstructed by bone. Potential applications of the technology to fish bioacoustics investigations will be discussed. [Work supported by ONR.]

4:00

2pABb11. Ontogenetic development of hearing and sound communication in catfishes. Walter Lechner and Friedrich Ladich (Dept. of Behav. Bio., Univ. of Vienna, Althanstrasse 14, 1090 Vienna, Austria)

Catfish possess a high diversity in accessory hearing structures, hearing sensitivities, and sound-generating mechanisms. Nevertheless, the ontogeny of their hearing and sound communication remains unknown. We investigated the development of Weberian ossicles and hearing sensitivity in the African bullhead catfish from postlarval stages up to adults and also examined the ontogenetic development of hearing and sound production in the yellow marbled squeaker catfish. In the smallest bullhead catfishes, the Weberian ossicles and interossicular ligaments are not fully developed. They are unable to detect sounds at low levels and high frequencies. In later stages of both species tested, hearing sensitivity increases with size at low frequencies and decreases at high frequencies. In the squeakers, the duration of stridulation sounds, sound pressure level, and pulse period increase, whereas the dominant frequency decreases with size. The most sensitive frequencies correlate with the dominant frequencies of stridulation sounds in all size groups, enabling all stages to detect communication sounds. Our studies show two different trends of changes in hearing sensitivity in catfishes: one prior to and the other after full development of the Weberian ossicles. Squeakers of all sizes are able to communicate acoustically. These results contrast with prior findings in teleosts. [Work supported by FWF.]

4:15

2pABb12. Ontogeny of early audition in the plainfin midshipman, *Porichthys notatus*. Peter W. Alderks (Dept. of Psych., Univ. of Washington, Box 351525, Seattle, WA 98195, pwa2@uw.edu) and Joseph A. Sisneros (Univ. of Washington, Seattle, WA 98195)

Early development is a time of great organizational activity. As sensory systems become functionally active, they often undergo a period of ontogenetic plasticity. The early ontogeny of the auditory system of the plainfin midshipman is of particular interest. During early ontogeny, the fertilized eggs of the midshipman are attached to the underside of rocks guarded by nesting (type I) males. These nesting males often guard multiple clutches of eggs while actively calling to attract additional mates. The males' advertisement call is high intensity (approximately 145 dB *re* 1 μPa at the source) and long duration (minutes to hours). During nest incubation, the auditory system of the developing embryos becomes active. We previously showed that when juveniles leave the nest, the auditory sensitivity of their saccule closely resembles that of nonreproductive adults. Here, we investigate auditory sensitivity from the time the auditory system becomes functional until the juvenile midshipman is free swimming using an acoustic startlelike response to broadband clicks. We characterize the startlelike response in embryos and juveniles and use this behavior as a measure to determine when the midshipman auditory system becomes functional. We also present physiology data on evoked saccular potentials during this developmental window.

4:30

2pABb13. Saccular-specific, seasonal differences in hair cell density in a vocal fish with seasonal auditory plasticity. Allison B. Coffin (Dept. of Otolaryngol.-HNS, Univ. of Washington, Seattle, WA 98195), Robert A. Mohr, and Joseph A. Sisneros (Univ. of Washington, Seattle, WA 98195)

Plainfin midshipman fish (*Porichthys notatus*) rely on acoustic communication for courtship. Courting males build nests in rocky intertidal areas and emit low-frequency hums to attract reproductive females. There is surprising physiological plasticity in the female midshipman's auditory system, with reproductive (summer) females exhibiting greater sensitivity to higher frequencies than

non-reproductive (winter) females. Recent work demonstrates that this physiological plasticity occurs at the level of the saccule, which is the primary hearing organ in this species. This research examines potential morphological correlates of this physiological plasticity. Summer females have greater saccular hair bundle density than winter females and there is a corresponding increase in net cell addition as determined by quantification of dividing and dying cells and by counts of immature hair bundles. No seasonal differences are seen in the other inner ear end organs. Saccular hair cell number is not correlated with fish size, suggesting that the hair cell increase is not due to differences in fish size between sampling seasons. These data suggest that the hair cell increase in summer females may contribute to the greater auditory sensitivity seen in reproductive females. Future studies will examine morphological and molecular differences between saccular hair cells in winter and summer females.

4:45

2pABb14. Monoamine innervation of vocal and auditory circuits in the plainfin midshipman fish. Paul M. Forlano, Lilja Nielsen, and Miky Timothy (Dept. of Biology and Aquatic Res. and Environ. Assessment Ctr., Brooklyn College, CUNY, 2900 Bedford Ave., Brooklyn, NY 11210, pforlano@brooklyn.cuny.edu)

Monoamines, which include catecholamines (e.g., dopamine and noradrenaline) and serotonin, are important regulators of reward, motivation and reproductive-related behaviors across vertebrates. Studies in tetrapods have demonstrated that monoamines can also serve a modulatory function for vocalization and audition. Antibodies directed against serotonin (5-HT) and tyrosine hydroxylase (TH) (the rate limiting enzyme in catecholamine synthesis) were employed to investigate monoamine innervation of known central vocal and auditory nuclei and the sensory epithelium of the saccule, the main endorgan of hearing, in the plainfin midshipman fish, *Porichthys notatus*. Both 5-HT and TH-immunoreactive (-ir) terminals are abundant in hindbrain (dorsal descending octaval nucleus), midbrain (torus semicircularis), thalamic (central posterior nucleus), and ventral telencephalic auditory nuclei. In addition, very robust TH-ir is found in the octavolateralis efferent nucleus, and prominent TH-ir projections travel along the eighth nerve and terminate within the hair cell layer of the saccular epithelium. Aside from the midbrain periaqueductal gray, TH-ir is much more pronounced than 5-HT-ir throughout the descending vocal motor system, particularly in the tuberal hypothalamus and the hindbrain-spinal vocal motor nucleus, which directly innervates vocal musculature on the swimbladder. These data provide anatomical evidence that monoamines are modulators of vocal and auditory-driven behaviors in fishes.

5:00

2pABb15. Tracing tonotopy in teleosts. Michael E. Smith (Dept. of Biology, Western Kentucky Univ., 1906 College Heights Blvd., Bowling Green, KY 42101, michael.smith1@wku.edu)

It is not known whether the auditory hair cells of fishes possess a tonotopic organization in the saccule. To investigate this question, we exposed groups of six goldfish (Carassius auratus) to one of four tones (100, 800, 2000, and 4000 Hz) at 176 dB *re* 1 μPa for 48 h. The saccules of each fish were dissected and labeled with phalloidin in order to visualize hair cell bundles. The hair cell bundles were counted at 19 specific points in each saccule to determine the extent and location of hair cell damage. In addition to quantification of anatomical injury, hearing tests (using auditory evoked potentials) were performed on each fish immediately following sound exposure. The location of hair cell loss varied along the length of the saccule in a graded manner with the frequency of sound exposure, with lower and higher frequencies damaging the more caudal and rostral regions of the saccule, respectively. Similarly, fish exposed to lower frequency tones exhibited greater threshold shifts at lower frequencies, while high-frequency tone exposure led to hearing loss at higher frequencies. These data suggest that the frequency discrimination ability of goldfish is at least partially driven by peripheral tonotopy in the saccule.

5:15

2pABb16. Localization of monopole and dipole sound sources by midshipman fish (Porichthys notatus). David G. Zeddies (Marine Acoust., Inc., 4100 Fairfax Dr., Ste. 730, Arlington, VA 22203), Richard R. Fay (Loyola Univ. Chicago, Chicago, IL 60626), Peter W. Alderks (Univ. of Washington, Seattle, WA 98195), Michael D. Gray (Georgia Inst. of Technol., Atlanta, GA 30332), Allison B. Coffin, Ashwin Bandiwad, Rober A. Mohr, Andrew D. Brown (Univ. of Washington, Seattle, WA 98195), Peter Rogers (Georgia Inst. of Technol., Atlanta, GA 30332), and Joseph A. Sisneros (Univ. of Washington, Seattle, WA 98195)

A series of experiments was undertaken to investigate methods of sound source localization by fish. In these experiments, positive phonotaxic responses of gravid female plainfin midshipman fish (*Porichthys notatus*) to low-frequency, playback tones (80–90 Hz) were studied as they approached sound sources. The sound fields for simple (monopole) and relatively complex (dipole) sources within the behavioral arena were measured and characterized in terms of pressure and particle motion. Results indicate that female midshipman fish are able to locate sound sources in the near field using acoustic cues alone, and that they used the particle motion vectors to locate the source in both the monopole and dipole sound fields. It was also found that neither the lateral line nor the swim bladder was necessary for localization behavior, and that the fish were able to solve the 180 deg ambiguity inherent in the particle motion vectors. [Work was supported by the National Science Foundation.]

Session 2pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Martin Siderius, Chair
Portland State Univ., Electrical and Computer Engineering Dept., 1900 S.W. Fourth Ave., Portland, OR 97207

Chair's Introduction—4:10

Invited Paper

4:15

2pAO1. Skipping stones along the boundary between marine mammal acoustics and acoustical oceanography. Aaron M. Thode (Marine Physical Lab., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238)

Reductions in the power consumption and cost of underwater acoustic data acquisition systems have created profound changes in how acoustic oceanography, particularly bioacoustics, is conducted. Swarms of autonomous bioacoustic recorders can now be deployed across regions formerly too isolated to be reliably monitored. The consequent flood of acoustic data has made automated detection and classification systems an essential tool in current research. This presentation reviews three recent efforts that exemplify these trends. The first focuses on gray whales in San Ignacio Lagoon, Baja, CA, where interesting relationships are being uncovered between visual census counts and background calling rates. The second concerns sperm whale depredation of longline fishing gear in Southeast Alaska, where acoustic tools have helped reveal how animals locate fishing activity, and how depredation success might be remotely monitored. The final effort involves large-scale acoustic research conducted in the Arctic Ocean by Shell Oil to monitor the fall migration of bowhead whales through regions subject to seismic airgun exploration activity. The development of large-scale automated processing methods was required to process millions of bowhead whale sounds collected on multichannel sensors and vertical arrays. All three projects illustrate the intimate connections between marine mammal acoustic behavior and the background noise environment.

TUESDAY AFTERNOON, 24 MAY 2011 GRAND BALLROOM A, 1:00 TO 5:00 P.M.

Session 2pBA

Biomedical Acoustics and Physical Acoustics: Tissue Erosion Techniques in Therapeutic Ultrasound

Vera A. Khokhlova, Cochair
Moscow State Univ., Acoustics Dept., 119992, Moscow, Russia

Lawrence A. Crum, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Chair's Introduction—1:00

Invited Papers

1:05

2pBA1. The quality of histotripsy produced tissue lesions can be remarkably insensitive to strong acoustic aberrations: Role of nonlinear phenomena. Charles Cain, Yohan Kim, Tzu-Yin Wang, and Zhen Xu (Biomedical Eng., Univ. of Michigan, 2202 Bonisteel, Ann Arbor, MI 48109, cain@umich.edu)

Cavitation induced by short high intensity pulses can produce confined lesions with remarkably narrow transition zones (often bisecting individual cells). This process (histotripsy) is a nonlinear phenomenon wherein energetic bubble clouds are produced only when the incident ultrasound beam is above the cavitation threshold. It has been shown that acoustic aberrations in the path of an acoustic beam tend to rob energy from the main lobe raising surrounding side-lobes, sometimes significantly. However, in many cases, the shape of the main lobe, while diminished in amplitude, retains its original confined Gaussian-shaped cross-section. If the transducer-driving electronics has sufficient head-room, the main lobe can be boosted to preaberration above-threshold levels while maintaining the original main lobe profile. If the boosted main lobe amplitude is above the side-lobe levels, the lesion producing bubble clouds often are

indistinguishable from the nonaberration case thus producing a clean confined lesion with little collateral damage even in the presence of strong aberrations. Confirmation of this strategy with porcine and polymer rib aberration phantoms is presented with a study of lesion shape as well as collateral damage with and without aberrations. Boosted intensities can be kept below thermally significant levels by adjusting pulse repetition frequency and pulse width.

1:25

2pBA2. A method of mechanical emulsification in a bulk tissue using shock wave heating and millisecond boiling. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Michael S. Canney (INSERM U556, Lyon, France), Michael R. Bailey, Joo Ha Hwang, Tatiana D. Khokhlova, Wayne Kreider, Yak-Nam Wang, Julianna C. Simon (Univ. of Washington, Seattle, WA), Yufeng Zhou (Nanyang Technolog. Univ., Singapore), Oleg A. Sapozhnikov, and Lawrence A. Crum (Univ. of Washington, Seattle, WA)

Recent studies in high intensity focused ultrasound (HIFU) have shown significant interest in generating purely mechanical damage of tissue without thermal coagulation. Here, an approach using millisecond bursts of ultrasound shock waves and repeated localized boiling is presented. In HIFU fields, nonlinear propagation effects lead to formation of shocks only in a small focal region. Significant enhancement of heating due to absorption at the shocks leads to boiling temperatures in tissue in milliseconds as calculated based on weak shock theory. The heated and potentially necrotized region of tissue is small compared to the volume occupied by the mm-sized boiling bubble it creates. If the HIFU pulse is only slightly longer than the time-to-boil, thermal injury is negligible compared to the mechanical injury caused by the exploding boiling bubble and its further interaction with shocks. Experiments performed in transparent gels and various *ex vivo* and *in vivo* tissues have confirmed the effectiveness of this emulsification method. In addition, since mm-sized boiling bubbles are highly echogenic, tissue emulsification can be easily monitored in real-time using *B*-mode ultrasound imaging. [Work supported by NIH EB007643, RFBR 09-02-01530, and NSBRI through NASA NCC 9-58].

1:45

2pBA3. The application of kilohertz-frequency ultrasound energy in performing advanced clinical procedures. Inder Raj S. Makin (Arizona School of Dentistry & Oral Health, and School of Osteopathic Medicine of Arizona, 5850 E. Still Circle, Mesa, AZ 85206, imakin@ieee.org)

The use of ultrasound energy for therapeutic applications is an ever-increasing option for the medical professional in managing clinical disease states. Ultrasound therapies in medicine are generally delineated based on frequency regimes, namely, kilohertz and megahertz. Based on the relatively long wavelengths in tissue (several millimeters), and the small size of the radiator source (1–4 mm), with respect to the wavelength, kilohertz-frequency medical devices generally rely on biophysical effects such as rapid frictional heating, cavitation, streaming, and mechanical disruption, in very close proximity to the ultrasound sources. These devices are therefore inserted intra-corporeally to achieve tissue cutting, hemostasis, thrombolysis, or lipid-based tissue emulsification. The bioeffects resulting from the use of kilohertz-frequency based devices are much more controlled compared to their counterparts such as radiofrequency, lasers, and microwave devices. This talk will provide an overview of the kilohertz-frequency techniques used in the areas of vascular (angioplasty) advanced surgical (ultrasonic scalpel), as well as cosmetic (lipolysis) procedures. Numerical simulation, experimental, as well as some clinical examples will be presented to illustrate various biophysical mechanisms that form the basis of different systems.

2:05

2pBA4. Focused destruction of renal tissue by a narrow focal width lithotripter. James A. McAteer, Andrew P. Evan, Bret A. Connors, and Philip M. Blomgren (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, 635 Barnhill Dr., Indianapolis, IN 46202)

Although lithotripters can differ substantially in acoustic output, renal response to SWs has been thoroughly characterized for only one type of lithotripter (moderate-pressure ~40 MPa, intermediate-focal-width 8–9 mm, Dornier-HM3). Therefore, we assessed injury using a high-pressure (~85 MPa), narrow-focal-width (3–4 mm) device (Storz-SLX). SWs were administered to the lower renal pole of pigs (15 kg) (PL-9, 2000 or 4000 SWs at 120 SWs/min; or 2000SWs at 60 SW/min) followed by serial sectioning to quantify lesion. Tissue damage with the SLX showed an abrupt transition between injured and unaffected tissue. Core of lesion was typically devoid of recognizable structure, with renal tubules, glomeruli, vessels essentially eliminated. The lesion extended from cortex to medulla, and in some kidneys homogenized tissue could be tracked across full thickness of the kidney. In comparison, the lesion using HM3 was not as severe with only focal spots of hemorrhage and no areas of complete tissue destruction. Doubling the SW-dose with SLX did not increase lesion volume as occurred with HM3. Also, slowing the SW-rate of the SLX to 60 SW/min did not have a protective effect as observed with other lithotripters. These findings suggest that lithotripters having different acoustic characteristics are capable of producing different forms of tissue damage. [Work supported by NIH-DK43881.]

2:25

2pBA5. Models for cavitation-induced displacement, stress, and strain in tissue. Mark F. Hamilton, Todd A. Hay, Yurii A. Ilinskii, and Evgenia A. Zabolotskaya (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Cavitation activity near tissue interfaces may cause the tissue to deform, inducing stresses and strains of sufficient magnitude to cause damage. Understanding the interaction between bubbles and the surrounding tissue is therefore critical in therapeutic ultrasound. Since interacting bubble clusters are often responsible for bio-effects, both bubble-bubble and bubble-tissue interactions should be considered. In this presentation, simulation results from models describing the dynamics of single bubbles and bubble pairs immersed

in a viscous liquid near tissue interfaces will be presented for two geometries. The first geometry is that of a bubble pair positioned between parallel tissue layers, and the second is for a bubble in a viscoelastic cylindrical tube. Tissue displacements caused by the motion of the bubbles may be estimated via models derived using Green's function and Lagrangian mechanics approaches. The Green's function approach further permits estimation of the time-varying strain and stress fields in the tissue, including effects of viscoelasticity. Aspherical bubble deformation and the onset of jetting due to bubble-bubble or bubble-tissue interaction is modeled using the Lagrangian mechanics approach. Simulations illustrating the dynamics of the bubbles and tissue for clinically relevant parameters will be presented. [Work supported by NIH Grant Nos. DK070618 and EB011603.]

2:45—3:00 Break

Contributed Papers

3:00

2pBA6. Mechanisms of action for ultrasound surgical instruments. Mark E. Schafer (Sound Surgical Technologies, LLC, 357 S. McCaslin Blvd., Louisville, CO 80027)

Ultrasound devices operating at kilohertz frequencies have been used in surgery for over 40 years. Examples range from cataract removal (phacoemulsification) to adipocyte extraction (ultrasound assisted liposuction). These devices make direct contact to erode tissue and work via a number of mechanisms including cavitation (both inertial and stable) and acoustic (micro)streaming. This talk will describe recent research into various factors which affect the mechanisms of tissue erosion. While all these devices operate in the frequency range of 23 to about 43 kHz, the exact combination of frequency, vibration amplitude and mode, and tip configuration creates a range of acoustic source strengths and thus acoustic outputs. Some devices work principally through inertial cavitation action, and experiments to demonstrate this will be reviewed. Other systems tend to favor stable cavitation, permitting, for example, the extraction of viable fat cells which can be re-implanted or harvested for other purposes, such as adipose-derived regenerative (stem) cells. Still other devices rely mostly on acoustic microstreaming and physical macromotion to dissolve blood clots. The range of device types and mechanisms creates difficulties in making direct comparisons between systems, even between those with similar overall design and clinical purpose. Potential solutions to this issue will also be presented.

3:15

2pBA7. Histological and biochemical analysis of emulsified lesions in tissue induced by high intensity focused ultrasound. Yak-Nam Wang, Tatiana D. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Mike S. Canney (INSERM, U556, 69424 Lyon, France), Vera A. Khokhlova, Lawrence A. Crum, and Mike R. Bailey (Univ. of Washington, Seattle, WA 98105-6698)

As recently shown, shock wave heating and millisecond boiling can be used to obtain mechanical emulsification of tissue with or without evident thermal damage, which can be controlled by varying the parameters of the high intensity focused ultrasound exposure. The goal of this work was to examine these bioeffects using histological and biochemical analysis. Lesions were created in *ex vivo* bovine heart and liver using a 2-MHz transducer and pulsing scheme with 71 MPa *in situ* shock amplitude, 0.01 duty factor, and 5–500 ms pulse duration. Mechanical tissue damage and viability of cells in the lesions were evaluated histologically using conventional staining techniques (H&E and NADH-diaphorase). Thermal effects were quantified by measuring denaturation of salt soluble proteins in the treated area and confirmed by histology. By visual observation, the liquefied lesions obtained with shorter pulses ($<$ 15 ms) did not show any thermal damage that correlated well with the results of both histology and protein analysis. Increasing the pulse duration resulted in an increase in thermal damage; both protein analysis and NADH-diaphorase staining showed denaturation that was visually observed as whitening of the lesion content. [Work supported by NSBRI through NASA NCC 9-58, NIH EB007643 and DK007742.]

3:30

2pBA8. *In vivo* tissue emulsification using millisecond boiling induced by high intensity focused ultrasound. Tatiana D. Khokhlova, Julianna C. Simon, Yak-Nam Wang, Vera A. Khokhlova, Marla Paun, Frank L. Starr, Peter J. Kaczkowski, Lawrence A. Crum, Joo Ha Hwang, and Michael R. Bailey (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105)

Shock-wave heating and millisecond boiling in high intensity focused ultrasound fields have been shown to result in mechanical emulsification of *ex-vivo* tissue. In this work, the same *in situ* exposures were applied *in vivo* in pig liver and in mice bearing 5–7 mm subcutaneous tumors (B16 melanoma) on the hind limb. Lesions were produced using a 2-MHz annular array in the case of pig liver (shock amplitudes up to 98 MPa) and a 3.4-MHz single-element transducer in the case of mouse tumors (shock amplitude of 67 MPa). The parameters of the pulsing protocol (1–500 ms pulse durations and 0.01–0.1 duty factor) were varied depending on the extent of desired thermal effect. All exposures were monitored using B-mode ultrasound. Mechanical and thermal tissue damage in the lesions was evaluated histologically using H&E and NADH-diphorase staining. The size and shape of emulsified lesions obtained *in-vivo* agreed well with those obtained in *ex-vivo* tissue samples using the same exposure parameters. The lesions were successfully produced both in bulk liver tissue at depths of 1–2 cm and in superficial tumors at depths less than 1 mm without damaging the skin. [Work supported by NIH (DK070618, EB007643, and DK007742) and NSBRI through NASA NCC 9-58.]

3:45

2pBA9. Models for coupled bubble dynamics between parallel elastic layers. Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Recent photographs of microbubbles driven by high-amplitude pulses in *ex vivo* blood vessels have provided quantitative measurements of coupled bubble dynamics near tissue [Chen *et al.*, J. Acoust. Soc. Am. **126**, 2175 (2009)]. There is also interest in exploiting bubble-bubble interaction to aid sonoporation of an adjacent cell membrane [Sankin *et al.*, Phys. Rev. Lett. **105**, 078101 (2010)]. These observations have motivated the development of models describing the dynamics of coupled bubbles near compliant interfaces. Two models describing the dynamics of a bubble pair positioned between parallel tissue layers immersed in a viscous liquid will be presented. One model is based on Lagrangian mechanics and incorporates bubble shape deformation accounting for the onset of jetting, but is limited to an orientation in which the axis connecting the bubbles is perpendicular to the tissue layers. The second model, which makes use of Green's function, accounts for fluid compressibility and tissue viscoelasticity, but is limited to spherical bubbles. Expressions for the displacement, strain, and stress fields in the tissue are derived. Simulations will be presented to illustrate the influence of bubble-tissue and bubble-bubble interaction, for a variety of clinically relevant parameters. [Work supported by NIH Grant Nos. DK070618 and EB011603.]

4:00

2pBA10. Miniature acoustic fountain mechanism for tissue emulsification during millisecond boiling in high intensity focused ultrasound fields. Julianna C. Simon (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Oleg A. Sapozhnikov, Vera A. Khokhlova (Univ. of Washington, Seattle, WA and Moscow State Univ., Moscow 119991, Russia), Tatiana D. Khokhlova, Michael R. Bailey, and Lawrence A. Crum (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Feasibility of soft tissue emulsification using shock wave heating and millisecond boiling induced by high intensity focused ultrasound was demonstrated recently. However, the mechanism by which the bubbles emulsify tissue is not well understood. High-speed photography of such exposures in transparent gel phantoms shows a milimeter-sized boiling bubble, and histological analysis in tissue samples reveals sub-micron-sized fragments. Here, a novel mechanism of tissue emulsification by the formation of a miniature acoustic fountain within the boiling bubble is tested experimentally using a 2 MHz transducer generating up to 70 MPa positive and 15 MPa negative peak pressures at the focus. The focus was positioned at or 1–2 mm off the plane interface between air and various materials including degassed water, transparent gel, thin sliced muscle tissue phantom, and *ex-vivo* tissue. Pulsing schemes with duty factors 0.001–0.1, and pulse durations 0.05–500 ms were used. Violent removal of micron-sized fragments and substantial displacement of the phantom surface were observed through high-speed filming. At the end of each exposure, the resulting erosion of the phantom surface and subsurface area was photographed and related to the exposure parameters. Work supported by NIH (DK43881, DK070618, EB007643, and DK007742) and NSBRI through NASA NCC 9-58.

4:15

2pBA11. Ultrasonic atomization on the tissue-bubble interface as a possible mechanism of tissue erosion in histotripsy. Oleg A. Sapozhnikov, Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119991, Russia, olegs@apl.washington.edu), and Michael R. Bailey (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

When an intense ultrasound beam is directed at a free surface of a liquid, an acoustic fountain is produced that is typically accompanied by ejection of tiny droplets, i.e., liquid atomization. This phenomenon is usually attributed to instability of cavitation-produced capillary waves on the surface. In addition to capillary effects, a process called spallation may also contribute. Although the acoustic fountain is typically observed at a flat liquid surface, nothing prohibits the atomization from occurring at a curved surface. This brings about the possibility to create an acoustic fountain and droplet emission at the surface of a gas cavity in liquid or, similarly, in the bulk of soft biological tissue. The appropriate condition occurs when high-intensity ultrasound is focused in tissue and creates large (0.1–1 mm in diameter) bubbles due to acoustic cavitation or rapid boiling. To test this hypothesis, acoustic pressure distribution and the corresponding radiation force on the empty spherical cavity were calculated using finite difference modeling and spherical harmonic expansion. It is shown that in histotripsy regimes appropriate conditions appear for the atomization, which may be considered as a possible mechanism of tissue erosion. [Work supported by NIH EB007643, RFBR, and NSBRI through NASA NCC 9-58.]

4:30

2pBA12. Ultrasound-induced fragmentation of connective tissue guided by passive acoustic mapping. M. B. Molinari, M. Arora (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7DQ, United Kingdom), J. P. G. Urban (Univ. of Oxford, Oxford OX1 3PT, United Kingdom), and C. C. Coussios (Univ. of Oxford, Oxford OX3 7DQ, United Kingdom)

To date, studies on the mechanical bio-effects of acoustic cavitation have been confined to soft tissues such as the liver or prostate. In the present work, acoustic cavitation caused by high intensity focused ultrasound (HIFU) is used to disrupt the highly collagenous, acellular connective tissue of the intervertebral disk (IVD). The IVD is comprised of three regions: a highly hydrated, amorphous nucleus pulposus (NP) enclosed circumferentially by the lamellar structure of the annulus fibrosus, and capped by cartilaginous endplates. In experiments using *ex vivo* bovine coccygeal disks as a model, two confocally-aligned HIFU transducers were positioned with their focus inside the disk, and driven at 500 kHz. Cavitation activity was localized and monitored using both a 3.5 MHz PCD and a 2-D passive mapping array and was found to be confined to the NP. Treatment parameters were optimized to minimize treatment time and thermal damage to adjacent tissue. Disruption of the tissue after treatment was determined visually and suggests that it should be possible to remove disrupted tissue through a small gauge needle, thus enabling several proposed treatments for disk degeneration.

4:45

2pBA13. Modeling single-bubble dynamics in histotripsy. Chengyun Hua and Eric Johnsen (Dept. of Mech. Eng., Univ. of Michigan, 2016 Walter E. Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, cyhua@umich.edu)

In several therapeutic ultrasound applications such as histotripsy, cavitation bubbles are generated in tissue by high-intensity ultrasound pulses. In order to understand bubble dynamics in human tissue related to histotripsy, the model of Yang and Church (2005) for spherical bubble dynamics in a compressible and viscoelastic medium is extended to include variations in the liquid density and non-spherical perturbations in viscoelastic media based on the stability analysis by Prosperetti (1977). Both continuous symmetric pulses and asymmetric pulses representative of histotripsy are considered. The bubble exhibits a strongly nonlinear response to the asymmetric pulses but recovers rapidly thereafter. Density variations in the liquid lead to a larger amplitude response of the bubble radius after the first cycle. The pressure and strain in the surrounding material are determined both analytically and numerically, thus providing estimates for potential damage away from the bubble. The violent bubble collapse leads to high local pressures at the bubble location and significant strains are achieved away from the bubble during the expansion. The effects of nonlinearity, amplitude, and frequency of the asymmetric pulses will be considered and more sophisticated viscoelastic models will be applied.

Session 2pNS

Noise and Physical Acoustics: Acoustical Modeling for Complex Outdoor Environments

Siu-Kit Lau, Cochair
Univ. of Nebraska, Lincoln, Architectural Engineering Program, 1110 S. 67 St., Omaha, NE 68182-0681

Kai Ming Li, Cochair
Purdue Univ., School of Mechanical Engineering, 140 Martin Jischke Dr., West Lafayette, IN 47906

Invited Papers

1:00

2pNS1. Time-domain simulations of outdoor sound propagation in presence of a complex topography. Philippe Blanc-Benon, Didier Dragna (LMFA, UMR CNRS 5509, Ecole Centrale de Lyon, Universit de Lyon, 36 Ave., Guy de Collongue, F-69134 Ecully Cedex, France), and Franck Poisson (Nationale des Chemins de Fer, 75379 Paris Cedex 08, France)

For transportation noise applications, the description of the environment, as well as the description of acoustic sources, is complex; and finite-difference time-domain methods have proved their capability to take into account both atmospheric effects and topography. Recently, a time-domain boundary condition has been proposed [Cotté *et al.*, AIAA J. **47**, 23912403 (2009)] and implemented in a solver using methods developed in the computational aeroacoustics community [Bogey and Bailly, J. Comput. Phys. **194**, 194214 (2004)]. First, propagation of an initial pulse over a distance of 500 m in a two-dimensional geometry is considered in a frequency band up to 1200 Hz. Ground effects are discussed depending on the impedance of the ground, and results are compared with available analytical data. Surface waves, which propagate close to the ground, are exhibited. Then, acoustic scattering by an impedance cylinder in a two-dimensional geometry is considered as a test case to validate the boundary conditions in the presence of an acoustic shadow zone. Comparisons are realized with conformal mapping numerical predictions. At last, a three-dimensional geometry of a railway track is considered to highlight the effects of topography on measurements of railway noise. [This work was performed using HPC resources from GENCI-IDRIS Grant No. 2010-022203.]

1:20

2pNS2. Use of meteorological classes for simplifying forecasts of outdoor sound propagation. Michael J. White and Michelle E. Swearingen (US Army Engineer Res. and Development Ctr., P.O. Box 9005, Champaign, IL 61826)

The detailed and variable atmospheric condition is mostly undersampled but certainly important for making accurate predictions of sound propagation. While prediction accuracy suffers due to incomplete knowledge of the current atmosphere, the benefit of coarse classification has not been fully explored. We use measured wind and temperature values from a 30 m tower to infer the nearest stability class and wind speed and to develop statistics for measurements of tones propagated from the tower to 1.6 km distance. The measured statistics for the classes are reinforced with computational modeling.

1:40

2pNS3. Investigations of environmental and terrain effects on the propagation of freeway noise. Nick C. Ovenden (Dept. of Mathematics, Univ. College London, Gower St., London WC1E 6BT, United Kingdom, nicko@math.ucl.ac.uk), Stephen R. Shaffer (Arizona State Univ., Tempe AZ 85281), and Harindra J. S. Fernando (Notre Dame, IN 46556-5637)

Noise caused by freeway traffic is a serious environmental concern in major urban areas. This study examines the impact of meteorological conditions and terrain features on the propagation of traffic noise from urban freeways. Meteorological and sound-level measurements were taken at freeway sites in Phoenix, AZ, and an analytical Green's function solution was used to produce a sound field in the neighborhood of the freeway that replicates the sound levels measured. A parabolic equation model for sound propagation is then coupled to this Green's function solution of the source field to compute the refracted sound field out to a range of half a mile from the freeway to predict the noise exposure in residential areas. The model demonstrates that atmospheric effects can raise sound levels by 10–20 dB at significant distances away from the highway, causing violations of acceptable limits imposed by the Federal Highway Administration in residential areas that are normally in compliance. A generalized terrain parabolic equation model is also extended to account for coupled two-way sound propagation, enabling the study of backscatter from certain terrain features. Terrain examples incorporating noise barriers and freeway canyons will be presented. [Work supported by Arizona Dept. of Transportation.]

2:00

2pNS4. Sound field of a fast moving source in a horizontally stratified medium above an impedance plane. Bao Tong and Kai Ming Li (Dept. of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031)

A continuous source motion model is developed for computing the sound fields of a uniformly moving monopole source in a horizontally stratified atmosphere. The numerical model is based on the Lorentz transformation and the one-dimensional fast field formulation. The global matrix method and a bounded Green's function solution expressed in the wavenumber domain is used in the

2p TUE. PM

numerical implementation. The solution is then expressed in the Lorentz frame. A transformation is performed to map the spatial Lorentz frame quantities into a reception time history in the stationary frame. Numerical results for an elevated source in a downward refracting atmosphere demonstrates the importance of a continuous source motion model especially at high Mach numbers in the presence of sound speed gradients where the pseudo-stationary source approximation becomes questionable. Coupled with the convective effects of continuous source motion and a frequency dependent ground impedance model, the resulting sound fields are significantly different from those predicted due to a pseudo-stationary source. Fluctuations on the order of 20 dB are predicted in the interference pattern in the far-field. A shift of the interference onset time to lower values is predicted when the source Mach number and/or sound speed gradient is increased.

Contributed Papers

2:20

2pNS5. A Markovian approach to the modeling of sound propagation in complex urban environments. David Oldham (School of Architecture, Univ. of Liverpool, Liverpool L69 3BX, United Kingdom.)

[The use of the radiosity method as a means of modeling sound propagation in complex urban environments has been extensively employed by a number of researchers over the past decade. A basic assumption for this approach is that all reflections at building facades are diffuse and hence sound is scattered rather than reflected specularly. The approach employed in the work described in this paper uses the same basic assumption of diffuse surface reflections as the radiosity method but combines this with a Markovian approach in which the sound scattered from small area elements on the facades to similar elements on another facade is treated in terms of probability. It is thus possible to construct a probability matrix describing the entire propagation process for a specified urban geometry. It can be shown that significant reductions in computational requirements can be achieved by considerations of factors such as reciprocity. The method is applied to an investigation of sound propagation for a variety of different urban forms.

2:35

2pNS6. Improving the efficiency of mean sound level calculations through importance sampling. D. Keith Wilson (U.S. Army Engineer Res. Dev. Ctr., 72 Lyme Rd., Hanover, NH 03755, d.keith.wilson @usace.army.mil) and Chris L. Pettit (U.S. Naval Acad., Annapolis, MD 21402)

Outdoor sound levels vary in space and time due to randomness in acoustic source emissions and environmental effects on wave propagation. Sound levels may also appropriately be viewed as random due to uncertainties in the modeling process. To assess the impact of the randomness on sound level predictions, Monte Carlo sampling of the model input space can be used; that is, equally likely sets of model inputs are generated, to which a sound propagation model is applied repeatedly. To improve the efficiency of the calculations, it is highly desirable to sample primarily the acoustical frequency bands and environmental conditions (such as downwind propagation and hard ground surfaces) contributing most strongly to the sound level. Importance sampling techniques, which adaptively focus sampling effort on the most important parts of the input parameter space, can be used for this purpose. This paper explores application of importance sampling to prediction of sound level statistics. The technique is found to be particularly valuable in limiting the calculation effort at high frequencies, where most computational methods are highly inefficient and propagation effects are most unpredictable.

2:50

2pNS7. Parabolic equation approach for modeling atmospheric noise propagation. Joseph F. Lingevitch and Laurie T. Fialkowski (Naval Res. Lab., Washington, DC 20375)

A wide-angle parabolic equation (PE) model for propagating atmospheric noise from a directional source is developed. The directivity pattern of a compact directional source is modeled with a vertical distribution of phased sources and the effects of atmospheric attenuation, sound speed and wind profiles, and range-dependent terrain are incorporated into the marching algorithm. The PE approach includes the effects of diffraction by terrain features using a piecewise linear terrain-flattening transformation to simplify the implementation of the impedance boundary condition at the ground. Comparisons with image, ray-based, and finite element solutions are shown to validate the model. [Work supported by the ONR.].

3:05—3:15 Break

3:15

2pNS8. Rocket noise predicton models: An update. Louis C. Sutherland (LCS Acoustics, 5701 Crestridge Rd., Apt. 109, Rancho Palos Verdes, CA 90275)

This paper summarizes revised or improved elements of rocket engine noise prediction models including overall sound power for rocket, noise source locations in undeflected/deflected rocket exhausts, sound power and/or pressure spectrum, and directivity for the rocket noise sources at, and after, launch with emphasis on locations on the launch tower. Finally, influence of the flow deflector and water injection on the rocket noise levels is considered. The data presented are from a number of previously published reports by the author, and others, including Eldred, Crocker, and Potter. The sound power of the rocket is predicted by a Lighthill theory-consistent model as proportional to a flow-dependent multiplier corresponding to the ratio of flow mechanical power at the supersonic core tip to mechanical power at nozzle exit. Data from 27 measurements of sound power of jets and rockets covering a range from 135 to 200 dB *re* 1 PW show a prediction accuracy of $+/-2.8$ dB. The model is a more robust approach to rocket sound prediction than previous methods such as assuming 1% acoustic efficiency. The paper also presents graphical models for the other rocket noise parameters defined above.

3:30

2pNS9. Spectrum of measured infrasonic emissions from clear air turbulence. Allan J. Zuckerwar, George R. Weistroffer (Analytical Services Materials, 107 Research Dr., Hampton, VA 23666), and Qamar A. Shams (NASA Langley Res. Ctr., Hampton, VA 23681)

An array of three infrasonic microphones (0.2–20 Hz), operating continuously in the field at NASA Langley Research Center, on several occasions received a class of signals interpreted as infrasonic emissions from clear air turbulence. The presence and location of the turbulence were confirmed by pilot reports (PIREPS), and the direction of emitted signals toward the array was determined by slowness mapping. The coherence of the signals among the three microphone pairs in the array was close to unity. The amplitude spectrum of the received signals was found to fit a power law having an exponent of $-7/2$, which disagrees with the exponent of $-7/4$ of Meecham and Ford [J. Acoust. Soc. Am. **30**, 318–322 (1958)], based on turbulence self-noise and with the exponent of -1 of Meecham [J. Acoust. Soc. Am. **33**, 149–155 (1971)], based on mean shear fluctuations. Thus the above models do not account for the observed spectrum. Two case histories are described in detail.

3:45

2pNS10. Response of infrasonic microphone field array to a controlled source. Qamar A Shams (NASA Langley Res. Ctr., M.S. 238, Hampton, VA 23681), Allan J. Zuckerwar, and George R. Weistroffer (Analytical Services and Materials, 107 Research Dr., Hampton, VA 23666)

A field test on a three-microphone array at NASA Langley Research Center was conducted using a mobile controlled infrasonic source. A Helmholtz resonator, used to provide a simulated point source for infrasonic propagation studies, had an output SPL of 99 dB (at 1 m) at its resonance frequency of 9.45 Hz. The three-microphone array was arranged as an equilateral triangle with microphone spacing of 30.48 m (100 ft) and at a distance of more than 85.3 m (280 ft) from the source. The signal level was 40

dB above the background noise in a 1-Hz band. Measurements of the acoustical response for each of the array microphones were recorded, and the received signal was measured at the nearest microphone to be 60 dB (6 dB per doubling of distance).

4:00

2pNS11. Determining the importance of building features in sonic boom simulation and comparison to measured data. Kimberly A. Riegel and Victor W. Sparrow (Dept. of Acoust., Pennsylvania State Univ., State College, PA 16801, kal337@psu.edu)

Accurately simulating sonic booms in and around structures is the first step to determining the effect on the structures and the people who inhabit them. In order to do these simulations, a combined ray tracing/radiosity method was developed. A variety of parameters was examined in order to determine their effect on the resulting sound field. These parameters include building façade features, building absorption, and diffusion coefficients. These simulations were compared to the NASA 2009 SonicBOBS test data to determine the accuracy of the simulations. Two building configurations were examined: a single building and a two building configuration. [Work supported by the NASA. The authors appreciate NASA making test data available for this work.]

4:15

2pNS12. Interactive noise mapping methods to support resource management in National Parks. Kurt M. Fristrup (Natural Sounds and Night Skies Div., Natl. Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov) and Cynthia Lee (U.S. Dept. of Transportation Res. and Innovative Technol. Administration, Cambridge, MA 02142)

Noise modeling of complex management scenarios spanning large areas requires many days of processing time. To enable a rapid, iterative process for formulating and evaluating management alternatives, the NPS Natural Sounds and Night Skies Division and the John A. Volpe National Transportation Systems Center developed methods to separate noise modeling into two steps. Noise models, such as INM 6.2, are used to predict noise exposures from each combination of aircraft type and route that could be included in the scenarios being considered. This processing step is embarrassingly parallel without modifying any software. The individual noise predictions are ingested by a second program, which combines noise doses to predict aggregate noise exposure from the traffic on all operational routes. For noise duration metrics, such as audibility or time above 35 dB(A), the aggregation of noise exposures requires an algorithm to account for the potential overlap between adjacent noise events. A relatively simple formula has been derived using the assumption that waiting times between noise events have an exponential distribution; this fits model validation data from Grand Canyon quite well. The interactive mapping tool evaluates alternatives in about 1 min of compute time, facilitating intuitive understanding of the consequences of management options.

4:30

2pNS13. Prediction of urban sound propagation via adaptive beam tracing. Carl R. Hart and Siu-Kit Lau (Durham School of Architectural Eng. and Construction, Univ. of Nebraska—Lincoln, 1110 S. 67th St., Omaha, NE 68182, carl.hart@huskers.unl.edu)

Prediction of urban sound propagation is an integral part for the assessment of noise levels in cities. Several propagation modeling methods exist to predict sound propagation in city environments including parabolic equation, diffusion equation, radiosity, ray tracing, image source, modal analysis, and transmission line matrix methods. For this study, adaptive beam tracing as a geometrical acoustic prediction method is applied in the context of predicting sound propagation in complex urban environments. Adaptive beam tracing is advantageous compared to the ray tracing and image source methods by eliminating aliasing errors and spurious image sources, respectively. A model of diffraction [Svensson *et al.*, J. Acoust. Soc. Am. **106**, 2331–2344 (1999)] is integrated with the adaptive beam tracing algorithm in order to enhance the prediction capability of the proposed method. The effects of facade roughness, street canyon shielding, and building spacing are investigated. Comparison of the proposed method is made with a ray tracing method. Computation times, as well as various geometric configurations, are compared between the proposed method and the ray tracing method.

4:45

2pNS14. Scale model investigation of factors affecting the performance of roadside noise barriers. Shira Daltrop, Murray Hodgson (Dept. of Mech. Eng., UBC, 2206 East Mall, Vancouver, BC V6T 1Z3, Canada), and Clair Wakefield (Wakefield Acoust., 2250 Oak Bay Ave., Victoria, BC V5N 1A5, Canada)

Besides numerical models, another way to model complex environments is using physical reduced-scale models. This project used scale models to study several factors which may influence the performance of roadside noise barriers. One is the barrier absorption. Making the barrier sound absorptive decreases reflections and may decrease amplification between parallel barriers. The other is tree foliage growing near the barrier. Tree foliage may scatter sound into the shadow zone behind the barrier, increasing noise levels and decreasing the insertion loss. Tree foliage may also attenuate sound that would normally be diffracted into the shadow zone, actually increasing the insertion loss. A 1:31.5 scale model was created in an anechoic chamber to test the effects of these two factors. Excess attenuation measurements were performed to choose scale model materials, which accurately represent full scale surfaces. Parallel barriers were modeled with a reflective surface and their IL's were measured for different configurations of absorptive covering. IL's of single barriers were measured both with and without scale model trees placed either in front or behind them. The results are compared to previously performed field test measurements.

Session 2pPA

Physical Acoustics and Engineering Acoustics: Advances in Physical, Nonlinear, Engineering, and Atmospheric Acoustics in Memory of Professor Rong Jue Wei–A Pioneer of Acoustics Research and Education in China

Junru Wu, Cochair
Univ. of Vermont, Dept. of Physics, Burlington, VT 05405

Cheri X. Deng, Cochair
Univ. of Michigan, Dept. of Biomedical Engineering, 2200 Bonisteel Blvd., Ann Arbor, MI 48109

Chair's Introduction—1:00

Invited Papers

1:05

2pPA1. Acoustics research in Nanjing University–In memory of Professor Rong-jue Wei. Shu-yi Zhang (Lab. of Modern Acoust., Inst. of Acoust., Nanjing Univ., Nanjing 210093, China, zhangsy@nju.edu.cn)

Acoustics research in Nanjing University was initiated by Prof. Rong-jue Wei in 1954, and developed as an Institute of Acoustics of Nanjing University in 1978, and then also a National Laboratory of Modern Acoustics in 1990. Under the leadership of Prof. Wei, the research of acoustics developed widely from audio acoustics to physical acoustics and ultrasonics, including electro-acoustics, architectural acoustics, speech signal processing, molecular acoustics, atmospheric acoustics, acousto-electronics, low temperature acoustics, nonlinear acoustics, photoacoustics, biological acoustics, environmental acoustics, and noise control. Most of the research fields and works were initial and creative in the country, and some of them have been continued and developed greatly. Up to now, the researches of the institute include five fields as follows: (1) *Intensive acoustics*: ultrasonic cavitation, somoluminescience, soliton and chaos; (2) *Photoacoustics and laser ultrasonics*: photoacoustic spectroscopy and imaging, characterization of nanoscaled materials, biological materials, and metamaterials; (3) *Ultrasonics*: phonon crystals, surface acoustic wave technique, and devices; (4) *Biomedical ultrasonics*: nonlinear ultrasonic phenomena and imaging of biological tissues; (5) *Audio acoustics*: electroacoustics, architectural acoustics, environmental acoustics and noise control, and acoustic signal processing.

1:25

2pPA2. Prof. Rongju Wei and the Institute of Acoustics: A valuable source of US graduate students. Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA) and Juan Tu (Nanjing Univ., Nanjing, People's Republic of China)

Prof. Rongjue Wei was very interested in ensuring that students at the Institute of Acoustics in Nanjing had an opportunity to do graduate work in the United States. He had many contacts with researchers at institutions around the world that performed acoustics research. Over a period of 25 years, Prof. Wei communicated with one of the authors (L.A.C.) about various research efforts undertaken in his laboratory and at various times, suggesting that students at Nanjing could move directly into these programs. On several occasions, these students were admitted to the author's institution and worked under him or his colleagues. This relationship with Nanjing was very productive, producing some outstanding graduates, and still continues.

1:45

2pPA3. Unresolved mysteries of acoustic solitons. Seth Putterman (Dept. of Phys. and Astronomy, UCLA, Los Angeles, CA 90095)

High amplitude motion in a continuous medium often results in localized states caled solitons. These nonlinear dynamical states can exist at amplitudes that are larger and widths that are smaller than are accessible to perturbation theory. A new type of nonlinear mathematics is needed to approach these stable structures.

2:05

2pPA4. Moments to cherish in his life: Our father Rongjue Wei. Yihua Wei (ICF Int., 101 Lucas Valley Rd., San Rafael, CA 94903) and Lanhua Wei (KLA-Tencor, Milpitas, CA 95035)

We would like to take this opportunity to celebrate the life of our father, Prof. R. J. Wei, who devoted his life and passion to physics and acoustics. We would like to share some of the precious moments in his life: from the time early in his career when he came to America to find his dream; to the time he went back to China to establish the first acoustics science and education foundation in the country; to the time he fought even bravely after the national disaster of the cultural revolution to catch up to the world's advances in science; to the time later in life, even though with failing memory, he still cared with all his heart about the scientific progress of his institute and university. Throughout it all, he was a loving father, a respectful advisor, and a good friend to his family, his colleagues, and his students some of whom were a couple of generations younger than he.

2:25

2pPA5. An *ex vivo* study on targeted cell surgery. Zhi Biao Wang, Liao Qiong Fang, Hua Wang, Fa Qi Li, and Junru Wu (Chongqing Key Lab. of Ultrasound in Medicine and Eng., Dept. of Biomedical Eng., Chongqing Medical Univ., Chongqing 400016, China)

High intensity focused ultrasound (HIFU) has become a new noninvasive surgical modality in medicine. It was shown by *ex vivo* experiments that the dimensions of the tissue which experiences CN can be as small as $50 \times 250 \times 250\ \mu m^3$ after 1 s exposure of ultrasonic pulse [the spatial and pulse average acoustic power is on the order of tens of watts and the local acoustic spatial and temporal pulse averaged intensity is on the order of 30 thousands per square meter generated by a 1.6 MHz HIFU transducer of 12 cm diameter and 11 cm geometric focal length (^{f}-number = 0.92)]. The numbers of cells which suffered CN were estimated to be on the order of 40. This result suggests that HIFU is able to interact with tens of cells at/near its focal zone while keeping the neighboring cells minimally affected, and thus the targeted cell surgery may be achievable. [It was funded by National Basic Research Program of China (No. 2011CB707902) and Key Program of National Natural Science Fund by the Chinese National Science Foundation (No. 30830040). J. W. was supported by the HAS fund at University of Vermont.]

2:45

2pPA6. To improve the outcome of intracellular delivery by microbubble-facilitated sonoporation. Cheri X. Deng and Zhenzhen Fan (Dept. of Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Boulevard, Ann Arbor, MI 48109)

Microbubble-facilitated sonoporation, or the ultrasound-induced disruption of plasma membrane, provides new opportunities for intracellular delivery of therapeutic agents. However, ultrasound mediated microbubble-cell interaction is difficult to control due to the very nature of unconfined microbubbles in solution, resulting in relatively low delivery efficiency and often variable delivery outcome. It is desirable to develop techniques in order to achieve reproducible, robust delivery outcome. By utilizing targeted microbubbles to control and confine microbubbles near the cells and examine the detailed biophysical and cellular processes of the interaction of ultrasound-driven microbubbles with cells, we seek to obtain improved understanding of the disruption of cell plasma membrane, cellular uptake, and subsequent downstream effects in sonoporation. In this presentation, we describe our studies that examined these aspects and the important factors involved in sonoporation outcome using multidisciplinary approaches and at the single cell level.

3:05—3:15 Break

3:15

2pPA7. Dislodgement and removal of dust-particles from a surface relevant to NASA project by vibration and acoustic levitation. Junru Wu and Di Chen (Dept. of Phys., Univ. of Vermont, Burlington, VT 05405)

There are many fine particles on the moon and on Mars, which may cause risk for the success of a long-term project for NASA. These dust-particles might cover the solar panels making them fail to generate electricity. The particles would be hazardous to human health if they were inhaled. Development of robust dust mitigation technology is needed for the viable long-term exploration and habilitation of either the moon or Mars. We report here a feasibility study to develop a dust removal technique, which may be used in space-stations or other enclosures for habitation. It is shown that the acoustic levitating radiation force produced by a 13.8 kHz 128 dB sound-level standing wave between a 3 cm-aperture tweeter and a reflector separated by 9 cm is strong enough to overcome the van der Waals adhesive force between the dust-particles and the reflector-surface. Thus the majority of fine particles (greater than 2 μm diameter) on a reflector surface can be dislodged and removed by a technique combining acoustic levitation and airflow methods. The removal efficiency deteriorates for particles less than 2 μm in size. [This work was partially supported by NASA under Cooperative Agreement No. NNX08AZ07A.]

Contributed Papers

3:35

2pPA8. Transmission characteristics of double negativity acoustic metamaterials studied with fluid impedance theory. Li Fan, Shu-yi Zhang, and Hui Zhang (Lab. of Modern Acoust., Inst. of Acoust., Nanjing Univ., Nanjing 210093, China, paslabw@nju.edu.cn)

Input acoustic impedances and transmission coefficients of acoustic metamaterials composed of double negativity structures, i.e., periodic membranes (negative density) and side holes (negative modulus) set along pipes, are studied with the fluid impedance theory, by which the performances of the pipy metamaterials can be studied at any frequency, and the mechanisms of forbidden bands are presented with the theory of acoustic impedance match. In the type of pipy metamaterials, a forbidden band and a pass band successively occur from zero to the upper cut-off frequency of the double negativity. Then, another forbidden band occurs in the frequency range of the single negativity structure formed by the membranes or side holes. Meanwhile, it is found that the input acoustic resistance of the metamaterial is zero in the forbidden bands. Furthermore, other high-frequency forbidden bands caused by the Bragg scattering can be found. All the forbidden bands result from the acoustic impedance mismatch at the positions of the membranes and/or side holes. Finally, the influences of the structural parameters of the metamaterials on transmission coefficients, forbidden and pass band widths, fluctuations in pass bands, etc., are evaluated. [This work is supported by the National Natural Science Foundation of China Nos. 10904067 and 11074125, and Ph.D. Programs Foundation of Education Ministry of China No. 20090091120050.]

3:50

2pPA9. Acoustic cavitation assisted vascular endothelial growth factor transfection *in vitro*. Juan Tu, Dong Zhang, Chunbin Zhang, Qian Li (Key Lab. of Modern Acoust. (Ministry of Education), Dept. of Acoust., School of Phys., Nanjing Univ., Nanjing, Jiangsu 210093, P. R. China, juantu@nju.edu.cn), Feng Gao (Jiangsu Province Hospital of TCM, Nanjing, Jiangsu 210001, China), and Xiasheng Guo (Nanjing Univ., Nanjing, Jiangsu 210093, P. R. China)

Angiogenesis is a complex process that is mediated by growth factors such as vascular endothelial growth factor (VEGF). Researchers have reported that therapeutic ultrasound (US) is able to promote VEGF production *in vitro*, with the acoustic cavitation-induced sonopration. Here, with the presence of ultrasound contrast agents, the vascular endothelia cells mixed with VEGF165 were exposed to 1-MHz US pulses with varied acoustic parameters. The inertial cavitation (IC) activities accumulated during US exposure were quantified as IC dose (ICD) based on passive cavitation detection. The US sonoporation pores were examined with electron scan microscopy. The VEGF expression and the proliferation of the treated cells

were evaluated based on ELISA and MTT assessments, respectively. The results show that (1) the ICD could be affected by US parameters (e.g., pressure, total treatment time, and PRF) and (2) the sonopration pore size, VEGF expression, and the cell proliferation were initially increased with the increasing ICD, then tended to saturate instead of achieving a maximum while the ICD kept going up. The results indicated that IC should play an important role in stimulating VEGF expression through sonopration, and ICD could be used as an effective tool to monitor and control the US-mediated drug delivery outcomes. [This work was funded by the National Basic Research Program of China (2011CB707900) and NSF of China (11074123 and 10974093).]

4:05

2pPA10. Nonlinearly torsional vibrations in acoustic friction force microscope. Hui Zhang, Shu-yi Zhang, and Li Fan (Lab. of Modern Acoust., Inst. of Acoust., Nanjing Univ., Nanjing 210093, China, paslabw@nju.edu.cn)

The tip motion in the acoustic friction force microscope shows the stick-slip behavior for a large displacement in the tip-sample contact, in which the tip begins to slide on the sample when the lateral force is larger than the threshold lateral friction. To investigate the torsional vibrations of the cantilever in the microscope, the ratio of the threshold lateral friction to the product of the lateral contact stiffness and the vibration amplitude is an important parameter. When the ratio tends to 0, the tip purely slips on the sample. When the ratio is equal to or larger than 1, the stick appears, in which the lateral force can be regarded as a linear force applied to the tip. If the ratio changes from 0 to 1, the stick-slip motions of the tip exist, in which the lateral friction is nonlinear and induces a complex torsional vibration of the cantilever. The vibration can be described by a transition state from the pure stick to the pure slip, which is determined by the tribological characterization of the interface of the tip-sample. [This work is supported by NNSF of China (11004099 and 10774074), and supported by State Key Laboratory of Acoustics, CAS.]

4:20

2pPA11. Late studies in Nanjing on dynamic behavior of nonpropagating solitons in Faraday wave experiment. Likun Zhang (Dept. Phys. and Astron., Washington State Univ., Pullman, WA 99164-2814)

The standing solitons [Wu *et al.*, Phys. Rev. Lett. **52**, 1421–1424 (1984)] in Faraday wave experiment with a vertically driven rectangular water tank manifest wide nonlinear phenomena [R. Wei, J. Acoust. Soc. Am. **97**, 3376 (1995)]. In this talk, we will briefly review the history, and then report some relatively recent studies in Nanjing on the dynamic behavior of such solitons, especially on the soliton's creation, temporal modulation, spatial rocking, and the possible route from solitary waves to chaos via spatiotemporal bifurcations and mode competition [Zhang *et al.*, Phys. Rev. E **75**, 036602 (2007)]. Possible mechanisms for the spatial symmetry breaking and head rocking of a solitary wave will be discussed.

4:35

2pPA12. On the acoustic performance of hybrid mufflers. Rong Bi, Jun Chen, Kai Ming Li (Dept. of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031), Rong Bi, and Zheng Shi Liu (Heifei Univ. Tech., Heifei, Anhui 230009, China)

The acoustic attenuation of hybrid mufflers including perforated screens and sound-absorbent materials is investigated in the current study. An analytical method based on a 2-D mode matching technique is used to model an axisymmetrical hybrid muffler. In the theoretical analysis, the hybrid muffler is divided into a number of sections in different homogeneous media with different acoustical characteristics. The acoustic properties of sound absorbent materials and perforated screens are taken into account to obtain eigenfunctions for wave propagation through the air and sound absorbent materials. A system of equations can be solved by matching the acoustic pressure and axial velocity across different sections, which leads to the solution for the determination of the muffler's transmission loss (TL). The numerical predictions of TL show good agreements with published experimental data. The developed analytical approach is then used to examine the effect of the inner structure of the hybrid mufflers on their acoustic performance. Finally, a hybrid muffler with mixed absorbent materials is investigated. The predicted results show that the hybrid muffler with full packed housing can lead to an improved acoustic attenuation at broad band frequencies. [Work sponsored by the China Scholarship Council.]

4:50

2pPA13. Nonlinear vibrations and heating phenomena of cracks in metal plates excited by intensive ultrasonic pulses. Shu-yi Zhang, Zhao-jiang Chen, Kai Zheng, Xiao-bing Mi, and Jiang Zheng (Lab. of Modern Acoust., Inst. of Acoust., Nanjing Univ., Nanjing 210093, China, zhangsy@nju.edu.cn)

When thin metal plates are excited by intensive ultrasonic pulses, nonlinear vibration phenomena are observed in experiments. As the plates contain cracks, the frictional heating of both faces of the crack appears under the excitation of the ultrasonic pulses. Based on the experimental conditions, a finite element modeling is established and transient dynamic analyzes are performed to explore the generation mechanism of the nonlinear vibrations including superharmonics, subharmonics, and chaotics, as well as the heating phenomena of the cracks. A piezoelectric-thermal analogy method is used to simulate the high-power ultrasonic transducer excitation, and a contact-impact algorithm is used to simulate the dynamical interaction between the ultrasonic horn and plate as well as between both sides of the cracks. By analyzing the vibration velocity waveforms, vibration frequency spectra, and contact force waveform, it can be found that the waveform distortion of the intermittent contact forces between the horn tip and the plate is the main reason for generating the nonlinear vibrations. Meanwhile, the intermittent contacts and frictions between both faces of the crack induce heating the area. The numerical simulations show that the acoustic nonlinearities greatly enhance the heating. Generally, the theoretical results agree well with that of the experiments.

Session 2pPP

\Psychological and Physiological Acoustics: Temporal and Spatial Factors (Poster Session)

Julie Bierer, Chair
Univ. of Washington, Dept. of Speech and Hearing Science, 1417 N.E. 42nd St., Seattle, WA 98015-6246

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

2pPP1. Frequency dependence of the interaural time difference thresholds in human listeners. Larisa Dunai (Departemento de Ingeniería Gráfica, Universidad Politécnica de Valencia, Camino de Vera, 46022 Valencia, Spain) and William M. Hartmann (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824)

Interaural time difference (ITD) thresholds for sine tones were measured as a function of frequency with unprecedented resolution along the frequency axis. The tone level was 70 dB SPL, and the method was a two-interval forced-choice, three-down one-up staircase. Overall, the lowest thresholds occurred near 1000 Hz. At lower frequencies, thresholds varied more rapidly than the expected 1/*f* law, suggesting a growing deficit in elemental ITD processors as characteristic frequency decreases. At higher frequencies, thresholds increased dramatically with increasing frequency. Measurements at 50-Hz increments were able to obtain a threshold for only one listener at 1500 Hz, but no threshold at 1550 Hz. In summary, performance varied from best to impossible over a range of about half an octave. In that sense, ITD thresholds appear to show the most dramatic frequency dependence of any auditory quantity. [Work supported by the Vice-rectorate for Faculty and Academic Planning, Universidad Politécnica de Valencia and by the US NIDCD Grant No. DC-00181.]

2pPP2. A re-examination of human underwater sound localization abilities in the azimuth. Michael K. Qin (Naval Submarine Medical Res. Lab., Naval Submarine Base New London, Box 900, Groton, CT 06349), Neil Aaronson (The Richard Stockton College of NJ, Pomona, NJ 08240), Matthew Babina, and Edward Cudahy (Naval Submarine Base New London, Groton, CT 06349)

Divers are frequently exposed to underwater sounds. The subjective impression of the diving community is that sound localization underwater is extremely difficult. However, Feinstein [1973a, 1973b] found underwater minimum audible angles (MAAs) to be approximately 10 deg. To the extent that underwater MAAs reflect the general performance of the binaural system, this would suggest that humans should be reasonably effective at underwater sound localization. A re-examination of the underwater MAAs, with greater subject and environment control, was performed. The present work indicates underwater MAAs at approximately 20–30 deg, significantly poorer than previous findings. [Work supported by the ONR.]

2pPP3. Human underwater and bone conduction hearing in the sonic and ultrasonic range. Michael K. Qin, Derek Schwaller, Matthew Babina, and Edward Cudahy (Naval Submarine Medical Res. Lab., Naval Submarine Base New London, Box 900, Groton, CT 06349)

Several investigators have reported that high-intensity bone-conducted sounds in the ultrasonic range (>20 kHz) can produce auditory sensations in individuals with normal hearing. Underwater hearing threshold studies performed at the Naval Submarine Medical Research Laboratory have found audibility curves similar to those previously reported in bone conduction studies. Human underwater and bone conduction behavioral thresholds in the frequency range between 20 Hz and 200 kHz are presented. Accelerometry measurements made with a mechanical human-head simulator are also presented. The present findings support the argument that the primary mechanism for human underwater hearing is bone conduction. Possible mechanisms for the underwater ultrahigh frequency hearing will be discussed.

2pPP4. Effects of spatial visual information and head motion cues on auditory spatial judgments. Mark A. Ericson and Rachel Weatherless (Army Res. Lab., 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005, mark.a.ericson@us.army.mil)

Virtual audio systems are commonly used today to create auditory environments for various listening tasks. The dimensions and absorption characteristics of the surrounding real visual space often do not match the virtual auditory listening space. This mismatch in spatial information has been shown to have a small or negligible effect in the perceived direction of sound sources. However, spatial cues can have a large effect on perceived auditory distance. The most veridical auditory spatial renderings are ones that match the visual space around the observer. Several experiments were conducted to investigate the effects of visual spatial information and head motion cues on perceived auditory distance. In the first experimental condition, naive observers were deprived of all visual cues while making judgments on the egocentric distance of sound sources played over loudspeakers. In the second condition, subjects were presented with mismatched visual and auditory spatial cues via a manikin. The acoustic manikin was either non-moving or would rotate in correlation with the listener's head motion. In the last (control) condition, sound sources were presented from loudspeakers with matching visual and auditory cues. The effects of the visual spatial cues and head motion cues on auditory distance judgments will be described.

2pPP5. Lateralized processing of audiovisual speech: Combining dichotic listening with visual hemifield effects. Marianne Boettiger (Dept. of Linguist., Univ. of Tuebingen, D-72076 Tuebingen, Germany, marianneboettiger@gmx.de) and Ingo Hertrich (Univ. Tuebingen, D-72076 Tuebingen, Germany)

The perception of acoustically presented speech syllables can be influenced by simultaneous visual presentation of a speaker's face uttering different syllables. Furthermore, previous studies have shown that the visual influence on the phonetic percept depends on the video signal being presented to the left or right hemifield, indicating lateralized processing of visual phonological information. The present study, combining a dichotic listening paradigm (/pa/ and /ta/ syllables) with visual hemifield presentation, investigated the interactions between auditory right-ear advantage, visual hemifield effects, and participants' gender. Most of the items were perceived as /ta/, /pa/, or /pta/. Apart from a strong and highly significant right-ear advantage, single consonant responses showed a right-hemifield advantage, irrespective of place of articulation and gender. By contrast, /pta/ responses were more frequent in females than males and showed an interaction of video side with place of articulation. The largest number of double articulations was perceived if right-ear /ta/ was combined with labial left-hemifield presentation and left-ear /pa/. These results can be taken as an in-

dicator that the mechanisms resulting in a fused single-consonant percept are different from the ones leading to more complex responses requiring cross-modal combinations of phonological information.

2pPP6. Accurate sound localization via head movements in listeners with precipitous high-frequency hearing loss. Ewan A. Macpherson, M. Alasdair Cumming, and Robert W. Quelch (Natl. Ctr. for Audiol., Univ. Western Ontario, 1201 Western Rd., London, ON N6G 1H1, Canada, ewan .macpherson@nca.uwo.ca)

Information about sound source location in the vertical plane is available via the direction-dependent filtering performed by the outer ears, but errors of localization such as front/rear reversals can occur when stimuli contain a limited range of frequencies or when high frequencies are inaudible due to hearing impairment. Information about front/rear sound source location is also available in the relationship between the rotation of the head and the resulting changes in interaural time and level differences. We have shown previously [Macpherson, J. Acoust. Soc. Am. 125, 2691(A) (2009)] that in normally hearing listeners, a minimum head movement angle (MHMA) of 5–10 deg is sufficient for accurate front/rear localization of low-frequency (0.5–1 kHz) noise-band targets. In the present study, we measured MHMAs for low-frequency and wideband (0.5–16 kHz) targets in listeners with near-normal low-frequency thresholds but precipitous hearing loss above 1–2 kHz. Neither stimulus could be localized accurately by these listeners without head movement, but for both stimuli, MHMAs of 5–10 deg sufficed for accurate localization at a rotation velocity of 50 deg/s. MHMAs increased with increasing rotation velocity similarly to those of normally hearing listeners. The results suggest that listeners with normal hearing and with high-frequency loss benefit similarly from dynamic localization cues. [Work supported by the NSF.]

2pPP7. Binaural synthesis of virtual moving sound sources. James T. Hamil, Ashok K. Krishnamurthy (Dept. of Elec. and Comput. Eng., The Ohio State Univ., Columbus, OH 43210, hamil.3@osu.edu), and Lawrence L. Feth (The Ohio State Univ., Columbus, OH 43210)

Limitations in the movement of physical sources for motion related studies motivate the creation of a processing chain for accurate placement of moving sound sources in a three-dimensional virtual auditory space over headphones. Monaural and binaural cues normally available to listeners are accurately reproduced in the ear waveforms. Head-related transfer functions (HRTFs) are updated at a high rate to reflect each individual source's changing directional information. A time-varying propagation delay is implemented for linear motion to account for the Doppler frequency shift that occurs when either the source or listener is moving in virtual space. The performance of the system is characterized through psychoacoustic measures of spatial resolution and compared to the existing literature. System constraints and potential future experiments are addressed.

2pPP8. Using spiking onset neurons and a recurrent neural network for sound identification. Michael Newton and Leslie Smith (Dept. of Computing Sci. and Maths, Univ. Stirling, Stirling FK9 4LA, United Kingdom, mjn @cs.stir.ac.uk)

Physiological evidence suggests that specific neurons within the cochlear nucleus specialize in sound onset detection. These are innervated by type 1 spiral ganglion fibers covering a relatively wide spectrum. Sudden increases in sound energy (e.g., during the initial portion of a sound) result in an increased firing rate in a downstream onset neuron. Onset timing and spectral location are thought to play a role both in auditory stream separation and sound identification and interpretation. Onset neurons are modeled using leaky integrate-and-fire units innervated by spiking data streams produced using a passive gammatone filterbank followed by positive-going zero-crossing detection. Signal level is coded using multiple spike trains per filter channel. The model is presented with a succession of 607 musical samples selected from the McGill dataset and the pattern of onset spikes recorded for each sound. Groups of onset spikes occur close to the beginning of each note. The objective is to use the pattern of spikes, produced by the onset neuron model, as a fingerprint of the original acoustic signal. These onset fingerprints are presented to a recurrent neural network (reservoir network) to attempt to classify them. The results are compared with a sound classification scheme based on cepstral coefficients.

2pPP9. A binaural model to simulate the precedence effect with adaptive stages to compensate for head movements. Jonas Braasch, Anthony Parks, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180)

Several models exist to simulate *localization dominance*, the ability of the auditory system to disregard room reflections when determining the location of a sound source. Psychoacoustic experiments have shown that the suppression of a reflection can improve over time, an effect known as the *build up of the precedence effect*. A recently introduced binaural model [Braasch, J. Acoust. Soc. Am. **124**, 2494] is an ideal candidate to simulate this buildup effect. The model estimates the optimal parameters to inhibit a reflection using an autocorrelation algorithm, and thus the buildup can be understood as a learning process to tune these parameters optimally. The question of how the auditory system adapts to stimuli changes when the listener changes position, however, has not been addressed adequately. An initial attempt to simulate this condition is made here by combining the aforementioned binaural model with one that compensates for head movements, which is achieved by rotating the binaural activity pattern in the opposite direction, but with the same magnitude, of the head movements [Braasch *et al.*, J. Acoust. Soc. Am. **127**, 1886]. This way, the precedence effect model is able to adapt to the new situation without having to relearn the configuration of direct sound and reflection.

2pPP10. The precedence effect: Spatial versus cue specificity. Andrew D. Brown and G. Christopher Stecker (Andrew D. Brown, Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, andrewdb@uw.edu)

Normal-hearing listeners effectively localize sound in everyday environments by responding to the directional cues carried by the early-arriving direct signal rather than the spurious cues carried by its reflected copies. This so-called precedence effect is known to depend strongly on the stimulus context. In a previous study, we demonstrated a divergence of echo thresholds for interaural time difference (ITD) and interaural level difference (ILD) under conditions of repeated stimulation. Specifically, while buildup and re-buildup effects were evident for both cues, a breakdown effect occurred for ILD only. This result suggests that dynamic precedence in the free field might depend on sensitivity to discrete ITD and ILD images. The present study assessed whether echo thresholds built up by ITD or ILD conditioner stimuli would remain elevated for subjectively matched ILD or ITD test stimuli. Stimuli were pairs or slow trains of broadband clicks presented over headphones. Echo thresholds were estimated from subjects' judgments of fusion and lateral position. To the extent that buildup depends specifically on the spatial character of the stimulus, built-up echo thresholds should transfer between spatially equivalent cues. Echo thresholds measured at baseline levels following a change in cue type, however, suggest that buildup occurs in a cue-specific manner. [Work supported by NIH Grant Nos. F31-DC010543 (A.D.B.) and R03-DC009482 (G.C.S.).]

2pPP11. Influence of a single reflection on speech intelligibility in simulated electric-acoustic stimulation. Kate Helms Tillery, Christopher A. Brown, William A. Yost, and Sid P. Bacon (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287, kate.helms-tillery@asu.edu)

In reverberant environments, reflections occurring within 30–50 ms of the direct speech assist intelligibility by perceptually fusing with the source, effectively increasing its level. Fusion occurs at similar delays in monaurally deaf and normal-hearing listeners, suggesting an independence from binaural processes [Litovsky *et al.*, J. Acoust. Soc. Am. **106**, 1633–1654 (1999)]. Its effects can thus be examined in unilateral cochlear implant (CI) users. Simulated CI listening shows little benefit from early reflections, and a detriment when delays exceed 20 ms [Whitmal and Poissant, Conference on Implantable Auditory Prostheses (2009)]. Data from our laboratory show that CI users with residual acoustic hearing benefit from electric-acoustic stimulation (EAS) in reverberation. The role of early reflections in this EAS benefit remains unclear. The present study examined the effect of a single unattenuated reflection at ten delay times on intelligibility in simulated EAS. Target stimuli consisted of sentences combined with unattenuated copies delayed by 0–66 ms. Four-talker babble was added to increase difficulty. A four-channel vocoder simulated electric stimulation; low-pass filtered speech represented residual acoustic hearing. Monaural intelligibility scores from normal-hearing listeners under anechoic and reflected conditions with electric and electric-acoustic processing suggest that EAS may facilitate a benefit from early reflections. [Work supported by the NIDCD.]

2pPP12. Threshold of the precedence effect in the presence of interfering noise. Richard L. Freyman, Amanda M. Griffin (Dept. of Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003), and Patrick M. Zurek (Sensimetrics Corp., Malden, MA 02148)

Recent work suggests that the precedence effect involves three distinct phenomena: (1) a transient onset that enhances early spatial cues relative to those arriving shortly after; (2) the Franssen effect, in which localization /lateralization estimates established at onset are maintained throughout a long-duration stimulus; and (3) an ongoing precedence effect, in which leading cues throughout a long stimulus determine localization and lateralization. The goal of the present study was to determine whether the three phenomena are influenced to differing degrees by interference from other sources. Brief and longer trains of noise bursts that alternated between 0- and 500-μs interaural time delays were created to elicit the three types of phenomena. Masked thresholds for the stimuli beginning with 0-μs interaural delays were nearly identical to those beginning with 500 μs across the different types, indicating no precedence effect for detection threshold. Listeners' ability to discriminate the 0-μs-leading and 500-μs-leading stimuli did not reach threshold performance until at least 12 dB above masked threshold, indicating that none of the three types of precedence effects is strong near detection threshold. However, the threshold of the Franssen effect was substantially higher than that of the other two phenomena. [Work supported by NIDCD Grant No. DC01625.]

2pPP13. Perceptual constancy for real-room reverberation in loudness judgements and the effects of interstimulus interval. Andrew P. Raimond and Anthony J. Watkins (Dept. of Psych., Univ. of Reading, Reading, UK)

Narrowband stimuli shaped with fast-onset and slow-offset temporal envelopes are judged to be less loud than their reversed counterparts, particularly when preceded by a slow-offset standard. It has been suggested that a "perceptual constancy" may be responsible for this "loudness context effect." The idea is that when a slow-offset "tail" is at the end of a sound it is largely attributed to listening-environment reflections and is discounted in the listener's loudness judgment. Previous experiments have shown that this effect is even bigger when real-room reflection-patterns are used. The present experiments varied the interval between the standard and comparison tones (ISI) to investigate the contribution of time-order effects to these loudness judgements. Such effects might arise as the peak-to-peak interval between sounds changes with reversal of the standard, thereby confounding the duration of this interval with the presence of a "tail" in the standard. Results show that the loudness context effect reported earlier remains prominent across a range of ISI times and that the influence of any time-order effects is relatively weak. Consequently, these findings remain consistent with the idea that the context effect arises through a perceptual constancy. [Work supported by EPSRC.]

2pPP14. Learning of reverberation cues for auditory distance perception in rooms. Norbert Kopčo, Beata Tomoriová (Dept. of Cybernetics and AI, Tech. Univ. of Košice, Letná 9, Košice 04001, Slovakia, kopco@tuke.sk), Pierre Silvera, Konstantin Tskhay, and Aaron Seitz (Univ. of California, Riverside, CA 92521)

Listeners must calibrate to the room acoustics in order to judge source distance using reverberation. A learning process underlies this calibration, resulting in improved performance when auditory distance is examined repeatedly in the same room over the course of days. The processes of calibration and learning are spontaneous, not requiring any feedback about the actual target location. The current study examined whether the amount of spontaneous learning in rooms is dependent on the relative strength with which the reverberation cue is used for the distance judgments. Listeners judged distance of broadband noise bursts presented from distances ranging from 0.15 to 2 m directly ahead of the listener in a small rectangular classroom. The stimulus presentation level was either roved from trial to trial (R runs) or it was fixed within an experimental run (F runs). The subjects performed several experimental sessions over multiple days. One subject group was trained on the F runs, one on the R runs. Learning was observed in the R group but not in the F group, confirming that focusing on the reverberation cue is required for the learning and room calibration to occur. [Work supported by NIH and the European Community.]

2pPP15. Auditory and visual distance estimation. Paul W. Anderson and Pavel Zahorik (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY, paul.anderson@louisville.edu)

Past research has shown auditory distance estimation improves when listeners are given the opportunity to see all possible sound sources when compared to no visual input. It has also been established that distance perception is more accurate for visual stimuli than auditory stimuli. The present study investigates the degree to which auditory distance perception is improved when matched with a congruent visual stimulus. Virtual sound sources based on impulse response measurements made from distances ranging between 0.3038 and 9.7536 m in a concert hall were used as auditory stimuli. Visual stimuli were photographs taken from the listener's perspective at each distance in the impulse response measurement setup presented on a large HDTV monitor. Listeners were asked to estimate egocentric distance to the sound source in each of three conditions: (A) auditory only, (V) visual only, and (A+V) congruent auditory/visual stimuli. Each condition was presented within its own block. Distance estimates from both V and A+V conditions were found to be considerably more accurate and less variable than estimates from the A condition. These results are consistent with visual capture phenomena that have been demonstrated in other spatial tasks with input from both auditory and visual modalities.

2pPP16. Adaptation to room acoustics using the modified rhyme test. Eugene J. Brandewie and Pavel Zahorik (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 2301 S. Third St., Louisville, KY 40208, eugene .brandewie@louisville.edu)

The negative effect of reverberant sound energy on speech intelligibility is well documented. Recently, however, prior exposure to room acoustics has been shown to increase intelligibility for a number of listeners in simulated room environments. This room adaptation effect, a possible extension of dynamic echo suppression, has been shown to be specific to reverberant rooms and requires binaural input. Because this effect has been demonstrated only using the coordinated response measure (CRM) corpus it is important to determine whether the increase in intelligibility scores reported previously was due to the specific nature of the CRM task. Here a comparable room-acoustic effect was demonstrated using the modified rhyme test corpus in multiple room environments. The results are consistent with the idea that the room adaptation effect is a natural phenomenon of listening in reverberant environments.

2pPP17. Amplitude modulation detection by human listeners in sound fields. Pavel Zahorik (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu), Duck O. Kim, Shigeyuki Kuwada (Univ. of Connecticut Health Ctr., Farmington, CT 06030-3405), Paul W. Anderson, Eugene Brandewie, and Nirmal Kumar Srinivasan (Univ. of Louisville, Louisville, KY 40292)

The temporal modulation transfer function (TMTF) approach allows techniques from linear system analysis to be used to predict how the auditory system will respond to arbitrary patterns of amplitude modulation (AM). Although this approach forms the basis for a standard method of predicting speech intelligibility based on estimates of the acoustical modulation transfer function (MTF) between source and receiver, human sensitivity to AM as characterized by the TMTF has not been extensively studied under realistic listening conditions, such as in reverberant sound fields. Here, TMTFs (octave bands from 2–512 Hz) were obtained in three listening conditions simulated using virtual auditory space techniques: diotic, anechoic sound field, and reverberant room sound field. TMTFs were then related to acoustic MTFs estimated using two different methods in each of the listening conditions. Both diotic and anechoic data were found to be in good agreement with classic results, but AM thresholds in the reverberant room were lower than predictions based on acoustic MTFs. This result suggests that simple linear systems techniques may not be appropriate for predicting

TMTFs from acoustical MTFs in reverberant sound fields, and may be suggestive of mechanisms that functionally enhance modulation during reverberant listening.

2pPP18. Localizing amplitude-modulated, high-frequency tones in free field: Perceptual. Eric J. Macaulay, Brad Rakerd, and William M. Hartmann (Michigan State Univ., East Lansing, MI, 48824)

High-frequency sinewave-amplitude-modulated (SAM) tones can be lateralized in headphone experiments based on the interaural level difference (ILD) and the interaural time difference in the envelope (eITD). Because of the vagaries of diffraction by the human head, the eITD may be an unreliable cue for localizing SAM tones in free field [Hartmann *et al.*, J. Acoust. Soc. Am. **128**, 2454 (2010)]. Localization experiments were done using a 13-loudspeaker array covering 90 deg of azimuth while binaural ear canal recordings were made to accompany each localization judgement. Parallel experiments for pure sine tones and SAM tones revealed the differences in localization judgements attributable to the modulation. These difference were compared with the differences in ILD and with the eITD for SAM tones, and also with differences in compressive functions of those interaural parameters. Correlations for eITD were low for most listeners, suggesting that the role played by eITD in azimuthal localization is not expressible by a monotonic function. Apparently localization of high-frequency SAM tones is quite different from lateralization. [Work Supported by the NIDCD, Grant No. DC-00181.]

2pPP19. Locating two sound sources: The role of amplitude modulation and spectral content. Willim A. Yost, Christopher Brown, and Farris Walling (Speech and Hearing Sci., Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287)

We have shown that listeners can locate two sound sources producing simultaneous sounds when the sounds are independently generated wideband noise bursts. Listeners' localization performance is better when the sound at one source is sinusoidally amplitude modulated (AM) 180 deg out-of-phase relative the AM of the other sound as opposed to conditions in which the envelope modulation phase is the same (in-phase) for the sound at both sources. This improved localization performance can be measured up to AM rates of 200 Hz. Listeners can discriminate out-of-phase from in-phase modulation between the two sounds up to AM rates of 500 Hz. In the current study, we use filtered noises (lowpass: 125–500 Hz, highpass: 1500–6000 Hz, and wideband: 125–6000 Hz) in an attempt to determine the role interaural time and level differences play in localizing the sources of two independently generated AM sounds. The results will be discussed in terms of the cues listeners might use in localizing multiple sound sources when the sounds occur simultaneously.

2pPP20. Binaural beat rate discrimination. John H. Grose, Emily Buss, and Joseph W. Hall, III (Dept. of OHNS, Univ. N. Carolina at Chapel Hill, 170 Manning Dr., Chapel Hill, NC 27599-7070)

Rate discrimination for binaural beats was measured for standard rates of 4, 8, 16, and 32 Hz in normal-hearing adults. Binaural beats were generated by presenting pairs of 1250-ms tones dichotically at a level of 65 dB sand pressure level (SPL) through insert earphones. This duration allowed for 5 cycles of the lowest (4 Hz) modulation. Tone frequency was roved between 400 and 500 Hz on a presentation-by-presentation basis to avoid monaural place cues for discrimination; starting phase was random. The ratio of a just-discriminable increase in rate relative to a standard rate was fairly constant across the four beat frequencies. The salience of the beat percept was matched to that of sinusoidal amplitude modulation (SAM) by varying the depth of modulation of a diotic SAM tone having the same rate as a given standard beat. Rate discrimination for this salience-matched SAM tone was then measured, again with frequency rove keeping all components within the 400–500 Hz region. Whereas rate discrimination for the salience-matched SAM tone tended to be poorer than that of binaural beat discrimination, rate discrimination for a 100% modulated SAM tone tended to be better. The implication of these results for binaural sluggishness will be discussed. [Work supported by NIH NIDCD Grant No. 5R01DC001507.]

2pPP21. Age-related changes in the binaural temporal window. David A. Eddins, Ann Clock Eddins, and D. Robert Frisina (Dept of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave, PCD 1017, Tampa, FL 33620)

Substantial data indicate that monaural temporal processing, sound localization ability, and spatial release from masking decline with advancing age. Physiological and psychophysical evidence indicate that the auditory system is quite sensitive to patterns of interaural cross correlation (IACC) as determined by the combined input from the two ears. Here we test the hypothesis that age-related changes in temporal processing lead to an elongation of the binaural temporal window as measured with an interaural correlation change interval (ICCI) task. The ICCI threshold was measured as a function of the duration change in interaural correlation. Band pass (400–3200 Hz) noise bursts were separated into three segments. Initial and final segments were interaurally uncorrelated while central segments were interaurally uncorrelated in the standard intervals and interaurally correlated in the signal interval. The duration of the central segment was adaptively varied to determine ICCI threshold. The just noticeable difference in static interaural correlation also was measured. The perception of static and dynamic changes in interaural correlation as well as estimates of the equivalent rectangular duration of the binaural temporal window will be compared across groups of younger and older adult subjects.

2pPP22. Effects of non-simultaneous modulated maskers on amplitude-modulation depth discrimination. Rebecca E. Millman (York Neuroimaging Ctr., The Biocentre, York Sci. Park, Heslington YO10 5DG, UK)

Wojtczak and Viemeister [J. Acoust Soc. Am. **118**, 3198–3210 (2005)] demonstrated that detection thresholds for amplitude modulation (AM) can be increased in the presence of an AM forward masker. In the present study, the effects of forward-masking on AM detection thresholds were extended to AM processing for easily detectable (suprathreshold) AM modulation depths by measuring the effects of nonsimultaneous AM maskers on AM modulation depth discrimination limens (AM DLs). Signal AM DLs were measured for a range of standard modulation depths. In one experiment the temporal delay between the masker and signal was varied. Consistent with the effects of AM forward maskers on AM detection thresholds [Wojtczak and Viemeister (2005)], the effects of the non-simultaneous AM maskers on AM DLs were greatest for short masker-signal delays. In another experiment, the masker-signal delay was fixed, the signal modulation frequency was fixed, and the modulation frequency of the non-simultaneous masker was varied. AM DLs for the signal were usually greatest when the signal modulation frequency was equal to the masker modulation frequency. The results of these experiments suggest that non-simultaneous AM maskers can affect suprathreshold AM processing but non-simultaneous AM maskers become less effective as the modulation depth of the standard increased.

2pPP23. Spectro-temporal specificity of learning on modulation depth discrimination. Andrew T. Sabin (Dept. of Commun. Sci. and Disord., Northwesten Univ., 2240 Campus Dr., Evanston, IL 60208, a-sabin@northwestern.edu), David A. Eddins (Univ. of South Florida, Tampa, FL 33620), and Beverly A. Wright (Northwesten Univ., Evanston, IL 60208)

Most natural sounds are composed of peaks and valleys of sound energy spread across both frequency (spectral modulations) and time (temporal modulations). Here a perceptual-learning paradigm was used to explore how these modulations are encoded. Three groups of normal-hearing listeners were trained to discriminate the depth of either spectral, temporal, or spectro-temporal modulation 1 h/day for 7 days. Each group improved on their trained condition by an amount greater than a group that received no training. However, this training did not lead to improvement on untrained modulation depth discrimination conditions, even when those conditions shared the spectral or temporal modulation frequency with the trained condition. Further, the influence of discrimination training on the detection of spectro-temporal modulation differed across the three trained modulations: spectral training yielded improvement, temporal training had no effect, and spectro-temporal training actually led to worsening on spectro-temporal modulation detection. These results imply that each training regimen led to a different modification. Critically, the specificity of the depth discrimination learning suggests that some aspect of the training-induced modification was selective for the combined spectro-temporal properties of the trained modulation and is therefore consistent with the idea that filters with combined spectro-temporal selectivity underlie modulation perception. [Work supported by NIH/NIDCD.]

2pPP24. Investigating the relationship between carrier frequency and effective bandwidth of peripheral filtering in an amplitude-modulation notch detection task. Allison I. Shim, Bruce G. Berg, and Ewa M. Borucki (Dept. of Cognit. Sci., Univ. of California, Irvine, 2201 Social & Behavioral Sci. Gateway Bldg., Irvine, CA 92697-5100, ashim@uci.edu)

Effective bandwidths of initial auditory temporal processing are measured in an amplitude-modulation in notched-noise detection task. The signal is a 200 Hz wide, sinusoidally amplitude-modulated narrowband noise centered at four different center frequencies. The frequency of the sinusoidal modulator is set at 10 Hz. The use of a slow modulation frequency aims to avoid possible central limitations of temporal processing at higher modulation frequencies. Threshold functions are obtained for 10–14 notch widths for each center frequency to determine the maximum notch width at which the masker has an effect. Averaged thresholds for each center frequency are fitted using a leaky integrator model with a min/max decision statistic. Two free-parameters define the high and low slopes of the initial bandpass filter and a third represents internal noise. Filter bandwidth estimates are wider than critical bands and do not exhibit a constant-*Q* relationship. It is proposed that the initial level of integration is not a basilar membrane process but a process that involves the reconstruction of information from the auditory nerve. [Work supported by NSF Grant No. BCS-0764003]

2pPP25. Dynamic range compression effects on modulation detection interference. Jungmee Lee, Pamela Souza, Andy Sabin (The Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL 60208, jmlee6@northwestern.edu), Bomjun Kwon (Ohio State Univ., Columbus, OH 43212), Marc Brennan (Boys Town Natl. Res. Hospital, Omaha, NE 68131), Gayla Poling, and Carla Petersen (Northwestern Univ., Evanston, IL 60208)

Some evidence suggests that perceived amplitude modulation (AM) depth may be exaggerated for listeners with cochlear hearing loss (HI) compared to listeners with normal hearing (NH) due to a less compressive auditory system. Consistent with this, several researchers have demonstrated that interfering AM causes more masking of a target AM [modulation detection interference (MDI)] in HI than NH listeners. This effect might be due to the exaggerated perceived modulation depth in HI listeners. There is also evidence that the modulation depth of amplitude envelope fluctuations will be reduced by dynamic range compression (often used in hearing aids). MDI should therefore be smaller with a compressed interferer, due to the reduced modulation depth of the interferer. However, Shen and Lentz [(2010)] showed the opposite result. Specifics of that study included a processor that resulted in some artifacts and somewhat extreme compression parameters. The present study uses a low-distortion compressor with a range of compression parameters to investigate the effect of compression representative of wearable aids on MDI. We will report the influence of input level (*re* kneepoint) and attack/release time on MDI in NH and HI listeners.

2pPP26. Amplitude-modulation detection in reproducible modulation maskers: Correlation between Behavioral and Physiological Responses. Muhammad Zilany (Depts. of Biomedical Eng. and Neurobiology and Anatomy, Univ. of Rochester, NY 14642), Kristina Abrams, Kelly-Jo Koch (Dept. of Neurobiology and Anatomy, Univ. of Rochester, NY 14642), Fabio Idrobo (Boston Univ., MA 02215), and Laurel Carney (Univ. of Rochester, NY 14642)

The use of reproducible noise maskers in studies of detection has proven useful to extend our understanding of coding and processing of complex sounds. In an effort to reveal potential neural mechanisms for the detection of amplitude-modulation (AM) in the presence of modulation maskers, tetrode recordings were made from inferior colliculus neurons of awake rabbits using reproducible stimuli. Both rabbit and human behavioral data have been collected for the same set of reproducible maskers. The target AM and maskers were applied to the envelope of a 5 kHz carrier signal. Neural responses were recorded to stimuli with the level of target AM slightly above masked threshold for both rabbit and human. Neural detection thresholds were estimated based on average discharge rate and several temporal metrics, such as synchrony to the target AM, temporal reliability, and rapid fluctuation in the peri-stimulus time histogram (PSTH) of responses. The hit and false-alarm rates for neural detection of AM were estimated for a set of 20 reproducible AM maskers. Preliminary results suggest that some neural responses, based on mean rate, synchrony to the target AM, or the slope of the PSTH, can explain the parallel rabbit behavioral results. [Work supported by NIH-NIDCD R01-001641.]

2pPP27. Stimulus-based diotic and dichotic models that combined cues for detection of tones in reproducible noise. Junwen Mao (Dept. of Elec. and Comput. Eng., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, maojunwen@gmail.com), Azadeh Vosoughi (Rochester, NY 14627), and Laurel H. Carney (Univ. of Rochester, Rochester, NY 14642)

Understanding speech in noisy backgrounds is a classical problem for people with hearing loss. The mechanisms for detection of pure tones in noise are still not fully understood. In this study, models that combined cues across epochs were constructed to predict binaural psychophysical experimental results. The data came from previous experiments, in which listeners were required to detect a 500-Hz tone in either narrowband or wideband reproducible noise waveforms in both diotic and dichotic conditions. For the diotic condition, cues were correlated and could be described by a multivariate distribution. Cues were combined using the covariance matrix. Decision variables based on the likelihood ratio of the tone presence were compared to the listeners' detection patterns. Energy, envelope-slope, and fine-structure cues were combined optimally in the model. For the dichotic condition, the same strategy was applied by combining interaural level and interaural time differences. Better predictions were obtained from this model compared with models that relied on a weighted sum of the two cues. In addition, a modified envelope-slope model that computed the slope of the difference of the envelopes at the two ears yielded better predictions than models based on interaural time and level differences for the dichotic condition. [NIH-NIDCD-R01DCO1O813.]

2pPP28. Differential contribution of target and masker temporal fine structure to the segregation of speech from noise. Frederic C. Apoux, Carla Y. Berg, Sarah E. Yoho, and Eric W. Healy (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210, apoux.1@osu.edu)

In recent years, it has been suggested that the normal auditory system relies on temporal fine structure (TFS) cues to segregate speech from noise. Based on previous work, we hypothesized that only the target TFS is critical for understanding speech in noise. To assess this hypothesis, the TFS of the target and that of the masker were manipulated simultaneously or independently. Four masker types were used. A 30-band vocoder was used to replace the original TFS of the stimuli with tones. Results showed a significant drop in performance when the TFS of the target and that of the masker were disrupted simultaneously. They also showed a significant drop in performance when only the TFS of the target was disrupted. In contrast, disruption of the masker TFS had no effect on intelligibility. Overall, the present data are consistent with previous work showing that TFS information plays a significant role in speech recognition in noise, especially when the noise fluctuates over time. However, the present study indicates that listeners rely primarily on TFS information in the target and that the nature of the masker TFS has a very limited influence on the outcome of the unmasking process. [Work supported by NIDCD.]

2pPP29. Localization in noise and the role of forward masker fringe. Brian D. Simpson (Air Force Res. Lab., WPAFB, OH 45433, brian.simpson@wpafb.af.mil), Robert H. Gilkey (Wright State Univ., Dayton, OH 45435), Douglas S. Brungart (Walter Reed Army Med Ctr., Washington, DC 20307), Nandini Iyer (Air Force Res. Lab., WPAFB, OH 45433), James T. Hamil (Ohio State Univ., Columbus, OH 43210), and Billy Swayne (Ball Aerosp. Technol. Corp., WPAFB, OH 45433)

Previous research suggests that the localization of a target sound in a masking noise is more accurate when the masker is preceded by a forward masker fringe. It is unclear what information listeners are exploiting to produce this improvement in performance. Does the fringe simply reduce spatial uncertainty (by cuing the location of the masker) or does it provide a "context" against which the onset of the signal is more easily discerned? In order to address these questions, we examined conditions in which the

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masker is (a) pulsed on and off with a 60 ms target (no fringe), (b) immediately preceded by a 500-ms masker fringe; or (c) preceded by a 500-ms fringe + a 500-ms gap. Masker spatial uncertainty was also manipulated by presenting the noise from a fixed and known location throughout a block, or from a location that varied across trials. The results suggest that listeners benefit from the presence of a forward masker fringe even when the masker location is known. However, localization performance is best when this fringe occurs immediately prior to the masker (i.e., no gap), suggesting that onset-related cues also contribute to the improvement in localization. [Work supported by the AFOSR]

2pPP30. Concurrent profile analysis: The effect of segregation cues on spectral-shape comparisons for simultaneous stimuli. Yi Shen and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 184 Social Sci. Lab, Irvine, CA 92697, shen.yi@uci.edu)

The ability to accurately perceive spectral information from one sound source in the presence of others is crucial for speech understanding in complex and challenging environments. In the present study, non-speech stimuli, a pair of concurrent complex tones, were used to obtain psychophysical measurements of this ability in a controlled setting. Listeners detected whether the spectral profiles of the two concurrent complexes were the same or exchanged across two observation intervals. Concurrent profile analysis thresholds were measured as the minimum amount of spectral alterations for the detection of the change. Thresholds were measured from four normal hearing listeners and as functions of three types of segregation cues: difference in fundamental frequency, difference in interaural time differences, and onset asynchrony. The roles of these cues in concurrent spectral-shape analysis are discussed in terms of the underlying psychometric functions.

2pPP31. Relationship between selective auditory attention and brainstem encoding in musicians and non-musicians. Li Zhu (Dept. of Biomedical Eng., Tsinghua Univ., Beijing 10084, China), Jing Xia, and Barbara Shinn-Cunningham (Boston Univ., Boston, MA 02215)

This study explores whether differences in performance in a challenging selective auditory attention task are related to the strength of the sub-cortical encoding of stimulus pitch. In each behavioral trial, two simultaneous digits, differing in pitch by 6 semitones, were presented. A preceding visual cue instructed listeners to report the target digit with either the higher or lower pitch. Subjects both were more accurate at reporting the target digit and had a shorter reaction time when the visual-cue-to-auditory-target interval was long compared to trials with a short preparatory interval. Moreover, behavioral performance was generally better for subjects with musical training than for non-musicians. The brainstem frequency-following responses (FFRs) were recorded in response to a complex tone in both quiet and in noise for the same listeners. Consistent with past reports, musicians demonstrated a stronger FFR than non-musicians; in addition, the musicians' FFR response was less affected by noise compared to non-musicians. The strength of the FFR correlated with individual performance on the pitch-based selective auditory attention task. Results suggest that the robustness of the brainstem encoding of acoustic inputs directly determines the ability to selectively attend to an auditory target using pitch differences.

2pPP32. Short-comings in direct translations of names for hearing sensations: A case study regarding the roughness sensation. Stephan Paul (Acoust. Eng. Program, DECC, Fed. Univ. of Santa Maria, Av. Roraima 1000, Camobi, Santa Maria 97060-900, RS, Brazil)

Many psychoacoustic models, now widely used, were first developed in Germany, especially at Zwickers laboratory, and originally named with German terms that correspond to hearing sensations as named by German subjects. Often the terms were readily translated into other languages, especially English and French for publication and nowadays these translations are widely used and accepted. This also applies to the concept of psychoacoustic roughness, named "Rau(h)igkeit" in German. The Munich school developed the first models to describe the sensation of Rau(h)igkeit, found to be caused by fast AM or FM modulation, also present in the spoken German "R" which initiates the term Rau(h)igkeit. Even when the direct translation of the term Rau(h)igkeit and roughness to other languages seems to be quite adequate particularly for Brazilian Portuguese it has been found, by means of subjective evaluation of different signals and signal analysis that this translation is problematic as the corresponding term in Portuguese ("aspero") is used by naive and non-naive Brazilian subjects to name the auditory sensation due to stochastic signals, especially white noise which is unmodulated. So far the hearing sensation described as aspero does not correspond to the sensation of originally described as Rau(h)igkeit and roughness models will fail.

TUESDAY AFTERNOON, 24 MAY 2011 DIAMOND, 1:00 TO 4:45 P.M.

Session 2pSA

Structural Acoustics and Vibration, Noise, and Engineering Acoustics: Near-Field Acoustical Holography

Kent L. Gee, Cochair
Brigham Young Univ., Dept. of Physics and Astronomy, Provo, UT 84602

Scott D. Sommerfeldt, Cochair
Brigham Young Univ., Dept. of Physics and Astronomy, Provo, UT 84602

Invited Papers

1:00

2pSA1. On the spatial resolution of using the Helmholtz equation least squares method to visualize coherent point sources in a free field. Richard Dziklinski, III and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr, Detroit, MI 48202, aa6036@wayne.edu)

Reconstruction of the sound fields by coherent sources has always been a great challenge for the acoustics community because any slight phase shifts among coherent sound waves can cause significant changes in the results. This paper explores the feasibility of using the Helmholtz equation least squares (HELS) and modified HELS based nearfield acoustical holography (NAH) to visualize coherent point sources in a free field. Our goal here is to locate the coherent sources precisely, but not to reconstruct their amplitudes accurately. In particular, we examine the spatial resolution limitations in using HELS and modified HELS to locate coherent sources, and why the guidelines established for the Fourier acoustics based NAH and Shannon–Nyquist sampling rate can by violated. Moreover, the impacts of stand-off distance, frequency, microphone spacing, source separation distance, and signal to noise ratio on the spatial resolution offered by HELS and modified HELS are studied through numerical simulations. The results are compared with those obtained by the Fourier acoustics. Test results show that the modified HELS always provides the highest spatial resolution over the widest range of application scenarios, and to achieve comparable spatial resolution, Fourier acoustics requires much more measurement points than HELS and modified HELS methods do.

1:20

2pSA2. Experimental validations of using the Helmholtz equation least squares method to visualize coherent sources. Richard Dziklinski, III and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, aa6036@wayne.edu)

This paper presents experimental validations of using the Helmholtz equation least squares (HELS) method to locate coherent sources. Tests were conducted inside a fully anechoic chamber. Two speakers were used to produce coherent sound waves and their phases controlled by the "waveplay" function in MATLAB©. The speakers were covered by two cones heavily insulated to minimize sound transmission through openings and side walls. A nylon tube (0.5 m long, 9 mm OD, and 3 mm ID) was used to facilitate plane wave propagation from the apex of the cone to the end of the tube to mimic sound radiation from a point source. The acoustic pressures were measured by a 5 x 5 microphone array with 12.5-mm spacing among individual microphones. Source separation, stand-off distance, and frequency were selected based on limitations of the methods determined from numerical simulations. Results show good correlation with numerical simulations, validating that HELS and modified HELS can be used to locate coherent point sources in violation of spatial resolution guidelines established for Fourier acoustic based NAH and Shannon–Nyquist sampling rates. Similar reconstruction resolution with Fourier acoustics based NAH requires significantly more measurement points and may not visualize the sources as well as HELS does.

1:40

2pSA3. Multi-reference methods for nearfield acoustical holography. Yong-joe Kim (Dept. of Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843-3123, joekim@tamu.edu), Moohyung Lee (Adv. R&D 1, Digital Appliance Div. Digital Media & Commun. Business, Samsung Electron. Co. Ltd., Suwon, Republic of Korea, moohyung.lee@samsung.com), and J. Stuart Bolton (Ray W. Herrick Labs., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031)

Holographic projection requires that sound fields be spatially coherent. That condition can be satisfied if data at all locations on the hologram surface are measured simultaneously. However, more commonly, a scanning procedure is used in which measurements are made on parts of the hologram surface in sequence. In the latter case, it is necessary to record signals from fixed-location reference transducers so that the phase of the pressure in each scan segment can be determined. Further, if the sound field is radiated by statistically incoherent sub-sources, the sound field must be decomposed into coherent partial fields before holographic projection is performed on the individual partial fields. For the partial field decomposition to be successful, the number of reference transducers must be greater than or equal to the number of incoherent sources generating the field. In this presentation, the state-of-the-art of multi-reference procedures applied to nearfield acoustical holography will be described. Requirements for the number and positioning of reference transducers will be given along with procedures to verify that the number of references chosen is sufficient. Further, post-processing of measured data to position virtual reference transducers close to physically meaningful sources and to compensate for source non-stationarity will be discussed.

2:00

2pSA4. Comparison of modal analysis results of laser vibrometry and nearfield acoustical holography measurements of an aluminum plate. Jennifer L. Potter (Structrual Dynam. Lab., The Boeing Co., P.O. Box 3707, Seattle, WA 98124, jennifer.l.potter@boeing.com)

Two methods for non-contact vibration measurements are compared using modal analysis. Laser vibrometry and nearfield acoustical holography are used to measure mode shapes of an aluminum plate excited by a mechanical shaker with random noise. Laser vibrometry provides a direct measurement of the panel surface velocity, while the nearfield acoustical holography measurements are used to calculate the surface velocity indirectly. Both data sets are compared in frequency, damping, and mode shapes. The results are also compared to finite element and theoretical analyzes. Results are presented from all four approaches.

2:20

2pSA5. Surface decomposition method for near-field acoustic holography. Nicolas P. Valdivia (Naval Res. Lab., Code 7130, 4555 Overlook Ave. SW, Washington, DC 20375, nicolas.valdivia@nrl.navy.mil)

Near-field acoustic holography reconstruction of the acoustic field at the surface of an arbitrarily shaped radiating structure from pressure measurements at a nearby conformal surface is obtained from the solution of a boundary integral equation. This integral equation is discretized using the equivalent source method to be transformed into a matrix system that can be solved using iterative regularization methods that counteract the effect of noise on the measurements. This work considers the case when the resultant matrix system is so large that it cannot be explicitly formed and iterative methods of solution cannot be directly implemented. In this case the method of surface decomposition is proposed, where the measurement surface is divided into smaller nonoverlapping subsurfaces. Each subsurface is used to form a smaller matrix system that will be solved and the result joined together to generate a global solution to the original matrix system. We use numerically generated data to study the use of subsurface extensions to increase the continuity of the global solution, the size of the subsurfaces, and the distance of the measurement surface from the surface of the vibrating surface. Finally a vibrating ship hull structure is considered as a physical example to apply and validate our results.

2:40

2pSA6. Transmission loss measurements using a scanned array nearfield acoustical holography technique in an anechoic/ reverberation facility. Part I. Measurement methodology. Alex J. Kremer and Bernard J. Sklanka (Aero/Noise/Propulsion Lab., The Boeing Co., Seattle, WA 98124, alex.j.kremer@boeing.com)

Transmission loss is an important metric when used to assist in the selection of materials for noise control and design. In order to assess and improve the performance of transmission loss measurements, two different techniques were compared: traditional microphone and/or matched pair microphone intensity anechoic chamber chamber-side techniques, and nearfield acoustical holography (NAH) techniques. Intensity and sound power measurements using traditional methods were made on a test panel installed in an anechoic-reverb facility transmission loss window, then repeated using a conformal nearfield microphone array and nearfield acoustical holography processing techniques. In order to process the data with the relative phase between microphones preserved, a set of stationary reference signals was used. A number of spectral preprocessing techniques and reference combinations are presented. The nearfield acoustical holography processing methodology is also presented with an emphasis on data quality and error reduction.

3:00—3:15 Break

3:15

2pSA7. Statistically optimal nearfield acoustical holography in subsonically moving fluid medium. Yong-Joe Kim and Yaying Niu (Dept. of Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843, joekim@tamu.edu)

Statistically optimal nearfield acoustical holography (SONAH) can be used to reconstruct 3-D sound fields by projecting 2-D sound pressure data measured on a small measurement aperture. When the sound pressure data are measured by using a small microphone array mounted on a high-speed vehicle, the mean flow effects of a moving fluid medium should be considered in the SONAH procedure. Here, an improved SONAH procedure is thus proposed that includes the mean flow effects. The backward projection performance of the proposed SONAH procedure is also improved by using a wave number filter to effectively reduce subsonic noise components that are exponentially amplified during the backward projection. For the purpose of validating the proposed SONAH procedure, a monopole simulation at a fluid medium velocity of Mach $= -0.6$ is conducted. The sound fields reconstructed by using the proposed SONAH procedure lead to more accurate sound source locations and radiation patterns than the conventional NAH and SONAH procedures. An experiment with two loudspeakers is performed in a wind tunnel operating at Mach $= -0.12$. The locations and radiation patterns of the two loudspeakers can be successfully identified from the sound fields reconstructed by using the proposed SONAH procedure.

3:35

2pSA8. Transmission loss measurements using a scanned array nearfield acoustical holography system in an anechoic/ reverberation facility. Part II. Comparison of transmission loss methods. Bernard J. Sklanka and Joel R. Tuss (Aero/Noise/Propulsion Lab., The Boeing Co., Seattle, WA 98124, bernard.j.sklanka@boeing.com)

Transmission loss is an important metric used to assist in the selection of materials for noise control and design. In order to assess and improve the performance of transmission loss measurements, two different techniques were compared: traditional microphone and/or matched pair microphone intensity anechoic chamber-side techniques, and nearfield acoustical holography techniques. Intensity and sound power measurements using traditional methods were made on a test panel installed in an anechoic-reverb facility transmission loss window, then repeated using a conformal nearfield microphone array and nearfield acoustical holography processing techniques. The full vectored intensity components were used to assess the radiation characteristics of the test panel. This assessment quantifies the acoustics as panel radiation, flanking path contamination, and boundary effects. Data are presented to qualify the best estimate for transmission loss by rejecting contamination. Also, the dynamic range of the transmission loss measurement is compared between traditional and nearfield acoustical holography techniques with high transmission loss panel samples.

3:55

2pSA9. Statistically optimized near-field acoustical holography applied to a high-power jet. Alan T. Wall, Kent L. Gee, Tracianne B. Neilsen, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, alantwall@gmail.com), Jonathan D. Blotter (Brigham Young Univ., Provo, UT 84602), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC 28801)

The characterization of jet noise can lead to prediction of the radiated sound field and can assist in the development of noise reduction technologies. Thus, the sources and radiation of high-power jets on military aircraft are being investigated. Patch-and-scan array-based holographic measurements were made in the geometric near field of the F-22 Raptor, including more than 6000 measurement points, making this the most extensive near-field measurement of high-power jet noise to date. A set of stationary reference microphones was used to tie together amplitude and phase information between scans by the virtual coherence method. The measurement aperture was then effectively extended through implementation of an analytic continuation procedure and utilization of the rigid ground reflection. Finally, the 3-D near sound field was reconstructed using statistically optimized near-field acoustical holography (SONAH). [Work supported by the Air Force SBIR.]

Contributed Papers

4:15

2pSA10. Cylindrical Fourier near-field acoustical holography applied to a high-power jet. David W. Krueger, Kent L. Gee, Alan T. Wall, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602 dvdkrueger@gmail.com), and Jonathan D. Blotter (Brigham Young Univ., Provo, UT 84602)

Near-field acoustical holography as a method to reconstruct sound fields near military jet aircraft is being investigated. Measurements were recently made on the F-22 Raptor using a linear, ground-based array of microphones at a distance of 11 m from the centerline. Analytic continuation is used to extend the measurement aperture, and, assuming axisymmetry, cylindrical Fourier near-field acoustical holography is used to reconstruct the sound field. Because of the highly directional nature of the source and the effects of wrap-around error from the Fourier transform, the desired reconstruction distance (10–20 m) and the number of points to analytically continue the measurement are directly proportional. Furthermore, given the measurement standoff distance, inverse propagation to the source requires near-complete regularization of the evanescent wave components. [Work supported by Air Force SBIR.]

4:30

2pSA11. Near-field measurement of the scattering characteristics of an object based on Doppler signal filtering. Boris M. Salin and Mikhail B. Salin (Inst. of Appl. Phys., Russian Acad. of Sci., 46 Uljanov St., Nizhny Novgorod 603950, Russia, salin@hydro.appl.sci-nnov.ru)

The scattering characteristics of an object can be measured by insonification with tonal signals and reception by a linear array located in the near field of the object. This paper discusses possible signal processing algorithms designed to measure the angular dependence of the target strength of the scatterer. The measurement is based on (1) moving the object parallel to the array, (2) obtaining 2-D Fourier transform of the received signal in time and space coordinates, (3) the use of a Doppler frequency filter, and (4) selecting the maximum values (as a function of time) of the power spectral density along the trajectory of the moving object in coordinates of frequency and angle. This method gives greater signal to noise ratio than traditional stationary (object not moving) measurement schemes. The possibility of these measurements is estimated using experimental data on the angular and frequency characteristics of reverberation caused by the tonal source. The minimum values of target strength that can be measured in an area with low sea state (i.e., quiet conditions) are determined. [B.M.S. and M.B.S. gratefully acknowledge the generous support of the U.S. Office of Naval Research, ONRG.]

TUESDAY AFTERNOON, 24 MAY 2011 GRAND BALLROOM C, 1:00 TO 4:05 P.M.

Session 2pSC

Speech Communication and Psychological and Physiological Acoustics: Technological, Methodological, and Theoretical Advances in Neuroimaging and Speech Perception

Patricia K. Kuhl, Chair
Univ. of Washington, Dept. of Speech and Hearing Science, Seattle, WA 98195

Chair's Introduction—1:00

Invited Papers

1:05

2pSC1. Cortical rhythms to native and non-native phonetic contrasts in infants and adults. Alexis N. Bosseler (Inst. of Learning and Brain Sci., Univ. of Washington, Fisheries Ctr. Bldg., Box 357988, Seattle, WA 98195-7988, bosseler@u.washington.edu), Samu Taulu (Elekta Oy, Helsinki 09 756 2400, Finland), Toshiaki Imada (Univ. of Washington, Seattle, WA 98195-7988), Elina Pihko (Aalto Univ., Espoo 09 451 4111, Finland), Antti Ahonen (Elekta Oy, Helsinki 09 756 2400, Finland), Jyrki P. Mkelä (Helsinki Univ. Central Hospital, FI-00029 HUS, Finland), and Patricia K. Kuhl (Univ. of Washington, Seattle, WA 98195-7988)

Infants change from universal language-general listeners to language-specific listeners in the first year of life. Studies show that statistical learning affects this change. As learning ensues, the power of statistical information subsides and a reliance on learned categories emerges. We posited that phonetic processing is anchored by two strategies that shift during development: statistical learning during the initial stage before phonetic categories are acquired, and learned categories during adulthood when categories are fully formed. We examined whether the strategy transition could be indexed using cortical brain rhythms (theta) that reflect cognitive effort. Using magnetoencephalography (MEG) we examined theta in response to native and nonnative speech sounds at 6–8 months, 10–12 months, and adulthood. Several advances in MEG analysis were required to test infants. Supporting our hypothesis, the results revealed that theta power increases for frequent stimuli, regardless of language, at 6–8 months. In adults, theta increases were shown for non-native categories regardless of frequency. 12-month-old infants are in transition and show intermediate values. The results indicate that cortical brain rhythms will be valuable in relating speech development to higher cognitive processes.

1:30

2pSC2. Investigating the spatiotemporal neural dynamics of lexico-semantic activity in 12–18 month old infants by combining magnetoencephalography and magnetic resonance imaging. Katherine E. Travis, Matthew K. Leonard, Timothy T. Brown, Donald Hagler, Jr., Anders M. Dale, Jeff L. Elman, and Eric Halgren (Dept. of Neurosci., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92037, ktravis@ucsd.edu)

Knowledge of the neural basis of early language is limited because of the difficulties in assessing functional brain activity during infancy. Consequently, it is unclear whether infants and adults depend on similar underlying neural structures to understand words. Advances in magnetoencephalography combined with magnetic resonance imaging now make it possible to study this population. This

method, dynamic statistical parametric mapping (dSPM), has been extensively validated during language processing in adults. Here, dSPM was used to examine the neural correlates of word understanding in 12–18 month old infants. Specifically, it was discovered that the lexico-semantic neural processes indexed as the adult N400 event-related component are functional during early stages of word learning and depend on similar left frontotemporal brain areas. These findings demonstrate that dSPM can be used to localize functional brain activity associated with language processes in pediatric populations. Moreover, discovery of a fundamental lexico-semantic process present throughout the rapid burst of language acquisition permits this process to be quantified and thus related to behavioral and neural development and their disorders. Together, these findings illustrate the potential for dSPM and other non-invasive neuroimaging techniques to increase current understandings of the neural basis of early language development. [Work supported by the Kavli Institute for Brain and Mind and NIH/NINDA R01 NS018741.]

1:55

2pSC3. Time dependency of cortical specialization and integration in speech perception. Yang Zhang (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu)

Speech perception research has been greatly enriched by technical advancement in neuroimaging. Here electroencephalography (EEG) and magnetoencephalography (MEG) data from infant and adult subjects are reported to demonstrate the feasibility and importance of distributed source estimation and time-frequency analysis in understanding the time-dependent nature of cortical specialization and integration in speech processing. Advantages and limitations of the current analysis approaches for EEG and MEG data are discussed in relation to functional magnetic resonance imaging. The results collectively call for a dynamic time-dependent approach toward proper characterization of the functional neuroanatomy and hemispheric lateralization of spoken language.

2:20

2pSC4. Discussing the advances in the characterization of the cortical involvement in speech and hearing perception using magneto- and electro-encephalography. Adrian K. C. Lee (Dept. of Speech and Hearing and Inst. for Learning and Brain Sci. (I-LABS), Univ. of Washington, Portage Bay Bldg., Rm. 204, Box 357988, Seattle, WA 98195, akclee@uw.edu)

Magneto- and electro-encephalography (M/EEG) are neuroimaging techniques that provide us with the high-temporal resolution particularly suitable to investigate the brain network involved in speech and hearing perception. In many of these studies, data are either interpreted at the sensor level (based on the magnetic/electric field recorded on the scalp) or at the cortical level (after making a judicious choice of the appropriate source modeling technique). There are unique challenges associated with each of these approaches in terms of data processing and, ultimately, making appropriate statistical inferences to illuminate the underlying neural mechanisms associated with the task under investigation. In this talk, the different technical challenges tackled in each of the studies presented in this session will be highlighted. Other methodological advances that are particularly relevant to experiments in the speech and hearing sciences will also be discussed.

Contributed Papers

2:45

2pSC5. The roles of relative distance and gaze in neural processing of spatial demonstratives. James Stevens (Linguist. Program & the Med. School, Univ. of Minnesota, Minneapolis, MN 55455, steve125@umn.edu) and Yang Zhang (Univ. of Minnesota, Minneapolis, MN 55455)

This study collected both behavioral and event-related potential (ERP) data to investigate neural processing of two spatial demonstrative expressions in English, this and that, in social communication. The experimental design employed congruent and incongruent conditions for audiovisual scenarios involving the speaker, the hearer, and the referred-to object. Behavioral data showed that distance determined the demonstrative form only when the hearer shared the speaker's gaze at the object. ERP data indicated signficant congruent versus incongruent differences in the poststimulus window of 525–725 ms for the expected demonstrative form, that, when an object was near the hearer and when the interlocutors shared gaze. Global field power (GFP) analysis showed the effect at an earlier window (200–500 ms). There were also significant GFP effects at 100–200 and 300–775 ms associated with specific spatial configurations in relation to the near-far perceptual distance and shared gaze. Furthermore, standardized low resolution brain electromagnetic tomography (sLORETA) revealed significant activation associated with processing the semantic incongruency effect at the left temporal cortex and the parietal lobe bilaterally. The results demonstrate that the neural mechanisms of spatial deixis processing are sensitive to contextual integration of distance and gaze in social interaction.

3:00

2pSC6. Why do blind listeners use visual cortex for understanding ultra-fast speech? Susanne Dietrich (Dept. of Neurology, Univ. of Tuebingen, D-72076 Tuebingen, Germany, susanne.dietrich@med.uni-tuebingen.de), Ingo Hertrich, and Hermann Ackermann

Individuals with vision impairment of a peripheral, e.g., retinal origin, may learn to understand spoken language at a rate of up to 22 syllables (syl) per second, exceeding by far the maximum performance level of sighted listeners (ca. 8 syl/s). To delineate the brain network subserving this skill, 14 blind individuals (different proficiency in understanding ultra-fast speech) and a sighted control group underwent functional magnetic resonance imaging. In addition to perisylvian cortex and left-hemisphere supplementary motor area (SMA), significant hemodynamic activation of right-hemisphere primary visual cortex (V1), left-hemisphere fusiform gyrus (FG), and bilateral pulvinar emerged during ultra-fast speech perception (16 syl/s) in trained blind people. Dynamic causal modeling analysis revealed skill-dependent intrinsic connections from Heschl's gyrus (right > left) and pulvinar to V1, from V1 to perisylvian areas and SMA as well as from FG to Broca's area. Conceivably, cooperation of right-hemisphere Heschl's gyrus, V1, and left SMA enhances the temporal sequencing of syllable structure of ultra-fast utterances which, in turn, accelerates the segmentation of speech sounds at the level of perisylvian cortex and left-hemisphere FG. The pulvinar—an audiovisual relais station–might support the synchronization of V1 activity with central-auditory processing. [Work supported by the German Research Foundation (DFG; SFB 550/B1) and the Hertie Institute for Clinical Brain Research, Tuebingen.]

3:15

2pSC7. Magnetic brain activity directly reflects syllable onsets and pitch periodicity of a perceived speech signal. Ingo Hertrich, Susanne Dietrich, and Hermann Ackermann (Dept. of Neurology, Univ. of Tuebingen, Hoppe-Seyler-Str. 3, D-72076 Tuebingen, Germany, ingo.hertrich@uni-tuebingen.de)

Continuous speech evokes electrophysiological brain activity following the speech envelope (ENV), resembling the N100 responses to single acoustic events. Using magnetoencephalography (MEG), the present study tested further aspects of evoked activity, considering the positive part of the first derivative of the speech envelope (SYL), emphasizing syllable onsets, and a sinosoidal signal representing pitch periodicity (PIT), obtained by 50–180 Hz bandpass filtering of the rectified speech signal. MEG signals, recorded while the participants ($n = 10$) listened to 4-s portions of natural or formant-synthesized speech at a moderately fast or ultrafast speaking rate, were

cross-correlated with ENV, SYL, or PIT. Using dipole models, the cross-correlated MEG signals were evaluated with respect to latency and amplitude of typical components. As expected, the ENV derivate showed an M100-like response that was stronger over the right as compared to the left hemisphere. Regarding the SYL derivate, a more anterior component could be isolated, in addition to M50/M100-like responses in the auditory system, presumably representing a late frontal component of the M50 deflection. The PIT derivate showed multiple peaks of alternating polarity bound to a central-auditory source. The strength of these MEG components depended on rate and signal type.

3:30

2pSC8. Optimized design and analysis of sparse-sampling functional magnetic resonance imaging experiments of speech and hearing. Tyler K. Perrachione (Dept. of Brain and Cognit. Sci., MIT, 77 Massachussetts Ave., Cambridge, MA 02139, tkp@mit.edu), John D. E. Gabrieli, and Satrajit S. Ghosh (MIT, Cambridge, MA 02139)

Functional magnetic resonance imaging (fMRI) offers an unparalleled opportunity to investigate the brain bases of speech and hearing. However, the high-amplitude (>90 dB) acoustic noise that occurs during MR image acquisition presents a serious obstacle to research on speech perception and production. "Sparse sampling" is an alternate acquisition strategy that mitigates the interference of this acoustic noise by inserting a delay between subsequent image acquisitions, allowing auditory stimulus presentation or speech production during this silent period. Although this technique is routinely employed in auditory fMRI, there has been no empirical attempt to optimize the design of sparse sampling paradigms to maximize detection of whole-brain blood-oxygen level dependent (BOLD) signal. Moreover, the discontinuous nature of the sparse-sampling timeseries has led to the use of analysis models that fail to account for dynamic properties of the hemodynamic response: thus seriously underestimating BOLD signal and limiting the types of cognitive brain activity, sparse sampling is able to detect. We present computational modeling and human neuroimaging experiments that explore the parameter space of sparse sampling experiment design and analysis—including delay duration, stimulation frequency, and hemodynamic response convolution—and offer significantly enhanced detection of brain activity during speech perception tasks versus conventional methods.

3:45—4:05 Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2011 METROPOLITAN A, 1:00 TO 4:00 P.M.

2p TUE. PM

Session 2pUWa

Underwater Acoustics and Acoustical Oceanography: Propagation, Modeling, and Inversion II

Lora J. van Uffelen, Chair
Univ. of Hawaii, Manoa, Ocean and Resources Engineering, 1000 Pope Rd., Honolulu, HI 96822

Contributed Papers

1:00

2pUWa1. Variations of inverted seabed sound speed from different receivers on a horizontal cross-range line array. Lin Wan, Ji-Xun Zhou, and Peter H. Rogers (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, lin.wan@gatech.edu)

In the Shallow Water 2006 experiment conducted on the New Jersey continental shelf, a set of broadband combustive sound source (CSS) signals was measured by a horizontal cross-range line array with 36 receivers. The 230-m-long array was exactly perpendicular to the direction of sound propagation. The modal dispersion characteristics are used to estimate the seabed sound speed by matching the theoretical and measured modal arrival time differences. The measured modal arrival time differences along such a short array vary from one receiver to another at 10–20 km away from the source. This causes variations in the inverted seabed sound speed. This paper analyzes both the variations of inverted seabed sound speed and the horizontal cross-range correlation coefficient as a function of receiver separation. [This work was supported by the Office of Naval Research.]

1:15

2pUWa2. Measurements and interpretation of mesoscale seabed spatial variability. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, cwh10@psu.edu), Peter L. Nielsen (NURC, 19126 La Spezia, Italy), Jan Dettmer, and Stan E. Dosso (Univ. of Victoria, Victoria, Br. Colombia V8W 3P6, Canada)

Acoustic propagation, reverberation, and clutter may be controlled by sediment spatial variability in some environments. Many of the geoacoustic measurement approaches currently employed in the ocean acoustics community provide measurements of sediment properties at either fine scales, less than $O(10^1)$ m (e.g., probes and cores) or at large scales spatially averaged over lateral scales greater than O(10III) m. Relatively little is understood about the mesoscale, defined here as scales from $O(10I–10^3)$ m. Recent wide-angle reflection measurements resolve lateral spatial variability down to a few meters over track lengths of order 10 km and provide early observations of sediment lateral variability at the mesoscale. [Work sponsored by the Office of Naval Research Ocean Acoustics Program and the NATO Undersea Research Centre.]

1:30

2pUWa3. Bayesian inversion of seabed reverberation and scattering data. Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada) and Charles W. Holland (The Penn State Univ., State College, PA 16804-0030)

This paper describes a Bayesian approach to the inversion of ocean acoustic reverberation data for scattering and geoacoustic parameters of the seabed. The seabed is modeled as a sediment layer over a semi-infinite basement. Interface scattering occurs at the (rough) upper and lower boundaries of the sediment layer, and volume scattering occurs within the layer (the scattering mechanisms are considered to be independent and are modeled using perturbation theory and the Born approximation). Unknown parameters include geoacoustic properties (sediment-layer thickness and sound speeds, densities, and attenuations for sediments and basement) and scattering properties (roughnesses and scattering strengths for upper and lower sediment boundaries, and volume scattering strength). One dimensional (1-D) and two-dimensional (2-D) marginal probability distributions are computed from the multidimensional posterior probability density (PPD) using Metropolis–Hastings sampling applied in a principal-component parameter space to provide efficient sampling of correlated parameters. Results indicate that reverberation inversion is a strongly nonlinear inverse problem, with highly multimodal marginal distributions and strong interparameter correlations. Addressing this nonlinearity is of key importance to understanding the information content of reverberation data. [Work funded by the ONR.]

1:45

2pUWa4. Trans-dimensional geoacoustic inversion. Jan Dettmer, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada), and Charles W. Holland (The Penn State Univ., State College PA)

This paper applies a general trans-dimensional Bayesian methodology to geoacoustic inversion. Trans-dimensional inverse problems are a generalization of fixed-dimensional inversion that include the number and type of model parameters as unknowns in the problem. By extending the inversion state space to multiple sub-spaces of different dimensionality, the posterior probability density quantifies the state of knowledge regarding inversion parameters, including effects due to limited knowledge about appropriate parametrization of the environment and error processes. The algorithm applies a reversible-jump Markov chain and the seabed is parametrized as a partition model. The unknown data errors are addressed by including data-error standard deviations as unknowns. Dimensional jumps are implemented with a birth-death methodology that allows transitions between dimensions by adding or removing sediment interfaces to/from the partition model while maintaining unbiased sampling. Trans-dimensional inversion results in an inherently parsimonious solution while partition modeling provides a naturally self-regularizing algorithm based on data information content. Together, this results in environmental estimates that quantify appropriate seabed structure as supported by the data, allowing sharp discontinuities while approximating smooth transitions where needed. This approach applies generally to geoacoustic inversion and is illustrated here for seabed reflection coefficient data collected on the Malta Plateau. [Work supported by ONR.]

2:00

2pUWa5. Hierarchical autoregressive error models in trans-dimensional matched-field geoacoustic inversion. Jan Dettmer and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

This paper applies a general trans-dimensional matched-field inversion algorithm to data with strongly correlated errors, which are addressed by augmenting the forward model with a hierarchical autoregressive error model. This approach accounts for the limited knowledge of the optimal seabed parametrization and of the data-error statistics in the resulting geoacoustic parameter uncertainties. The assumed seabed parametrization influences estimates of parameter values and uncertainties since different parametrizations lead to different ranges of data predictions. The data support for a particular model is often non-unique and trans-dimensional formulations account for this by including an unknown model indexing parameter to consider groups of models (indexed by this parameter) in the posterior results. The hierarchical autoregressive error model accounts for a wide range of possible error correlations with few parameters, with no requirement to explicitly specify a covariance matrix (for which an inverse and determinant must be computed) at every Markov chain step, thereby increasing efficiency. This approach is particularly useful for trans-dimensional inversion since point estimates may not be representative of the state space which spans multiple subspaces of different dimensionalities. The order of the autoregressive process required by the data is determined here by posterior residual-sample examination. [Work supported by ONR.]

2:15—2:45 Break

2:45

2pUWa6. Estimation of the empirical Green's function from ambient noise in the ocean. Ravi Menon, Peter Gerstoft, and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093-0238)

The empirical Green's function obtained from cross-correlations of ambient noise has several applications such as passive imaging, acoustic monitoring, remote sensing, etc. In general, long periods of averaging are necessary to obtain stable estimates. Short time averages suffer from directional biases due to the changing noise field and the dominating effect of loud sources such as ships, in addition to having a low signal to noise ratio. In this paper we analyze the eigenvalues and eigenvectors of the cross-spectral density matrix from short time estimates. Recent results in statistical theory are used to separate the random noise component from meaningful information. By retaining only those eigenvalues and eigenvectors that contribute toward the buildup of Green's function, we can reduce the time required to obtain a stable estimate and achieve higher signal to noise ratios. Results are presented from the SW06 data.

3:00

2pUWa7. Noise interferometry, in the geometric limit, in a multipathing environment. Michael G. Brown (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149, mbrown@rsmas.miami.edu)

It is now well established that under commonly encountered conditions a good approximation to the transient Green's function $G(\mathbf{x}_a|\mathbf{x}_b,t)$ between points $\mathbf{x}_a$ and $\mathbf{x}_b$ can be estimated by taking the time derivative of the correlation function $C_{ab}(t)$ of records of ambient noise measured at locations $\mathbf{x}_a$ and $\mathbf{x}_b$. We consider the relationship between $C_{ab}(t)$ and $G(\mathbf{x}_a|\mathbf{x}_b,t)$ in a multipathing environment in the geometric limit, and we assume that the noise field has a broad bandwidth. Under these conditions $C_{ab}(t)$ for $t > 0$ consists of a superposition of signed step functions and two-sided logarithmic singularities that are delayed in time by the travel times of the rays connecting $\mathbf{x}_a$ and $\mathbf{x}_b$. The reason that C_{ab} has this structure is explained. [Work supported by ONR.]

3:15

2pUWa8. Spatial coherence of acoustic reverberation and its possible use for characterization of the environment. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Earth System Res. Lab./Physical Sci. Div. Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

Cross-correlation function of diffuse noise fields generated by random sound sources distributed on a surface or in a volume contains valuable information about the propagation environment. It was recently demonstrated [Godin *et al.* Geophys. Res. Lett. **37**, L13605 (2010)] that this information can be successfully retrieved and used for passive ocean acoustic tomography. However, long averaging times are typically required for underwater noise to become sufficiently diffuse and for coherent features to become pronounced. In a waveguide with rough boundaries, one might expect that waves scattered by random roughness and waves that would be generated by random sources distributed on the boundaries have similar statistics. This paper investigates the possibility of using rough surface reverberation in an underwater waveguide, instead of the ambient noise, as a source of information about physical properties of the water column. Two-point correlation function of the surface-scattered waves is calculated in the second approximation of the method of small perturbations and compared to the two-point correlation function of diffuse noise. Feasibility is discussed of retrieving deterministic travel times of sound traveling in opposite directions between two receivers from the cross-correlation of the reverberant field recorded by the receivers.

3:30

2pUWa9. Internal waves' role in determining probability distribution of coherent integration time near 133 Hertz and 3709 kilometers in North Pacific Ocean. John Spiesberger (Dept. of Earth and Environ. Sci., Univ. of Pennsylvania, 240 S. 33rd St., Philadelphia, PA 19104-6316, johnsr@sas.upenn.edu)

The hypothesis tested is that internal gravity waves limit the coherent integration time of sound at 3709 km in the Pacific Ocean at 133 Hz and a pulse resolution of 0.06 s. Five days of continuous transmissions at 2 min intervals are examined. The source and receiver are mounted on the bottom of the ocean with timing governed by atomic clocks. Measured variability is only due to fluctuations in the ocean. A model for the propagation of sound through fluctuating internal waves is run without any tuning with data. Excellent resemblance is found between the model and data's probability distributions of integration time up to a day, which is the largest lag explored. The probability that the integration exceeds a day or more is about 0.15. The model under-predicts the probability of occurrence of integration times less than 10 min. However, the overwhelming agreement at longer times supports the conclusion that the standard spectrum of internal waves accurately explains almost all of the distribution of measured integration time. The ocean supports very long coherent integration times because its fluctuations cause the acoustic phasors to fluctuate following non-canceling probability distributions. [Work supported by the Office of Naval Research Contracts N00014-06-C-0031 and N00014-10-C-0480.]

3:45

2pUWa10. Spectra of acoustic travel time at periods shorter than 33 h at basin-scales in the North Pacific Ocean: Horizontal diffusion of upper turning points. John Spiesberger (Dept. of Earth and Environ. Sci., Univ. of Pennsylvania, 240 S 33rd St., Philadelphia, PA 19104-6316, johnsr@sas.upenn.edu)

Previously unexplained fluctuations of acoustic travel time (133 Hz, 0.06 s resolution, 3709 km transmission) at periods between 3.3 and 10 h are likely due to a standard spectrum of internal gravity waves even though the travel times of features arriving 3 to 4 s apart change travel times by nearly the same amount at the same time. This paradox is resolved by appreciating that nearly all acoustic arrivals consist of tens to hundreds of temporally-unresolved arrivals whose upper turning points diffuse horizontally between the normal 50-km periodicity of paths. Similar behavior is obtained from numerical calculations with fluctuations based only on internal waves. The horizontal diffusion of acoustic turning points appears to also explain the presence of non-stationary oscillations at periods up to 33 h. A simple algebraic calculation shows that these longer oscillation periods with amplitudes of about 5 ms are probably due to inertial currents generated by surface winds. If these oscillations are due to inertial currents, these data might be used to study their spatial and temporal scales both in and below the mixed layer. This might be useful for understanding dissipation mechanisms. [Work supported by ONR contract N00014-10-C-0480.]

TUESDAY AFTERNOON, 24 MAY 2011 GRAND BALLROOM D, 1:00 TO 3:30 P.M.

Session 2pUWb

Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Measurement, Characterization, and Mitigation of Underwater Anthropogenic Noise II

Peter H. Dahl, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

George V. Frisk, Cochair
Florida Atlantic Univ., Dept. of Ocean Engineering, 101 N. Beach Rd., Dania Beach, FL 33004-3023

Lisa M. Zurk, Cochair
Portland State Univ., Electrical and Computer Engineering Dept., 1900 S.W. Fourth Ave., Portland, OR 97207

Invited Papers

1:00

2pUWb1. Potential causes of increasing low-frequency ocean noise levels. Michael A. Ainslie (TNO, Stieltjesweg 1, 2628 CK, Delft, The Netherlands)

A simple expression for the global average ambient noise is derived in terms of parameters representing the characteristics of the sound sources, the number of such sources and the propagation properties of the deep ocean, and used to explore possible causes of increasing low frequency noise in the deep ocean. A 5 dB increase in noise level over 35 years, reported by Andrew *et al.* [ARLO **3**, 65–70 (2002)], is explained by the increase in the total number of ocean-going ships and their average tonnage, with no evidence of any significant change in source level during that time. The theoretical effect of increasing transparency to low-frequency sound, which has been proposed as a possible cause of part of this increasing noise [Hester *et al.*, Geophys. Res. Lett. **35**, L19601 (2008)], is less than 1 dB to date, increasing in the 21st century by an amount that depends on the accompanying change in temperature.

1:25

2pUWb2. On the use of noise budgets to assess the effects of offshore wind farms on marine life. James H. Miller, Gopu R. Potty, David S. Casagrande, Kathleen J. Vigness Raposa, Lisa A. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882), Jeffrey A. Nystuen (Univ. of Washington, Seattle, WA 98105), Peter M. Scheifele, and John Greer Clark (Univ. of Cincinnati, Cincinnati, OH 45267)

An ocean noise budget is a list of sources of noise along with their average intensity in a particular frequency band [Frisk *et al.* (2003)]. The budget can be calculated from acoustic data collected by the passive aquatic listener (PAL) systems [Nystuen and Howe (2005); Miller *et al.* (2008)]. In the far field, the average acoustic intensity of plane waves can be computed in 1/3-octave bands over some duration. The assumption is made that the noise in the band at any one time is dominated by a single, identifiable source such as wind, rain, shipping, fish, marine mammals, etc. The duration and instantaneous intensity of the acoustic signal is used to calculate the average. Identification is carried out using the ratios of various spectral levels as outlined in the work of Ma *et al.* [(2005)]. We apply the concept of noise budgets to the assessment of the impact of long term noise from offshore wind turbines on marine life. A developer has proposed to construct more than 200 wind turbines south of Rhode Island. A noise budget has been calculated for the region from data collected on PALs. The addition of the 200 turbines will be incorporated into the budget parametrized on source level.

Contributed Papers

1:50

2pUWb3. Operational wind farm noise and shipping noise compared with estimated zones of audibility for four species of fish. Mathias H. Andersson (Dept. of Zoology, Stockholm Univ., S-106 91 Stockholm, Sweden), Peter Sigray, and Leif K.G. Persson (Dept. of Underwater Res., Swedish Defence Res. Agency, S-164 90 Stockholm, Sweden)

A comparison between different underwater sound sources such as ships and operational wind farms are not straightforward to perform due to their qualitative difference in signal properties. However, in order to study possible effects of noise on fish, this comparison has to be made, and possible cumulative effects investigated. This study shows that a wind farm located in a shallow area in the Baltic Sea will add significant noise levels to the region even though intense shipping activities occur. This is due to the fact that the wind farm produces a strong tonal component at 127 Hz, a frequency otherwise relatively unaffected by shipping noise. The results were obtained by developing a numerical acoustic model of the wind farm taking account of their integrated effect. The model was evaluated against measurements performed at several distances from the wind farm during various weather conditions. Additionally, zones of audibility for cod, herring, salmon, and European eel were estimated.

2:05

2pUWb4. Estimating the acoustic impact of a tidal energy project. Christopher Bassett (Dept. of Mech. Eng., Univ. of Washington, Seattle, WA 98105), Jim Thomson, and Brian Polagye (Univ. of Washington, Seattle, WA 98105)

A pilot-scale tidal hydrokinetic energy project is proposed for Admiralty Inlet, Puget Sound, WA (United States). Quantifying the potential impact on the underwater acoustic environment requires knowledge of pre-installation ambient noise levels, sound production characteristics of tidal turbines, periods for turbine operation, transmission losses in the complex coastal environment, and the response of marine species to received sound levels. Acoustics data obtained during six 3-month deployments at the proposed site are used to characterize ambient noise levels at the site between 20 Hz and 30 kHz. Ancillary data sets reveal shipping and ferry traffic to be the most important sources of anthropogenic noise, and these sources are used to estimate (empirically) transmission losses in the project area. During strong currents, pseudosound from turbulent pressure fluctuations can mask ambient noise at low frequencies while increases in background noise levels at higher frequencies are consistent with bedload transport. Ambient noise due to natural physical processes such as rain, breaking waves, and surf noise are identifiable but secondary to anthropogenic sources. The intensity, frequency, and duration of turbine noise are discussed in the context of existing ambient noise and provide an initial estimate of the impact of the underwater acoustic environment.

2:20

2pUWb5. Noise measurements of a prototype tidal energy turbine. David M. Deveau, Peter J. Stein, Nicholas A. Rotker (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063), Herbert C. Scribner (Ocean Renewable Power Co., LLC., 120 Exchange St., Ste. 508, Portland, ME 04101), and Patrick Edson (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063)

Characterizing the acoustic footprint of power-generating tidal turbines is necessary to assess possible effects of the radiated noise on local marine life. However, as is the case for a wide-range of offshore installations, cohesive and accurate measurement of both the radiated noise of the turbine and the ambient noise conditions is a difficult task. In this case, the usual difficulties with propagation and contaminating noise sources are exacerbated by flow induced noise in the measurement system. Here we present results from noise measurements made of the Ocean Renewable Power Company prototype TidGen™ power generation turbine unit deployed in Cobscook Bay near Lubec and Eastport, Maine. These measurements were made using a drifting measurement system in currents upwards of 6 kn.

2:35

2pUWb6. Airborne noise contributions to the underwater noise sound field. David R. Dall'Osto and Peter H. Dahl (Dept. of Mech. Eng. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Contributions of airborne noise sources to the underwater noise field are the result of two acoustic fields: the transmitted and evanescent. The transmitted field can be represented by only those rays confined to a small cone (about 26 deg) where the reflection coefficient is real-valued. The evanescent field, which arises when rays are totally reflected from the surface, can also contribute to the underwater noise field. Unlike the transmitted field, the evanescent field does not propagate and decays exponentially with depth with a decay rate as a function of frequency. Determining the individual contribution of these two fields to the overall sound field is experimentally difficult to observe. One situation where these two fields can be observed individually occurs when an airplane flies overhead. The Doppler shift associated with tonal propeller noise is dependent on the acoustic path. The frequency separation of the two fields allows for separate analysis of the two fields. Measurements from aircraft (altitude 1000 ft) passing over a buoy equipped with a microphone 3 m above the surface and a hydrophone 2.5 m below the surface will be presented. Numerical simulations are presented along with the experimental observations. [Research supported by the Washington Sea Grant and the U.S. Office of Naval Research.]

2:50—3:00 Break

3:00

2pUWb7. The under-ice soundscape in Great Slave Lake near the City of Yellowknife, Northwest Territories, Canada. Bruce S. Martin (JASCO Appl. Sci., 32 Troop Ave., Halifax, NS B3B 1Z1, Canada, bruce.martin@jasco.com) and Pete A. Cott (Dept. of Fisheries and Oceans, Yellowknife, NT)

The Department of Fisheries and Oceans and JASCO deployed an AMARs sound data recorder in Great Slave Lake near Yellowknife, Northwest Territories, between December 2009 and March 2010. One of the objectives of these recordings was to provide long-term ambient noise measurements in a large frozen lake near a major urban center. A recent study reported spot-measurements of under-ice noise in a subarctic lake from anthropogenic activity raising the sound levels up to 46 dB above ambient, and 10-h average ambient levels that were very low (spectral density levels of ~45 dB *re* 1 μPa). The current Great Slave Lake data provide the opportunity to determine the long-term baseline of under-ice sound levels and how anthropogenic activity, such as air and ice-road traffic, adjacent to the study site affects the ambient levels and how these sounds may impact aquatic biota. Regressions of the ambient levels against wind speed and temperature are also discussed.

3:15

2pUWb8. Long-term ambient noise measurements in the Hudson River. Bruce S. Martin and Jonathan Vallarta (JASCO Appl. Sci., 32 Troop Ave., Halifax, NS B3B 1Z1, Canada, bruce.martin@jasco.com)

Long-term recordings of ambient noise levels in the Hudson River near the Tappan Zee bridge were made between August and November 2010. Twelve recorders were deployed for 2 days in August, and six recorders were deployed for the complete 3 month program. Concurrent logging of meteorological and tidal data was performed, as well as measurements of bridge traffic and rail traffic along the east side of the river. The overall sound levels in the river are comparable to sea-state zero. Analysis of the data has shown that bridge traffic strongly influences the sound levels only within a few hundred feet of the bridge. Tidal and river flow are dominant noise sources near the river's deep channel. Anthropogenic sources, especially pleasure craft, are transient noise sources throughout the river.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

P. Battenberg, Chair, ASC S1
Quest Technologies, Inc., 1060 Corporate Center Drive, Oconomowoc, WI 53066-4828

R. J. Peppin, Vice Chair, ASC S1
Scantek, Inc., 6430 Dobbin Road, #C, Columbia MD 21045

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note — those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 24 May 2011.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 24 MAY 2011 QUEEN ANNE, 3:30 TO 5:00 P.M.

Meeting of Accredited Standards Committee (ASC) S12 Noise

W. J. Murphy, Chair, ASC S12
NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

R. D. Hellweg, Vice Chair, ASC S12
Hellweg Acoustics, 13 Pine Tree Road, Wellesley, MA 02482

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note — that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 24 May 2011.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. On Tuesday and Thursday the meetings will be held starting immediately after the Social Hours at 8:00 p.m. On Wednesday, two technical committees will meet at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday are as follows:

Acoustical Oceanography	Metropolitan A
Engineering Acoustics	Cirrus
Musical Acoustics	Aspen
Physical Acoustics	Willow A
Psychological and Physiological Acoustics	Issaquah
Structural Acoustics and Vibration	Diamond

Session 3aAAa

Architectural Acoustics: Quantitative and Qualitative Effects of Diffusion in Rooms

Jonathan Botts, Cochair
Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180

Peter D'Antonio, Cochair
RPG Diffusor Systems Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Chair's Introduction—7:30

Invited Papers

7:35

3aAAa1. The state of the art in measurement and characterization of scattered sound. Peter D'Antonio and Brian Rife (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

For over 100 years, the acoustical industry has measured and characterized sound absorption. And yet, standard measurements of the random incidence absorption coefficient, according to ISO 354 and ASTM C423-09, still do not provide an accurate answer. Nevertheless, acousticians utilize this inaccurate material property in several formulas, including Sabine, Norris–Eyring, etc., to predict a room property called the reverberation time. All measurements and predictions rely on the concept of a diffuse sound field, which is yet to be fully characterized in the standards. The field of measuring and characterizing scattered sound is in its infancy by comparison, yet in the past 3 decades much progress has been made. Two standards have emerged for measuring scattering (ISO 17497-1) and diffusion (ISO-17497-2) and the current state of the art for both will be reviewed. While much progress has been made in measuring and characterizing scattering surfaces, we still need to develop a relationship between these material metrics and a room property, which we can call diffusivity, as well as defining how to measure and characterize a diffuse sound field. This presentation is intended to be a tutorial on the evolution and current state of the art.

7:55

3aAAa2. Counting local peaks in impulse responses for evaluation of the in-situ diffusion in concert halls. Jin Yong Jeon, Yong Hee Kim (Dept. Architectural Eng., Hanyang Univ., Seoul 133-791, Korea, jyjeon@hanyang.ac.kr), and Michael Vorländer (RWTH Aachen Univ., 52066 Aachen, Germany)

This paper presents methods for identifying reflections in measured room impulse responses using a wavelet transform and counting local peaks. In a performance space, the temporal structure of the impulse response is affected by diffusive wall surfaces. Because wavelet theory assumes the original form of a reflection as a mother wavelet, scattered reflections in the impulse responses can be detected through the continuous wavelet transform. From scale model testing, it is found that the summation of wavelet coefficients increases with diffuser installation. For practical application, a method on counting local peaks at −20 dB after the direct sound is suggested as number of peaks (Np). In measured results, Np shows good agreement with wavelet coefficients. Robustness and effectiveness of Np are discussed for *in-situ* diffusivity evaluation.

8:15

3aAAa3. A review of mixing time estimators. Guillaume Defrance (Institut JLRA-Lam team, Univ. Pierre et Marie Curie (UPMC-Paris 6), 11 rue de Lourmel, 75015 Paris, Fance, defrance.all@gmail.com)

In 1967, Joyce introduced the concept of dynamical systems in the room acoustics community. Based on statistical physics from the early 20th century, Joyce assumed large halls to be ergodic and mixing. Since that time, the hypothesis of mixing rooms has been accepted, but only a few studies have focused on the experimental validation of such an assumption. In 1982, Polack proposed a heuristic formulation of the mixing time, based on perceptual criteria. After that time, the sound field is, theoretically, uniform throughout the room, i.e., the density of sound rays must be homogeneous and isotropic at any time in the phase space. Many acousticians have related the mixing time to the beginning of the diffuse sound field, also called the late reverberation. In a recent work, we have investigated two different approaches for experimentally estimating the mixing time from measured impulse responses. We present these two estimators and explain our results. In particular, we show that the relationship between mixing and late reverberation is not straightforward. Finally, we conclude that mixing may occur later after the beginning of late reverberation. Consequently, instead of *mixing time*, we propose the term of *cross-over time*.

8:35

3aAAa4. Is mixing the source of diffusion? Jean-Dominique Polack (LAM/IJLRA, UPMC/CNRS/Ministre de la Culture, 11 rue de Lourmel, F-75015 Paris, France jean-dominique.polack@upmc.fr)

In a recent Ph.D. thesis, Defrance attempted to attain the essence of diffusion by tracking the so-called mixing time, that is, the time after which sound fields are homogeneous and isotropic. What he achieved, though, was only sparking a discussion between acousticians and physicists on the nature of mixing. Traditionally, mixing is defined in dynamics theory as the process that randomizes the positions of particles after they have been launched in a restricted area of the space phase. Thus, the process should be statistical, and evenly distributed through space: random distribution should be achieved everywhere at the same time. However, the data of Defrance do not confirm this theory. They show that impulse responses become random shortly after the arrival of the direct sound. In other words, the randomization process is position dependent, which is not compatible with mixing. This confirms a recent declaration by Cédric Villani, Field Medal 2010, that mixing cannot explain diffusion, because it is too slow a process. This paper closes with a review of several other mechanisms contending for solving the diffusion puzzle.

8:55

3aAAa5. A theoretical framework for quantitatively characterizing sound field diffusion and sound energy decay curves based on the scattering coefficients and absorption coefficients of walls. Toshiki Hanyu (Dept. of Construction, Junior College, Nihon Univ., 7-24-1 Narashinodai, Funabashi, Chiba 274-8501, Japan, hanyu@arch.jcn.nihon-u.ac.jp)

A theoretical framework for characterizing sound field diffusion based on the scattering coefficients and absorption coefficients of walls was developed. The concepts of the equivalent scattering area, equivalent scatter reflection area, average scattering coefficient, and average scatter reflection coefficient are introduced in order to express each room's capability of scattering. Using these concepts and the mean free path, the scatter-to-absorption ratio, mean scatter time, and the diffusion time are defined in order to evaluate the degree of diffusion of a space. Furthermore a theoretical model for characterizing sound energy decay curves in a non-diffused sound field based on the above mentioned concepts was also developed. First a reverberation model for a room which consists of perfect specular reflective walls was examined. Second the scattering coefficient was introduced to the above reverberation model in order to consider the effects of the sound scattering from the walls on the energy decay curves. Using this theoretical model, a non-linear energy decay curve in a non-diffused sound field, such as a room which has uneven absorption, can be calculated considering the effect of the walls' scattering. The results of computer simulations supported the theoretical framework that the ideas presented here are basically valid.

9:15

3aAAa6. Perception of scattering coefficient in auralized concert halls. Renzo Vitale, Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, D-52066 Aachen, Germany, rvi@akustik.rwth-aachen.de), and José Augustín Garrido Alcázar (Univ. of Granada, Granada, Spain)

The scattering coefficient has become a determinant parameter in room acoustic software because it enhances the precision of simulation results. Although there is a wide agreement on the necessity of using scattering coefficients for characterizing surfaces and pattern elements within an enclosed space such as a concert hall, there is still a lack of knowledge on the influence of this on the human perception. The primary purpose of this study is to investigate how changes in scattering coefficients of diffusing surfaces could possibly affect the perception of music among the audience in concert halls. A simplified shoebox-like geometry was used to generate binaural room impulse responses under various scattering conditions for different seats. Computer simulation results were convolved with sound examples, which were properly selected as representative of typical music textures. Results from listening tests will be shown and discussed.

9:35

3aAAa7. Understanding the perceptual effects of diffuser application in rooms. Philip W. Robinson, Jonas Braasch, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, philrob22@gmail.com)

Acoustic diffusers have been applied in rooms for echo elimination and reduction of spatial variation, but perceptual effects are not fully understood. At worst, diffusive surfaces have been blamed for tonal distortion and coloration, while other studies have found high correlation between preferred concert halls and diffusive interior surfaces. Evaluating perceptual effects of diffusive surfaces presents several unique challenges. While many geometric acoustic modeling programs incorporate spatial scattering algorithms which produce perceivable differences in auralization results and may accurately reproduce acoustic parameters of real halls, these differences are not necessarily representative of the perceptual effects of diffuser application in rooms. Accurate investigations of diffusion can be conducted using real or scale model room measurements, but equalization must be performed to isolate the independent effects of scattering and absorption. This paper presents experimental methods to objectively gauge the perceptual effect of scattering surfaces.

9:55—10:05 Break

10:05

3aAAa8. A multi-faceted study of sound diffusing elements in an auditorium. Timothy Gulsrud (Kirkegaard Assoc., 954 Pearl St., Boulder, Co. 80302, tgulsrud@kirkegaard.com), Arthur Van der Harten, Andrew Kiel, and Larry Kirkegaard (Kirkegaard Assoc., Chicago, IL 60607)

Sound diffusing elements are often provided in auditoria as a means of providing diffuse sound reflections intended to improve listening conditions. Some recent consulting experience, however, has revealed that sound diffusing elements can sometimes have unintended consequences that detract from, rather than improve, listening conditions in an auditorium. This paper discusses a specific example from Hamer Hall at the Arts Centre, Melbourne, Australia. The influence of large scale sound diffusing elements along the upper sidewalls and ceiling of this hall has been studied subjectively and objectively through critical listening, *in situ* measurements, computer modeling, and scale model measurements. We review the results of this multifaceted study, and discuss the delicate balance between positive and negative impacts of sound diffusion in auditoria.

10:20

3aAAa9. Relationships between diffusion and scattering. Brian Rife and Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, brife@rpginc.com)

ISO 17497-1 and ISO 17497-2 describe two approaches to measure and characterize scattering and diffusion, respectively, from surfaces. This paper will present experimental results on several samples illustrating both coefficients and a comparison between results obtained from each standard. Scattering coefficients determined from ISO 17497-1 and those obtained from the polar responses of ISO 17497-2, using correlation between reflector and scatterer, as well as the ratio of specular energy to total energy, will be presented. A discussion of the significance of ISO 17497-1 for 1D extruded shapes and samples containing absorption will also be given. Lastly, a theoretical proposal to relate the diffusion and scattering coefficients will be presented, in which a bound on the diffusion coefficient for a given scattering coefficient is derived.

10:35

3aAAa10. Numerical prediction of sound diffusion from surfaces with fractal geometry. Erik Snow and David T. Bradley (Dept. of Phys. + Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604-0745)

Sound diffusive and scattering surfaces can be implemented in architectural spaces to improve the acoustical qualities of the space, particularly by attenuating the effects of harsh reflections and by producing a more diffuse sound field. These surfaces typically are effective only for a limited range of frequencies, dependent on the scale of the surface geometry. Given the broad frequency range of human hearing, an ideal diffuser would provide scattering across many frequencies. There is a direct relationship between surface roughness size and the wavelength of the scattered sound; therefore, scale-invariant fractal surfaces can be useful in achieving this ideal. In this study, virtual two-dimensional (2-D) fractal surfaces have been generated using the random midpoint displacement (RMD) algorithm, with fractal characteristics in both one and two dimensions. The sound diffusive properties of these surfaces were numerically predicted using the boundary element method (BEM). Several input parameters of the fractal generation algorithm have been varied, such as the number of iterations to which the algorithm is carried out, and the average difference of the random displacements from a mean flat surface. The effects of these parameters on the diffusion coefficient of the surfaces have been predicted. Results and analysis will be presented.

10:50

3aAAa11. Effects of surface scattering and room shape on the correspondance between statistical- and geometrical-acoustics model predictions and real sound fields. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St., S.W., Washington, DC 20024-2302)

Much of room acoustics relies on approximate models of sound. The high-frequency approximation of geometrical acoustics, in which sound is modeled by ensembles of classical phonons, is the basis for many computational modeling techniques. The earliest mathematical model of sound in rooms, given by Sabine's decay equation, is a statistical approximation that assumes homogenous and isotropic (diffuse) sound fields and continuous absorption processes. Sabine's statistical-acoustics model is related to geometrical acoustics as a limiting case for rooms that are ergodic, sufficiently mixing, and weakly absorbing. Various semi-empirical corrections are possible (e.g., Eyring and Kuttruff), though a true first-order correction requires additional information. Surface scattering (as characterized by scattering and diffusion coefficients) and room shape are critical in determining the relationship between statistical- and geometrical-acoustics predictions, as these physical attributes determine key statistical properties of phonon trajectories: path-length distributions, statistical independence of reflections, and the sequence with which possible paths are explored. Along with a theoretical overview of the role of scattering and room shape, this presentation describes several computational investigations into these relationships together with a corresponding series of scale-model experiments in which the results were validated. [Prior work supported by the Bass Foundation and RPG Diffusor Systems, Inc.]

11:05

3aAAa12. Impacts of architectural design variables to diffusion in room acoustics. Sentagi S. Utami and Mojtaba Navvab (Dept. of Architecture, Univ. of Michigan, 2000 Bonisteel Boulevard, Ann Arbor, MI 48105, sentagi@umich.edu)

The claim on the importance of having diffusive surfaces as a room acoustics design solution is based on the arising demand of architectural spaces requiring need of reducing echoes and maintaining sound energy. Realization of the design goal is commonly by introducing diffuser panels as architectural elements. In practice, implementing this design variable alone may not be the solution. This study demonstrates the use of integrated methods to better understand the impact of architectural design variables to room acoustics diffusion. Impacts are characterized by evaluating the relationship of the degree of diffuseness indicated by the coherence of impulse response, room acoustics parameters, and the auditory perception. A system of multi-microphones arrays based on beam forming with acoustical imaging algorithms is utilized for *in-situ* measurements along with computer simulation of recording studio, classrooms, auditoriums, large multi-purpose halls, and arenas. The results of selected scenes are visualized with current computer simulation capabilities for subjective evaluation to obtain real time spatial experience given the auralization capabilities. The integrated method applied in this study and the relationships of indices help in decision making as to what extent the variation of design variables is required to create diffusion that fulfills the expected room acoustics quality.

Session 3aAAb

Architectural Acoustics: Room Acoustics I

Michael R. Yantis, Chair
Sparling, 720 Olive Way, Seattle, WA 98101-1833

Chair's Introduction—11:30

Contributed Papers

11:35

3aAAb1. Sound strength parameter G in concert hall design. Leo Beranek (776 Boylston St., Apt. E10A, Boston, MA 02199)

The parameter "Strength of Sound G" is closely related to loudness. Its magnitude is dependent, inversely, on the total sound absorption in a room. By comparison, the reverberation time (RT) is both inversely related to the total sound absorption in a hall and directly related to its cubic volume. Hence, G and RT in combination are vital in planning the acoustics of a concert hall. A newly proposed "bass index" is directly related to the loudness of the bass sound and equals the value of G at 125 Hz minus its value at mid-frequencies. Listener envelopment (LEV) is shown for most halls to be directly related to the mid-frequency value of G. The broadening of sound, i.e., apparent source width (ASW), is given by effective spatial impression (ESI) which is determined from the combined effect of early lateral reflections and strength G. The importance of considering G when planning the acoustics of concert halls is emphasized.

11:50

3aAAb2. A rule based approach to the Design of Auditoria. David Oldham (School of Architecture, Univ. of Liverpool, Liverpool L69 3BX, United Kingdom)

The past 100 years has witnessed a considerable amount of research aimed at improving our understanding of the way in which sound behaves in auditoria. However, there remains a need for transferring much of this knowledge into the design process. Ideally, the designer should be able to access this knowledge at all stages of design and thus to ensure the achievement of good acoustical conditions in the finished project. In this paper, a rule based approached to room acoustic design, implemented on a personal computer, is described. All rules are related to relevant research findings and presented in such a way as to leave the designer with sufficient leeway to enable him or her to explore a range of innovative solutions.

WEDNESDAY MORNING, 25 MAY 2011 ISSAQUAH, 8:00 TO 10:15 A.M.

Session 3aABa

Animal Bioacoustics: Memorial Session in Honor of Ronald Schusterman and David Kastak III

Roger M. Gentry, Cochair
ProScience Consulting LLC, 22331 Mt. Ephraim Rd., Dickerson, MD 20842

Robert Gisiner, Cochair
OPNAV N45, Navy Energy and Environmental Readiness Div., Arlington, VA 22202

Patrick W. Moore, Cochair
National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106

Chair's Introduction—8:00

Invited Papers

8:05

3aABa1. New approaches for studying the perception of vocal signals in otariid pinnipeds. Jason Mulsow and James J. Finneran (US Navy Marine Mammal Program, SSC Pacific, Code 71510, 53560 Hull St., San Diego, CA 92152, jason.mulsow@gmail.com)

The pioneering work of Ronald Schusterman and David Kastak considered the acoustic signaling of pinnipeds from the perspectives of both sender and receiver. Their work often highlighted the otariid pinnipeds (sea lions and fur seals), which are known for their frequent, loud aerial vocalizations. Numerous aerial vocalizations have been described for otariids, and research has demonstrated that a number of call characteristics are sufficient for individual and species recognition. In contrast to the marked interspecific variability found in these vocalizations, laboratory studies with pure-tone stimuli suggest that the aerial hearing capabilities of otariids are nearly identical among species. This is a surprising result, as many components in the calls of otariids exist at frequencies well below the range of best hearing sensitivity. Future studies can potentially examine the relevance of frequency composition and other vocalization parameters using psychophysical methods that include complex acoustic stimuli and an analysis of subject response latencies. Such methods have proven useful for studying the manner in which birds perceptually categorize vocalizations [Dooling *et al.*, J. Comp. Psychol. **101**, 367–381 (1987)] and they may provide novel tools for investigating the link between vocalization structure and auditory perception in otariids. [Work supported by ONR.]

8:25

3aABa2. Why do Weddell seals shout? Jack Terhune (Dept. of Biology, Univ. of New Brunswick, 100 Tucker Park Rd., Saint John, NB, E2L 4L5, Canada, terhune@unb.ca)

Source levels (SLs) of Weddell seal (*Leptonychotes weddellii*) underwater calls near Mawson, Antarctica, were determined using a two hydrophone array. SLs were 161 ± 10 dB *re* 1 μPa m (range 135–179, $n = 280$). SLs from 0.1–6 kHz varied little with frequency ($r^2 = 0.02$, $t = -2.46$, $P = 0.01$, $n = 251$). One-sixth octave ambient noise levels (ANLs) from 0.1–6 kHz were measured on low ($n = 1$), medium ($n = 7$), and high ($n = 7$) noise level days. The ANLs were flat (0.1–6 kHz) and the mean 1/6 octave ANLs were 77 ± 2.8, 96 ± 6.5, and 110 ± 6.1 dB *re* 1 μPa. SLs were randomly paired against ANLs in a Monte Carlo ($n = 100\,000$) model to calculate the seal communication ranges (*m*), assuming spherical spreading and received levels 20 dB above threshold. The mean communication ranges for low, medium, and high ANLs were 2806 ± 2718, 428 ± 662, and 83 ± 124 m, respectively (median distances were 2006, 205, and 43*m*). The distributions were highly skewed toward the shorter distances. The high amplitude calls of Weddell seals may have evolved to facilitate local communication under noisy conditions rather than for very long range purposes.

8:45

3aABa3. Vocalization source levels of adult male northern elephant seals (Mirounga angustirostris). Brandon L. Southall (SEA, Inc., 9099 Soquel Dr., Ste. 8, Aptos, CA 95003), Stephen Insley (Univ. of California, Santa Cruz, OCA.), Marla Holt (Northwest Fisheries Sci. Ctr.), and Colleen Reichmuth (Univ. of California, Santa Cruz, CA)

Aerial vocalization source levels were obtained for adult male northern elephant seals from 1999–2010. Vocalizations from known individuals (marked within years) were selected from three breeding seasons (1999–2000, 2004–2005, and 2009–2010) evenly spaced during this interval so that it is unlikely animals were resampled across seasons. Sound pressure levels of calls were measured on-axis (0-deg orientation) at 1-m range using Brel and KJær 2203 and 2250 precision sound level meters. Calls almost always occurred in series, as previously described; the maximum received level for any pulse in each series is used as the source level. Source levels are reported for individuals with at least four complete bouts. All sex/age classes were measured; the adult male data will be discussed and compared with other mammals in terms of rms levels (two temporal weighting functions) and peak sound pressure levels. Results indicate that calls almost always exceed 100 dB rms (re: 20 μPa) with peak levels regularly exceeding 120 dB. These loud sounds might suggest long-range communication, but considering hearing data Ron Schusterman and Dave Kastak first collected on this species and environmental noise levels in their rookeries, intense signals are likely required even for the small ranges over which they apparently function. [Funding provided by ONR.]

9:05

3aABa4. Vocal recognition of individuals versus relative dominance rank among breeding male northern elephant seals (*Mirounga angustirostris*). Stephen J. Insley (Dept. of Biology, Univ. of Victoria, P.O. Box 3020, Station CSC, Victoria, BC V8W 3N5, Canada, sinsley@uvic.ca), Marla M. Holt (Northwest Fisheries Sci. Ctr., Seattle, WA 98112), and Brandon L. Southall (Southall Environ. Assoc., Santa Cruz, CA 95060)

Whether an animal truly recognizes an individual or a simple rule-based category (e.g., neighbor or offspring) has important behavioral and evolutionary implications such as the accuracy of social reciprocity. Many tests of individual recognition have focused on neighboring territorial males (“dear enemy” or “neighbor-stranger” recognition). Unfortunately the static territorial context of these tests, mostly with male songbirds, opens them to the criticism of being merely associative habituation. More dynamic mating assemblages, such as leks where vocally advertising animals encounter numerous others, are a potentially rich and largely untested alternative. The female defense polygyny practiced by male northern elephant seals during terrestrial breeding is such a dynamic system. To examine whether elephant seals were recognizing individuals or dominance categories we conducted a total of 53 playback experiments to 18 males at Año Nuevo State Reserve. Each playback was a series of threat calls assigned to four dominance conditions relative to the subject. Dominance was based on the outcomes of interactions among contesting male dyads. Responses were measured using three assays *in situ* and from video records of each experiment. Results thus far are consistent with the males not recognizing individuals but instead recognizing and responding appropriately to relative rank.

9:25

3aABa5. Directionality of male northern elephant seal (*Mirounga angustirostris*) threat calls and how it influences receiver behavior. Marla M. Holt (Marine Mammal Ecology Team, NOAA Fisheries Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. East, Seattle, WA 98112, marla.holt@noaa.gov), Brandon L. Southall (Southall Environ. Assoc. Inc., Santa Cruz, CA 95060), Stephen J. Insley, and Ronald J. Schusterman (Univ. of California, Santa Cruz, CA 95064)

Many animal sounds are directional in which the sound energy is focused in a direction that depends on the signaler’s orientation. In the 1970s, Ron Schusterman quantitatively showed this in barking California sea lions and dogs. Several investigators have suggested ways that such features might be particularly useful among individuals in acoustic communication networks. However, only a few have tested such hypotheses experimentally and even fewer have investigated how directional signals affect receiver behavior. In this study, we measured directivity patterns of male northern elephant seal threat calls and used an acoustic playback approach to determine how call directionality influenced the responses of male seals in reproductive competition. We collected data on adult and older subadult seals on a breeding rookery (Ano Nuevo State Park) over three field seasons. Threat calls had substantial directionality, particularly at frequencies above 1 kHz and responses to playbacks depended on call directivity patterns. Males moved farther away from the playback source when it simulated a caller oriented toward them compared to when playbacks simulated a caller oriented away from them. These results suggest that threat call directionality provides meaningful information about the auditory scene and spatial orientation of male elephant seals in reproductive competition.

Contributed Papers

9:45

3aABa6. Source characteristics of the underwater knocking displays of a male Pacific walrus (*Odobenus rosmarus divergens*). William R. Hughes (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, CA 95064), Colleen Reichmuth (Univ. of California, Santa Cruz, CA 95060), Jason L. Mulsow (U.S. Navy Marine Mammal Program, SSC Pacific, San Diego, CA 92152), and Ole Næsbye Larsen (Univ. of Southern Denmark, Odense DK-5230, Denmark)

Walruses breed in winter at high latitudes in conditions that make close-range observations difficult. Males are known to produce complex underwater songs that can extend over multiple days and propagate over several kilometers. These acoustic displays are comprised of highly rhythmic sharp "knocks" punctuated by occasional metallic "bells." The source characteristics of the knocking sounds that were regularly emitted by a male walrus raised in captivity were examined. Knocks were produced as single 20 ms pulses, or as doublets and triplets, and were typically repeated at rates of 0.8/s to 1.2/s. These were loud sounds with greater bandwidth than previously reported: mean source levels were 186 dB pk-pk *re* 1 μPa at 1 m (range 161–196) with maximum frequency >24 kHz. Production of each knock was associated with visible impulsive movement of the forehead. During rut, this walrus had difficulty inhibiting sound production and would often continue to emit knocks in air during haul-out and even while eating, suggesting an endogenous component to this behavior. A strong correlation between his seasonal testosterone levels and the persistence of knocking displays was confirmed. Captive research provides unique access to acoustic and reproductive behavior that is presently impossible to study in wild walruses.

10:00

3aABa7. Automatic localization of individual Hawaiian minke whales from boing vocalizations. Stephen W. Martin (Biosciences Div., SPAWAR Systems Ctr. Pacific, 53366 Front St., San Diego, CA 92152, steve.w.martin@navy.mil), Tom Norris (Bio-Waves Inc., 517 Cornish Dr., Encinitas, CA 92024), Eva-Marie Nosal (Univ. of Hawaii, 2540 Dole St., Honolulu, HI 96822), David K. Mellinger (Oregon St. Univ., 2030 SE Marine Sci. Dr., Newport, OR 97330), Ronald P. Morrissey, and Susan Jarvis (Naval Undersea Warfare Ctr., Bldg. 1351, Newport, RI 02841)

A method is described to automatically localize Hawaiian minke whales from their boing vocalizations. Recorded passive acoustic data from 15 deep water seafloor mounted hydrophones at the Pacific Missile Range Facility is utilized. A critical step is the automatic association of the same vocalization as received by the widely spaced hydrophones. The peak frequency of the vocalization in the detection bandwidth is shown to aid in the association process. Temporal integration of standard time difference of arrival localizations reduces erroneous automatic localizations, which occur for a variety of reasons. A case study of a 2009 minke visual sighting by a field team, which was facilitated by radioing near real-time location information from shore is described. The peak frequency feature (PFF) has unexpectedly been observed to be very stable for what is believed to be the sighted individual over a 6 hour time period (n = 57, $\overline{\mathrm{PFF}}$=1384.0Hz, σ = 1.78 Hz). When the minke ceased vocalizing at 13:44 HST, no vocalizations at this frequency were again observed until 18:30 HST. This suggests a possible acoustic feature unique to individual animals with potential anatomical relationship with the sound production mechanism.

WEDNESDAY MORNING, 25 MAY 2011 — ISSAQUAH, 10:30 TO 11:45 A.M.

Session 3aABb

Animal Bioacoustics: General Topics in Bat Acoustics

James A. Simmons, Chair
Brown Univ., Neuroscience Dept., Providence, RI 02912

Contributed Papers

10:30

3aABb1. Echolocation of fluttering insects by using the frequency modulated sound. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., 2-1-1 Tenjinzawa, Sendai 981-3193, Japan, matsuo@cs.tohoku-gakuin.ac.jp) and Takuma Takanashi (Forestry and Forest Products Res. Inst., Tsukuba 305-8687, Japan)

Using the echolocation, bats can capture insects in real 3-D space. The echoes from the insect were changed with the wing beats and its orientation. In the case of emitting the constant-frequency (CF) sound, the wing beats could be estimated from the amplitude modulation and frequency modulation (FM) dependent on the Doppler-shift. In this study, the echoes were measured from several kinds of insects when both the CF and FM sounds were intermittently emitted from the ultrasonic loudspeaker. At the same time, the movements of the wing were measured by the high speed camera. The impulse responses and time-frequency pattern were computed by using the cross-correlation function and the convolution of the chirplet filters, respectively. It was examined that these patterns were related to its orientation and the wing beats, that is, the change of wing positions along the time axis. [Work supported by the Research and Development Program for New Bio-industry Initiatives.]

10:45

3aABb2. Developmental change in ultrasonic echolocation sounds of Japanese echolocating bats, *Pipistrellus abramus*. Shizuko Hiryu and Hiroshi Riquimaroux (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani Tatara, Kyotanabe 610-0321, Japan)

The development of vocalization during the first post-natal month in *Pipistrellus abramus* was studied. Vocalizations were recorded from each pup (five pups from two mothers; captive-born and captive-raised in a laboratory) everyday when isolated from its mother. The sounds produced by pups on the day of birth were categorized into a long isolation call and seemingly an echolocation precursor call (EP call). The terminal frequencies of the fundamental (TF) was 19.3 ± 1.9 kHz (n = 98), indicating that TF ranges of the second harmonic produced by newborn bats almost corre-

sponded to the fundamental TF of adult pipistrelle (40~45 kHz). The bats over 2 weeks of age mainly produced EP calls whose characteristics were changed with age. (1) increased TF, (2) decreased pulse duration, (3) increased bandwidth, and (4) constant inter-pulse interval. The TF at 4 weeks old showed no significant difference from TF of adult pipistrelle (41.6 ± 2.0 kHz, $P = 0.8$). We recorded echolocation pulses of pups (31 days old) during their first flight in a flight chamber. The results show that (1) practice flights appeared to be unnecessary to modify IPI or duration with a distance and (2) the bats emitted paired pulses (strobe groups) in 2 days after the first flight. [Work supported by JSPS and ONR.]

11:00

3aABb3. Shape space analysis of structures in bat biosonar. Rolf Müller, Lvyin Cai (Dept. of Mech. Eng., Virginia Tech, IALR, 150 Slayton Ave., Danville, VA 24540, rolf.mueller@vt.edu), Cindy Grimm (Washington Univ. in St. Louis, One Brookings Dr., St. Louis, MO 63130), and Washington Mio (Florida State Univ., Tallahassee, FL 32306)

Sound diffracting shapes such as the noseleaves and outer ears are prominent features of the biosonar systems in bats. Across different bat species, a remarkable diversity exists in the shapes of these structures. Interspecific differences between shapes are found in global shape properties such as size and aspect ratio as well as in local features such as flaps, ridges, and grooves. Because of the irregular shapes of these biological structures, the relationship between physical shape and acoustic function across all their different shapes is difficult to comprehend. As a first step to address this problem, an attempt is made to describe the different pinna geometries in a shape space intended to capture the salient trends in biological variability. Descriptions of variability require an alignment of shapes. This can be achieved by virtue of invariant features within a low spatial frequency band. Here, modal analysis is used for this purpose. The shapes can be transformed by mapping each point into a new space defined by the amplitudes of the shape's vibration eigenmodes. The acoustic properties of the shapes can be changed significantly by local features. Hence, local shape descriptors are used to quantify these features and establish correspondences across different species.

11:15

3aABb4. Acoustic analysis of bat eigenears. Jianguo Ma, Lin Feng (School of Phys., Shandong Univ., Shanda South Rd. 27, 250100 Jinan, China), and Rolf Müller (Virginia Tech, IALR, 150 Slayton Ave., Danville, VA 24540, rolf.mueller@vt.edu)

The shape diversity in bat pinnae can be explored using principal component analysis. For this purpose, a vector-space description of the shapes is obtained by transforming the positions on the inner and outer pinna surfaces into cylindrical coordinates. This yields two functions of two independent variables: inner and outer surface radius as a function of angle and height. Pinnae from different species were aligned at the centers of gravity of these functions. The domain of the two functions can be discretized and the corresponding function values arranged to form a vector. In order to explore the acoustic properties of the resulting eigenvectors ("eigenears"), the center of the eigenear coordinate system was positioned at the average pinna, which is similar to an obliquely truncated cone and has a simple beampattern dominated by a single, upward-pointing mainlobe. The first eigenear is similar in shape to the average ear, subtracting or adding it to the first eigenear results in a wider or narrower cone. In the vicinity of the average ear, this operation will likewise result in a wider or narrower mainlobe. The second eigenear has a symmetry-breaking effect on both the pinna shape and its beampattern. Higher eigenears will also be explored.

11:30

3aABb5. Computational model of a bioinspired broadband receiver for sonar clutter reduction. Jason E. Gaudette (Ctr. for Biomedical Eng., Brown Univ., Providence, RI 02912, jason_gaudette@brown.edu), Jeffrey M. Knowles, Jonathan R. Barchi, and James A. Simmons (Brown Univ., Providence, RI 02912)

Echolocating bats have the ability to seamlessly navigate through dense foliage and other obstacles at flight velocity using only the information available in acoustic returns. The spatial resolution required to perform this feature cannot be explained by conventional beamforming and pulse design techniques. We describe a biologically inspired broadband sonar receiver that mimics parallel neural processing by echolocating bats to suppress clutter in complex acoustic environments. These results are incorporated into an improved version of the spectrogram correlation and transformation (SCAT) receiver by replacing the original spectrogram transformation block with a process that translates both harmonic coherence and spectral interference patterns into estimates of echo-delay separations for closely spaced echo highlights. The model treats simple target echoes as highlight reconstructions, and broader lowpass filtered echoes—characteristic of most off-axis clutter—as numerous overlapping poorly defined shapes. The monaural SCAT model for range-only resolution is also expanded to a binaural, 2-D model for high-resolution sonar imaging in the range-azimuth plane. Receiver performance is compared with conventional array processing methods through acoustic simulation and sonar image reconstruction amidst dense clutter. [Work supported by the ONR and internal investments by the NUWC, Division Newport.]

Session 3aAO

Acoustical Oceanography and Underwater Acoustics: Integrating Ocean and Acoustic Observations With Models

James F. Lynch, Chair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543-1053

Chair's Introduction—7:55

Invited Papers

8:00

3aAO1. Geoacoustical oceanography: Integrating geology with ocean acoustics. N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada)

Knowledge of the structure and material properties of the ocean bottom is essential for modeling sound propagation in shallow and deep water environments. Significant progress has been made in the past decade in developing inversion methods for estimating geoacoustic profiles that are used to account for the interaction of sound with the ocean bottom in numerical calculations of the sound field in the water. These methods rely heavily on ground truth information about the material properties and structure of the sediment to define realistic prior estimates of the bottom. Ground truth is obtained by various different geological and geophysical techniques such as sediment grab samples and cores, and chirp sonar and high resolution seismic surveys. This paper uses examples from the Shallow Water '06 experiment to illustrate the interplay between geological and geophysical information and acoustic data in generating geoacoustic profiles that are effective for modeling sound propagation in shallow water. One example demonstrates the use of prior information in Bayesian matched field inversion, and another example shows the combined use of chirp sonar survey data and modal wavenumber estimation to generate a geoacoustic map of the region.

8:20

3aAO2. Sound and biological models. Michael B. Porter (3366 N. Torrey Pines Ct., Ste. 310, La Jolla CA 92037, mikeporter@hlsresearch.com), Dorian S. Houser (Biomimetica, 7951 Shantung Dr. Santee, CA 92071), David C. Mountain (Boston Univ., 44 Cummington St., Boston, MA 02215), and Martin Siderius (Portland State Univ., Portland, OR 97201)

We consider the coupling of biological and acoustic models in the ocean (marine bioacoustics). A casual look at journal publications and meeting programs over the last 10 years shows the extraordinary growth in research. Consider these five examples: animats, which are virtual marine mammals, swim in simulated oceans, acting as acoustic dosimeters to simulate the exposure of real animals in the ocean. The animats themselves respond to the sound field, avoiding unpleasant noises or seeking out sounds that suggest feeding or reproductive opportunities. Meanwhile, the songs, whistles, clicks, rumbles, etc., of living mammals allow a listener to follow their paths, or even to coarsely image the ocean environment. As they move through the sound field they may also disrupt it, scattering energy. Thus we see an intimate coupling between the biological models and the sound field in both virtual and real worlds. This talk will present examples of each of these couplings.

8:40

3aAO3. Study of underwater sound propagation in the continental shelf of the Mid Atlantic Bight with integrated physical oceanographic and acoustic models. Ying-Tsong Lin, Weifeng Gordon Zhang, Timothy F. Duda, James F. Lynch, and Arthur E. Newhall (Dept. of Appl. Ocean Phys. and Engineering, Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation in continental shelf regions is influenced by many physical oceanographic and geological features. With a data-assimilated ocean model, the physical oceanographic (PO) influences can be taken into account in a sound propagation model. This talk presents an integrated modeling framework with the regional ocean modeling system (ROMS) and parabolic-equation (PE) based sound propagation models. Technical issues on the model integration will be addressed, and the final product is implemented in the continental shelf of the Mid Atlantic Bight. The water column variability caused by shelfbreak fronts and linear internal waves is captured in the ocean model and directed into the acoustic model. The sound propagation effects by realistic shelfbreak fronts, frontal intrusions, and linear internal waves can be studied. In considering the geological influences on both PO and acoustic fields, this integrated model is implemented over the Hudson Canyon area. Three dimensional sound focusing due to the canyon topography is observed, and the temporal variability of such focusing patterns caused by the changes in the PO field is also quantified. In conclusion, by combining the state-of-the-art ocean and acoustic models, numerical simulations of underwater sound propagation are moving forward and toward more reality.

9:00

3aAO4. Shallow-water acoustic studies with regional ocean models. Timothy F. Duda, Ying-Tsong Lin, Weifeng Gordon Zhang (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, tduda@whoi.edu), Aurelian L. Ponte, Bruce D. Cornuelle (UCSD, La Jolla, CA 92109-0230), and Pierre F. J. Lermusiaux (MIT, Cambridge, MA 02139)

The value of acoustic propagation modeling using water-column sound-speed fields from computational physical oceanographic models is well established. With a proper investment in measurements and models it is now possible to make predictions of real-time and future acoustic effects of mesoscale eddies and other structures. This capability depends on properly incorporating data from satellite and *in situ* instruments into data-assimilative models. Many challenges remain, however. First, mesoscale feature uncertainties are high when data are scarce. Second, models do not fully handle gravity waves (internal and surface) and submesoscale features. The present utility of models is high despite these remaining challenges. For example, models running open-loop or with partial boundary condition and internal constraints will generate realistic fields for acoustics studies that have goals other than sound-field prediction. Recent analyzes of the sound propagation effects, at 50–1000 Hz, of internal waves, fronts, and canyons show that realism can improve studies of transmission loss uncertainty and fluctuation characteristics. As examples, internal-wave curvature affects acoustic beam generation and horizontal interference patterns, and internal-tide amplitude and direction affect shallow-water acoustic mode wavenumbers and attenuation parameters. Acoustic fluctuation effects computed with three-dimensional (3D) acoustic modeling through internal-wave permitting ocean models will be presented.

Contributed Papers

9:20

3aAO5. Waveguide invariant analysis for modeling time-frequency striations in a range-dependent environment. Alexander W. Sell and R. Lee Culver (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802)

The waveguide invariant is a useful parameter for understanding the behavior of interference patterns that arise from broadband acoustic sources in shallow water waveguides. These interference patterns often appear in time-frequency plots in the form of striations, whose properties are directly related to the environmental parameters that characterize the waveguide. These parameters include, but are not limited to, the sound speed profile, bathymetry, sediment properties, and source depth. Prior formulations of the waveguide invariant are not fully range- and depth-dependent and thus perform well only in selected environments. This talk will discuss the use of a novel range-dependent waveguide invariant distribution to describe striations seen in acoustic data taken during a 2007 experiment that occurred along the continental shelf of southeast Florida. The performance of this approach and its applicability to range and depth localization will be discussed. [Work supported by ONR Undersea Signal Processing.]

9:35

3aAO6. Acoustic modeling of the southeast Florida continental shelf and slope in three dimensions. Megan S. Ballard and David P. Knobles (Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX)

This work is concerned with modeling the multipath arrivals observed during the CALOPS experiment which occurred off the southeastern coast of Florida in 2007 [K. D. Heaney and J. J. Murray, J. Acoust. Soc. Am. **125**, 1394–1402 (2009)]. The received signals were measured on a horizontal line array and a direct and a refracted path were identified from their arrival angles. The received levels of these signals differed significantly in amplitude owing to the different sediments encountered along each of the respective propagation paths. Using previous results from geological surveys as well as from smaller, localized acoustic experiments, a 3-D model of the seabed over a 70×40 km^2 area of the continental shelf and slope was constructed. The seabed parameters (sound speed, density, attenuation, and seafloor roughness) were then used as inputs to a 3-D propagation code to model the observed data. [Work supported by ONR.]

9:50

3aAO7. Correlating depth-integrated acoustic intensity fluctuations with observed oceanographic events during the Shallow Water 2006 experiment. Jason D. Sagers and David P. Knobles (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, Texas 78713-8029)

During the Shallow Water 2006 experiment, a 224 Hz moored source located on the New Jersey shelf broadcast 7.5 min transmissions every half hour over several weeks. These transmissions were recorded on three vertical line arrays (VLAs) located between 22 and 34 km seaward of the source. The depth-integrated intensity on each VLA is described as a function of time and VLA position. The observed intensity fluctuations (and statistics) and their correlation with temporally and spatially dependent oceanographic events, such as solitons, soliton packets, diffuse internal wave fields, and Tropical Storm Ernesto, will be presented. The intensity fluctuations are examined primarily in terms of 2-D acoustic coupled-mode theory. Seabed attenuation is likely the primary loss mechanism, and it may be possible to use the observed intensity fluctuations to infer information about seabed properties (such as sound speed ratio and attenuation). [Work supported by ONR]

10:05—10:20 Break

10:20

3aAO8. An acoustic tomography scheme with a bottom mounted horizontal line array. Zhenglin Li, Li He, Fenghua Li, Renhe Zhang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing 100190, China, lzhl@mail.ioa.ac.cn), and Mobson Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu)

The sound speed profile plays an important role in shallow water sound propagation. And much attention has been paid on how to obtain it. Concurrences with in-site measurements, many inversion methods have been put forward to estimate sound speed profile from acoustic signals received by a vertical line array. The feasibility and robustness of an acoustic tomography scheme with matched field processing are studied. The acoustic signals from a horizontal line array in South China Sea are analyzed to invert sound speed profiles, in which the sound speed profile is described by the empirical orthogonal functions to reduce the unknown parameters. Parallel genetic algorithm is adopted as the optimization algorithm to increase the inversion speed. The results show that the inverted sound speed profiles are in good agreement with in-site measurements. Moreover, the *a posteriori* probability analysis is used to verify the validity of the inverted results. [Work supported by the National Natural Science Foundation of China under Grant No. 10974218 and the Knowledge Innovation Program of the Chinese Academy of Sciences, Grant No. KZCX1-YW-12-2.]

10:35

3aAO9. Evolving mesoscale structure of two-dimensional sonic layer depth based on multisatellite altimetry and ARGO float profiles. Viviane Menezes (Instituto de Estudos do Mar Almirante Paulo Moreira, Brazilian Navy, Rua Kioto 253, Arraial do Cabo 28930000, Brazil, viviane@vmoceanica.com.br) and Marcio Vianna (VM Oceanica Ltda, São Paulo 12240710, Brazil)

The influence of upper ocean warm core rings (WCRs) on long-range low-frequency sound propagation is well-known in ocean acoustics, through

modeling the effects of sonic depth changes in ray paths and wave scattering. The predictions of any range-dependent acoustic detection model applied to a realistic upper-layer ocean acoustic scenario is strongly dependent on characterization of evolving mesoscale and submesoscale features. Since mesoscale dynamics is not directly predictable from climatology, we describe for the South Atlantic a method that statistically relate ARGO-float subsurface profile positions and derived sonic depths with altimetric and geodetic data objectively mapped in high resolution by an improved fast Gauss transform algorithm. The advantage of the method is that it permits us to go one-step ahead in the practical near-real-time prediction of the horizontal structure that can be quickly modeled to serve as input into the chosen acoustic match-field model, tested against the control model without the WCR complexity. Results from a few experiments are described.

10:50

3aAO10. Integration of ocean and acoustic field modeling in partially specified environments. Steven Finette (Naval Res. Lab., Washington, DC 20375), Frank Gerdes, Thomas Evans, and Colin Shen (Naval Res. Lab., Washington, DC 20375)

The problem of accurately predicting the characteristics of acoustic fields propagating within an ocean environment depends on the amount of information available concerning that environment, an issue that has received considerable attention in recent years. From a modeling perspective, a major goal of this research effort is to achieve robust quantitative estimates of acoustic field uncertainty associated with incomplete knowledge of environmental parameters and fields, so that decisions or actions conditioned on the results of numerical simulation accurately reflect the current state of information. Exact knowledge of these quantities is not available and one can represent such an incomplete state of knowledge (uncertainty) by probability distributions on environmental parameters and fields. We consider a polynomial chaos framework for modeling both environmental and acoustic field uncertainty in an integrated framework. An example is presented of submesoscale space-time evolution of the sound speed field under uncertain initial conditions associated with tidal forcing and temperature in a canonical shelf-break environment, with corresponding estimates of acoustic field intensity caused by this uncertain ocean environment. Alternatives to probability-based modeling of uncertainty are briefly addressed. [Work supported by the Office of Naval Research.]

11:05

3aAO11. Environmental knowledge and operational effectiveness at the NATO Undersea Research Centre. Kyle M. Becker (NURC, Viale San Bartolomeo, 400, 19126 La Spezia (SP), Italy)

One of the primary areas of research at the NATO Undersea Research Centre (NURC) is environmental knowledge and operational effectiveness. The objective of this work is to provide environmental information in support of maritime operations from the planning stages through to real-time decision making. Generically, this is accomplished by parametrizing the environment according to need and providing the best knowledge available for each parameter including uncertainty. In practice, the environment is measured using a variety of mobile and remote sensors, including AUVs, with the aim of improving high-resolution models of the oceanographic environment. In turn, these models and measurements are used as input for operational planning and prediction tools. This talk will describe recent and upcoming experimental work at the NURC designed to test these concepts and their application to acoustic based planning and prediction tools. The goal is to demonstrate the use of these concepts to improve prediction capabilities for transmission loss and signal excess in an operational area.

11:20

3aAO12. Validation of ice-ocean models using acoustic measurements. Hanne Sagen, Laurent Bertino, Pavel Sakov, Stein Sandven (Nansen Environ. and Remote Sensing Ctr., Thormoehlensgt. 47, N-5006 Bergen, Norway, hanne.sagen@nersc.no), and Svein Arild Haugen (NAXYS AS, N-5042 Bergen, Norway)

The Fram Strait is the main passage through which the ocean mass and heat exchange between the Atlantic and Arctic Oceans take place. On the eastern side of the strait the northbound West Spitzbergen Current transports Atlantic water to the Arctic Ocean, whereas on the western side the southbound East Greenland Current transports sea ice and polar water from the Arctic Ocean to the Nordic Seas and the Atlantic Ocean. The currents in the Fram Strait are characterized by significant recirculation and numerous propagating mesoscale eddies. Under the EU project DAMOCLES single-track acoustic travel time measurements were carried out for a year. Acoustic travel time pattern is modeled by feeding oceanographic fields from ice-ocean models into acoustic propagation models. By comparing modeled travel times to accurate acoustic observations we can validate the ocean models. Two different ice-ocean models are considered: a single member high-resolution model and a multi member ice-ocean model system at coarser resolution. This validation approach is complementary to the use of point measurements or single profiles form oceanographic moorings and can provide new insight to the ocean model performance and assess feasibility of assimilating acoustic travel times in ocean models.

11:35

3aAO13. Frequency and angular (vertical and azimuthal) characteristics of reverberation of low-frequency tonal signals (200–300 Hz) in shallow water due to wind waves. Boris M. Salin, Mikhail B. Salin (Inst. of Appl. Phys., Russian Acad. of Sci., 46 Uljanov St., Nizhny Novgorod 603950, Russia, salin@hydro.appl.scinnov.ru), and Robert C. Spindel (Univ. of Washington, Seattle, WA 98105)

This presentation considers the main schemes of calculating the frequency and angular characteristics of reverberation of low-frequency cw signals in shallow water caused by scattering from a rough, wind-driven surface. Experimental and calculated data are given in order to display directivity patterns and levels of the reverberation signals in both vertical and horizontal planes and to display it as a function of the water depth, the propagation path length, the carrier frequency, and the surface roughness spectrum. We note the horizontal angular asymmetry of the scattered side lobes. [B.M.S. and M.B.S. gratefully acknowledge the generous support of the U.S. Office of Naval Research, ONRG.]

11:50

3aAO14. Reconstruction of reverberation levels based on measured surface roughness: Theory and experiment. Boris M. Salin, Mikhail B. Salin (Inst. of Appl. Phys., Russian Acad. of Sci., 46 Uljanov St., Nizhny Novgorod 603950, Russia, salin@hydro.appl.sci-nnov.ru), and Robert C. Spindel (Univ. of Washington, Seattle, WA 98105)

This presentation is devoted to the investigation of sound scattering by wind waves in case of propagation in shallow water (~20 m) at distances of 100–1000 m and frequencies of 1–5 kHz. We consider several models and direct numerical calculation based on small perturbation theory that allows reconstruction of the reverberation levels when given knowledge of the wind-driven rough surface displacement spectrum. Some data were obtained in an experiment where simultaneous acoustic and surface roughness measurements were made. We also describe a new optical method to measure the surface roughness in detail. The technique uses a video recording of the surface over an area about 10×10 wavelengths (the most energetic surface waves) and a zoomed-in video recording of a spar buoy located within this area. Processing the entire image gives a distribution of brightness as a function of time and horizontal coordinates. Processing the buoy's video allows one to obtain a record of surface displacement at that point. The distribution of brightness of the entire area can then be calibrated and recalculated to yield the 2-D (x and y) elevation as a function of time and thus the 3-D surface spectrum. [B.M.S. and M.B.S. acknowledge the support of ONRG.]

WEDNESDAY MORNING, 25 MAY 2011 GRAND BALLROOM A, 8:00 TO 11:15 A.M.

Session 3aBA

Biomedical Acoustics: Assessment and Applications of Contrast Agents

E. Carr Everbach, Cochair
Swarthmore College, Dept. of Engineering, 500 College Ave., Swarthmore, PA 19081-1397

Pei Zhong, Cochair
Duke Univ., Dept. of Mechanical Engineering and Materials Science, 101 Science Dr., Durham, NC 27708-0300

Contributed Papers

8:00

3aBA1. Theoretical considerations for ultrasound contrast agent amplitude modulation techniques at high frequencies. Amin Jafari Sojahrood and Michael Kolios (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada)

Increasing the nonlinear response of ultrasound contrast agents (UCAs) in response to two consecutive pulses in amplitude modulation (AM) techniques can significantly increase the contrast to tissue ratio (CTR). It has been recently shown that one way to increase the nonlinearity is to take advantage of the buckling behavior of lipid shell bubbles. However, this method is limited to frequencies below the resonance of the bubble. To achieve enhanced nonlinearity at high frequencies, the oscillations of the UCAs should be optimized. In this work, the Hoff model was solved for a wide range of the parameters of the UCAs. Results showed that when the bubbles are sonicated with a frequency slightly less than the integer multiples of its resonance frequency (e.g., 2.7 and 3.6), and above a pressure threshold (PT), there is a significant increase in the UCA stable radial oscillations, which corresponds to a significant backscattered signal enhancement. If the sonication frequency of the AM technique is optimized according to the mentioned frequencies, and the amplitudes of the AM first and second pulses are smaller and greater than this PT, the CTR of the AM technique may be enhanced significantly.

8:15

3aBA2. The use of pressure dependent subharmonic resonance to increase the signal to noise ratio of ultrasound contrast agent imaging. Amin Jafari Sojahrood and Michael Kolios (Ryerson Univ., Toronto, ON, Canada)

Conventional methods of subharmonic (SH) imaging of ultrasound contrast agents (UCAs) employ a frequency which is approximately twice the resonance frequency of the UCA (the SH resonance frequency). However, this approach may be limited by a low signal to noise ratio. To address this problem, the Hoff model was solved for several UCAs and the SH resonance frequencies of the bubble were calculated for different applied pressures (PAs). By increasing the PA above the pressure threshold for SH generation, the SH resonance frequency shifts toward lower values. Calculation of the scattered pressure in the regime of stable oscillations show that, if the bubble is sonicated with its pressure dependent SH resonance frequencies, the backscattered signal can be significantly stronger than when sonicating with the conventional SH resonance frequency (although the initiation of the SH occurs at a higher PA). The enhancement of the SH and ultraharmonic amplitudes can be up to 7 and 14 dB, respectively. This enhancement may lead to better UCA SH imaging, especially for molecular based imaging protocols.

8:30

3aBA3. Characterization of cavitation based on autocorrelation of acoustic emissions. Miklós Gyöngy (Fac. of Inf. Tech., Pázmány P. Cath. Univ., P.O. Box 278, Budapest 1444, Hungary, mgyongy@digitus.itk.ppke.hu) and Carl R. Jensen (Univ. of Oxford, Oxford OX3 7DQ, United Kingtom)

Cavitation is a phenomenon that causes different bioeffects depending on its stability and strength (that is, inertial character). In current thinking, stable cavitation is associated with acoustic emissions at the harmonics, subharmonics, and ultraharmonics of the driving insonation frequency while inertial cavitation is associated with broadband emissions. To assess the validity of these assumptions, cavitation emissions from microbubbles excited at 1 MHz in 3% agar gel were simulated as well as experimentally recorded. The results show that cavitation may occur "stably" (repetitively) without sub-, ultra-, or integer harmonic emissions, highlighting the need for more precise measures of stability that cannot be captured by frequency-based methods, namely, the variances of emission amplitude and phase. The results also show that inertial cavitation need not cause broadband emissions. In contrast to these ambiguities, the autocorrelation of cavitation emissions allows the estimation of amplitude and phase stability, while the first zero-crossing of the auto-correlation is shown to be closely related to the widely-used measure R_{max}/R_0 of cavitation strength. Therefore, autocorrelation-based cavitation characterization promises to be a more accurate method of inferring cavitation bioeffects than its frequency-based counterpart.

8:45

3aBA4. Comparing the Marmottant model for ultrasound contrast agents with experimental postexcitation signals. Daniel A. King (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801) and William D. O'Brien, Jr. (Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801)

When insonified with sufficiently large rarefactional pressures, single unconstrained ultrasound contrast agents undergo inertial collapse with postexcitation emissions. Experimental measurements of postexcitation signal data for Definity microbubbles are compared with the Marmottant theoretical model for large amplitude oscillations of ultrasound contrast agents (UCAs). After taking into account the insonifying pulse characteristics, microbubble properties, and size distribution of the population of UCAs, a good comparison between simulated results and experimental data is obtained by determining a threshold maximum radial expansion (R_{max}) to indicate the onset of postexcitation activity. Though this threshold R_{max} is found to vary depending on insonification frequency, the values obtained are well above the typical free bubble inertial cavitation threshold commonly chosen at 2R0. The close agreement between the experiment and models suggests that lipid shelled UCAs behave as unshelled bubbles during most of a large amplitude cavitation cycle, as proposed in the Marmottant equation. [NIH Grant No. R37EB002641.]

9:00

3aBA5. On the dissolution of microbubble contrast agents in an ultrasound field. Jean-Pierre O'Brien, Eleanor Stride, and Nick Ovenden (Dept. of Mathematics, Univ. Col. London, Gower St., London WC1E 6BT, United Kingdom, jpobrien@math.ucl.ac.uk)

Accurate understanding of the behavior of ultrasound contrast particles driven by an external sound source is crucial to enabling the development of better therapeutic and imaging techniques. Unfortunately current models are unable to fully explain, amongst other factors, changes in bubble characteristics following pulsed excitation. One aspect that has hitherto received little attention is the time dependent nature of the microbubble coating, specifically its gas permeability and variable surface tension. We show initially through nonlinear numerical simulations that acoustically driven microbubbles can dissolve more rapidly than stationary bubbles. However, the increased rate observed seems insufficient to account for the significant changes in bubble response over a few pulse cycles observed experimentally. Then, in an attempt to explain this continued discrepancy between models and experiments, we consider the impact of changes in surfactant concentration on the bubble behavior and gas diffusion. In particular, a model of surfactant mass transfer, including shedding, between the surface of the bubble and the bulk liquid is developed and subsequently coupled to a nonlinear bubble equation. Results from this model are presented that can replicate and potentially explain bubble response changes which may occur, leading to improvements in propagation modeling and quantitative imaging software.

9:15

3aBA6. Investigation of microbubble response to long ultrasonic pulses used in therapeutic applications. Christophoros Mannaris and Michalakis A. Averkiou (Dept. of Mech. and Manufacturing Eng., Univ. of Cyprus, 75 Kallipoleos, 1678 Nicosia, Cyprus)

In current drug delivery approaches, microbubbles and drugs can be coadministered intravenously while high intensity ultrasound is applied. The exact mechanisms of microbubble interactions with ultrasound the drug and the cells are not fully understood and sometimes, the optimal ultrasound parameters are simply a case of trial and error. A better understanding of microbubble response to long ultrasonic pulses is important for drug delivery. Two different *in-vitro* setups to examine the response of microbubbles are presented: with the microbubbles suspended in a large enclosure and with the microbubbles flowing in a capillary. Various commercially available and drug delivery specific bubbles were considered. Acoustic streaming, which greatly influences the observed response from bubbles, was observed in many "typical" drug delivery conditions in the large enclosure setup. With the capillary setup, streaming was eliminated and accurate bubble responses were recorded. In the capillary setup and at high pressures where bubble destruction is prominent, all bubble activity disappears within 80–100 cycles despite the length of the excitation pulse, mainly due to diffusion phenomena. A question is raised whether the enhanced drug delivery seen with longer ultrasound pulses at high pressures is related to the microbubble response or whether other factors are involved.

9:30

3aBA7. Estimating concentration of ultrasound contrast agents with backscatter coefficients. Scott M. Leithem, William D. O'Brien, Jr., and Michael L. Oelze (Dept. of Elec. and Comput. Eng., Univ of Illinois at Urbana-Champaign, 1406 W Green St., Urbana, IL 61801, leithem2 @illinois.edu)

Ultrasound contrast agents (UCAs) are used clinically to enhance the contrast of ultrasound images. Recently, microbubbles have been explored as a means to enhance therapeutic techniques. Because the effectiveness of these techniques relies on the UCA concentration at a target site, it would be beneficial to acquire real-time estimates non-invasively. A method for measuring concentration, based upon backscatter coefficients (BSCs) at frequencies above resonance, was developed. Calculation of the BSC was accomplished using plane reference measurements from the back wall of a Plexiglas chamber. For each trial, an average of 500 snapshots of ultrasonic backscatter from Definity microbubbles flowing through the chamber was acquired. Immediately following this procedure, a sample of the UCAs was extracted from the flow path in order to optically verify the concentration estimates. Using measurements of attenuation coefficient through the bubble cloud, BSC was calculated for the 15–25 MHz range. UCA concentration was estimated by using a Levenberg–Marquardt fitting algorithm to match the experimental BSC to a linear scattering model. All BSC-based estimates were within one standard deviation of optically derived estimates. These results indicate that the BSC can be used to measure UCA concentration. [This work was supported by NIH Grant No. R37EB002641.]

9:45—10:00 Break

10:00

3aBA8. Effects of ultrasound on osteoblast proliferation in presence of encapsulated contrast microbubbles. Amit Katiyar, Kausik Sarkar (Mech. Eng., Univ. of Delaware, Newark, DE 19716), and Randall Duncan (Univ. of Delaware, Newark, DE 19716)

The actual mechanism behind the observed accelerated healing of fractures triggered by application of low-intensity pulsed ultrasound is not known. To understand it, the effects of ultrasound on osteoblast cells are studied *in vitro*. Cell proliferation showed significant increase under pulsed ultrasound excitation (0.5–5 MHz, 1–500 mW/cm^2, 200 μs, and 1 kHz PRF) for a duration of 5–30 min daily for the intensity variation of 1–100 mW/cm^2. Intracellular calcium storage and its flux through cellular membrane are believed to play a critical role in cell response to mechanical stimuli. In this study, encapsulated microbubbles are introduced during sonicaton. The bubbles trigger streaming against cell layer and their oscillation may concentrate mechanical forcing on the cell wall. Such activities have been shown to cause sonoporation and enhanced calcium transport. The results of variation in bubble type, concentration, and other excitation parameters such as duration, frequency, and intensity will be presented.

10:15

3aBA9. Ultrasound-induced angiogenesis requires contrast agent collapse. Chenara A. Johnson (Dept. of BioEng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, cjohns42@illinois.edu) and William D. O'Brien, Jr. (Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801)

Ultrasound (US) and ultrasound contrast agents (UCA) ability to induce a particular bioeffect, angiogenesis, has been explored as a means of restoring blood flow to ischemic muscle. Determining the mechanistic motivation of the angiogenic response (AR) is integral to the evaluation of bioeffects, US therapy development, and understanding the physical process. Using a 1 MHz transducer (PRF = 10 Hz, PD = 10 μs), this study explored the effects of oscillation and collapse cavitation (CC) on both acute bioeffects (day 0) and subsequent AR (day 5) in the gracilis muscle of Sprague Dawley rats ($N = 20$). The UCA used, Definity, was determined, in separate experiments, to have a collapse threshold of approximately 0.25 MPa. Because UCAs demonstrate an increasing percentage of CC with increasing US pressure, this study used pressures ranging from 0 to 0.9 MPa. The acute bioeffects were measured via permeability assessments with Evans Blue Dye. ARs were assessed by vascular endothelial growth factor expression. Capillary density was determined by CD31 staining for both time points. The findings support that CC is necessary to elicit an AR and suggest that acute effects could potentially be used to gauge the AR. [Work Supported by F31 HL097653 (C.A.J.) and R37EB002641.]

10:30

3aBA10. Characterization of polymer ultrasound contrast agents. John S. Allen, III, Pavlos Anastasiadis (Dept. of Mech. Eng., Univ. of Hawaii-Manoa, Honolulu, HI 96822), Parag Chitnis, and Jeff Ketterling (Riverside Res. Inst., 156 William St., 9th Fl., New York, 10038-2609)

Ultrasound contrast agents are encapsulated microbubbles with unique acoustic scattering signatures. Polymer shelled agents have been shown to be advantageous for tissue perfusion studies and also more recently high-frequency ultrasound applications. The thickness and material of the poly-

mer shell can be more finely adjusted for the specific application in the production process. Only recently have independent measurements of the shell material parameters been attempted. We present a series of elastic shell parameter measurements made using two complementary techniques: high-frequency acoustic microscopy and nano-indentation (Hysitron device). The elastic properties of the thin films of the polymer material (Philips agents) were measured using both methods. Also variations of these techniques were applied to the full spherical shells to examine curvature effects. Acoustic microscopy measurements indicate values for acoustic impedance of 1.73 MRayl. Simulations are highlighted. [Work supported by NIH EB006372 and 2G12RR003016121.]

10:45

3aBA11. Passive cavitation imaging of echogenic liposomes insonified with 6 MHz pulsed Doppler ultrasound in a flow phantom. Kevin J. Haworth (Intern. Med., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45209), T. Douglas Mast, Kirthi Radhakrishnan, Jonathan A. Kopechek (Univ. of Cincinnati, Cincinnati, OH 45209), Mark T. Burgess (Boston Univ., Boston, MA 02215), Shaoling Huang, David D. McPherson (Univ. of Texas Health Sci. Ctr. at Houston, Houston, TX 77030), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45209)

Microbubble activity has been linked to bioeffects in a variety of therapeutic ultrasound applications including thermal ablation and localized drug delivery. Passive cavitation imaging of microbubble activity has been demonstrated *in vitro* during continuous-wave ultrasound exposure [Salgaonkar *et al.* (2009), JASA **126**, 3071–3083; Gyongy and Coussios 2010, IEEE Trans BME **57**, 48–56]. Here, we perform passive cavitation imaging of echogenic liposomes (ELIPs) in a flow phantom exposed to 6 MHz pulsed Doppler ultrasound (3.1 μs pulse duration, 1250 Hz pulse repetition frequency) transmitted from a diagnostic scanner (Philips HDI 5000, CL15-7 transducer). Frames of received ultrasound signal were obtained passively on 64 parallel channels recording for 112 μs (L8-3 transducer, Zonare z.one ultra scanner). Received echoes were beamformed by frequency-domain phase shifts to produce cavitation images. Localization of cavitation activity was tested by moving the Doppler exposure volume to three different positions and observing a concurrent spatial shift in cavitation activity. At the highest insonation pressures a loss in echogenicity was observed. Cavitation activity coincided spatially with this loss. Subharmonic and broadband cavitation images for multiple insonation pressures were consistent with previously determined stable and inertial cavitation thresholds for ELIP. [Work supported in part by NIH grants F32HL104916, R01HL074002, R21EB008483, R01HL059586, and R01NS047603.]

11:00

3aBA12. Subharmonic response from ultrasound contrast microbubbles for noninvasive blood pressure estimation. Amit Katiyar, Kausik Sarkar (Mech. Eng., Univ. of Delaware, Newark, DE 19716), and Flemming Forsberg (Thomas Jefferson Univ., Philadelphia, PA 19107)

Estimation of local organ-level blood pressure can help in diagnosing and monitoring heart and vascular diseases. Subharmonic signals from ultrasound contrast microbubbles have been proposed as a noninvasive alternative to the current practice of using a manometer-tipped catheter. Approximately 10 dB linear decrease in subharmonic component with 25 kPa pressure increase (typical blood pressure variation) has been reported for several contrast microbubbles. A theoretical investigation of the underlying phenomenon will be reported here. First the well-established model of a free microbubble is studied to show that reduction in subharmonic with ambient pressure increase occurs only below a certain excitation frequency. Above another critical frequency, subharmonic signal increases with ambient pressure. In between the variation is nonmonotonic. Furthermore, where it decreases with ambient pressure, the relationship is linear only above a certain excitation pressure. The behavior is explained by analyzing the dependence of resonance frequency of a bubble on the ambient pressure. The dependence of the critical frequencies on bubble radius and possibly bubble size distribution will be discussed. Behaviors for several models for encapsulated contrast microbubbles will also be reported. [Work Supported by the NSF and the NIH.]

WEDNESDAY MORNING, 25 MAY 2011 DIAMOND, 8:00 TO 9:40 A.M.

Session 3aEAa

Engineering Acoustics, Signal Processing in Acoustics, Noise, and Physical Acoustics: Sensors and Systems for Acoustic Detection, Localization, and Characterization of Underground Structures, Objects, and Tunnels

Michael V. Scanlon, Chair
U.S. Army Research Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197

Chair's Introduction—8:00

Invited Papers

8:05

3aEAa1. Laser sensing techniques for acoustic detection of buried mines. Vyacheslav Aranchuk, James M. Sabatier, and Richard Burgett (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, aranchuk@olemiss.edu)

Acoustic detection of buried mines is based on excitation of vibrations in the ground with airborne sound and measuring vibration of the ground surface by using a laser interferometric sensor. The principle of detection uses the fact that buried mines have mechanical impedances that are smaller than that of soils. That makes the soil above a mine vibrating with magnitude higher than the surrounding soil. A laser sensor creates a vibration image of the ground by measuring vibration in many points on the surface. The presence of a buried mine can be detected by an abnormality in the vibration image. The method has shown excellent performance in field tests. One limiting factor of the method is a long measurement time due to low frequency of vibrations. Different laser interferometric techniques have been investigated to reduce the time of detection. The paper discusses application of single-beam scanning laser Doppler vibrometry (LDV), multiple-beam LDV, and speckle-interferometry based methods for acoustic detection of mines.

8:30

3aEAa2. Development of a portable system for acoustical reconstruction of tunnel and cave geometries. David L. Bowen (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, dbowen@acentech.com)

Noninvasive remote determination of the geometry of caves, tunnels, and piping complexes has value for both military and commercial activities. Troops that must enter caves, tunnels, or urban storm sewer piping systems, for example, would be greatly aided by first knowing the distance to the next turn, opening, branch, or closure. A system employing a sound source and a pair of acoustic sensors was developed to do this, based on the medical techniques of acoustic laryngometry and rhinometry, and has been demonstrated both in laboratory and in full-scale field situations. Acoustical reflectometry techniques are first used to compute an area versus distance profile of the passageway under interrogation. Other procedures are then used to classify and characterize the various types of "junctions" that could give rise to the observed changes in area (branching versus a simple area expansion, etc.). Results obtained using a prototype field system in full-scale tests indicate that small arms fire can be used in place of a loudspeaker as the sound source (with hydrophones as the sensing elements), thus increasing the portability of the system. Other techniques for remote geometry mapping of tunnel-like spaces will also be discussed. [Work supported by the U.S. Army Corps of Engineers.]

Contributed Papers

8:55

3aEAa3. Soil plate oscillator: Modeling nonlinear mesoscopic elastic behavior and hysteresis in acoustic landmine detection. Dang V. Duong (Dept. of Weapons and Systems Eng., U.S. Naval Acad., Annapolis, MD 21402) and Murray S. Korman (U.S. Naval Acad., Annapolis, MD 21402)

An experimental apparatus, designed to study flexural vibrations of a soil loaded plate, consists of a thin circular elastic plate clamped at the boundary between thick round flanges. The soil column is supported by the plate and upper cylindrical wall. A small magnet attached to the center of the plate is driven by a rigid ac coil (located coaxially below the plate) to complete the electrodynamic soil plate oscillator (SPO) design. Measurements of the electrical motional impedance Z_{mot} of the SPO versus frequency are made using the complex output to input response of a Wheatstone bridge that has an identical coil element in one of its legs. The mechanical impedance Z_{mech} (force/particle velocity, at the plate's center) is inversely proportional to Z_{mot}. Experiments, with and without mass loading at the plate's center, verify Snowden's theory for a clamped plate [J. Acoust. Soc. Am. (1971)] and help establish mechanical parameters. Resonant oscillations (for various point mass loadings) provided effective mass, spring, damping and coupling constant parameters of the system. "Tuning curve" behaviors of real $\{Z_{mot}\}$ and imaginary $\{Z_{mot}\}$ at successive vibration amplitudes (for various soil column heights) are used to develop a general mesoscopic nonlinear elastic model for granular media.

9:10

3aEAa4. Monaural source separation in underground spaces via non-negative and complex matrix factorization. Brian J. King and Les E. Atlas (Dept. of Elec. Eng., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr., Rm. AE100R, Campus Box 352500, Seattle, WA 98195-2500, bbking@u.washington.edu)

Automated separation of multiple independent acoustic sources collected on a single monaural sound channel is an active research area and can aid in many applications, including practical automatic speech recognition, speaker identification, and keyword identification in multitalker, noisy, and/or reverberberant multisource environments. Some of the latest and most promising methods of single-channel source separation are non-negative and the even more recent complex matrix factorization, which decompose a signal into a sparse linear combination of source-specific building blocks, commonly referred to as bases. Once the bases and accompanying weights are calculated, separating a source from the mixture is achieved by multiplying and summing together its corresponding bases and weights. While experiments exhibit significant separation, the majority of this work has been done on studio-recorded audio and consequently little done in more realistic acoustic environments. Some of the most important environments are underground spaces, which are usually heavily damped, highly reverberant, and noisy. In this work, we will develop and compare matrix factorization single-channel source separation algorithms in reverberant conditions which simulate acoustic transmission through tunnels and other underground environments. We will conclude by demonstrating the most promising methods and discussing their current limitations. [This research was supported by AFOSR Grant No. FA9550061019.]

9:25

3aEAa5. Rayleigh wave technique for underground explorations. Zhiqu Lu (Natl. Ctr. for Physical Acoust., The Univ. of MS, University, MS 38677), Glenn Wilson (USDA-ARS Natl. Sedimentation Lab., Oxford, MS 38655), and Tianyu Zhang (Beijing Normal Univ., Beijing, 100875, China)

A laser Doppler vibrometer (LDV) based multi-channel analysis of surface wave (MASW) method has recently been developed. In the method, an electro-mechanical shaker was used as a seismic source operating in a frequency sweeping mode to excite Rayleigh waves propagating through the soil surface. The surface vibrations along a straight line were detected by a moving LDV. Unlike conventional MASWs that explore soil profile from a few meters to a few tens of meters below the ground, the present MASW investigates underground depth from a few centimeters to a few meters due to its high frequency excitation and high spatial resolution of moving LDV. Two approaches will be used to create two-dimensional (2D) image of subsurface features: dynamic moveout and dispersive curve inversion. The geological anomalies can be identified from the contrasts of the 2D image. Some details and cases will be addressed.

Session 3aEAb

Engineering Acoustics, Underwater Acoustics, and Animal Bioacoustics: Advanced Acoustic Systems For Small Underwater Vehicles (Autonomous and Towed)

Kenneth M. Walsh, Chair
K. M. Engineering Ltd., 51 Bayberry Ln., Middletown, RI 02842

Chair's Introduction—9:55

Invited Papers

10:00

3aEAb1. Synthetic aperture sonars for imaging of marine geology and buried objects. Steven G. Schock and Victoria Ringle (Dept. of Ocean and Mech. Eng., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431, sschock@fau.edu)

Large acoustic apertures are necessary for generating high quality imagery of marine sediments and objects buried beneath the seafloor. These aperture sizes are usually much larger than the widths and lengths of small underwater vehicles. A large aperture is formed along the track of the vehicle by utilization of synthetic aperture processing. Vehicle wings, containing hydrophone arrays that are extended after vehicle submergence, are used to form a large across vehicle track aperture. Imagery of sediment layering is formed by time domain focusing on planar surfaces. However, buried object images are formed by time domain focusing at subsurface focal points. Images of sediments and buried objects are used to measure the performance of these techniques.

10:20

3aEAb2. Advances in acoustic communication and navigation transducers for small underwater vehicles. David A. Brown (BTech Acoust., ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbacoust.@cox.net)

The application of piezoceramic (PZT) and single crystal (PMN-PT) transducers for underwater acoustic communication modems and navigation systems is reviewed. Many applications demand increased acoustic bandwidth, moderate to high acoustic power, compact form factors, and in some cases directionality. The presentation reviews Lagrangian based analytical modeling methods and the development of a MATLAB based users-tool kit for virtual prototyping. Experimental results and field testing of many devices are summarized including single crystal fully active segmented modem transducers with measured coupling factors of 0.87 having usable fractional bandwidth of greater than 100%. Several transducers for navigation aids are presentment including pressure gradient multimode, baffled cylinders and spheres, compact arrays, and spiral-wavefront transducers. Examples of the influence of the vehicle body on transducer performance are considered in an experimental investigation of a Webb sea glider with a BTech WHOI-micromodem transducer and hull and wing mounted sensors for trawler avoidance. [Work supported by the ONR.]

10:40

3aEAb3. Vibrational response of a simulated 1-3 composite array. Andrew Hull and Benjamin Cray (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841)

The objective of this work was to identify structural waves (and their corresponding parameters) that propagate in plates that consist of periodic elements embedded in filler materials. To this end, wavevector decompositions were made of highly-sampled (in time and space) measurement data, using two different plates, one of which was isotropic and the other fabricated so that its mechanical structure emulates a typical 1-3 composite array plate. Knowledge of the various wave types that propagate and attenuate will reveal the lateral deformation of the elements (sensors) along with the normal stresses and shear stresses that are applied by the filler material onto the sensors. These stresses produce secondary voltages in the sensors which are detrimental to the primary voltage output and thus degrade array performance. A secondary objective therefore will be to determine values of the parameters (e.g., distance between sensors and filler material) that minimize these detrimental effects in the frequency band of the sensor system.

11:00

3aEAb4. Modem-based aids for underwater navigation. Dale Green and Steve McManus (Teledyne Benthos, 49 Edgerton Dr., North Falmouth, MA 02556)

Both autonomous and semi-autonomous underwater platforms require navigation capabilities. Acoustics have long been the preferred method because all other methods are so limited in range and/or are so costly. Traditional methods have relied on tonal-based waveforms used with any of three approaches: long baseline (LBL), short baseline, or ultra-short baseline (USBL). These systems can be highly effective when propagation conditions are "good" but may incur high cost in terms of false and missed detections. The use of wide-band waveforms can substantially improve performance, but a significant cost is incurred in the complexity required at the receiver. However, given this complexity, consideration should be given to the use of underwater acoustic communications (acomms) as the provider of enhanced navigation aids. We describe three modem-based navigation aids based, two of which have never exhibited a

false alarm, and each of which provides substantial improvement over tonal-based systems. We describe a spread spectrum signaling method used in LBL navigation, an underwater GPS system in which any number of platforms can obtain accurate geoposition in a completely passive sense, and we describe a physically small, wide-band modification to conventional USBL. All of these systems provide high accuracy navigation while retaining full acomms links.

11:20

3aEAb5. System level tradeoffs for broadband transduction in unmanned underwater vehicle applications. Stephen C. Thompson, Richard J. Meyer, Jr., Eric M. Beinert, and Thomas C. Montgomery (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

Unmanned underwater vehicle (UUV) applications, with their inherently severe constraints on size, weight, and power consumption, are being asked to perform acoustic tasks that might previously have been accomodated shipboard. When an acoustic system needs a projector with more than an octave bandwidth at moderate or higher power with a high total system efficiency, there is a dearth of options available. In fact, there are no well established figures of merit for rating transducers in this operating regime, as those appropriate for conventional systems generally assume a single degree of freedom system operating near reasonance. This paper will discuss the many dimensions of this issue.

Contributed Paper

11:40

3aEAb6. Flow noise of underwater vector sensor embedded in a flexible towed array. Vladimir Korenbaum and Alexander Tagiltsev (Dept. of Underwater Technique, Pacific Oceanologic Inst., 43 Baltiiskaya Str., Vladivostok, 690041 Russia)

The objective of this work is the simulation of the flow noise of a vector sensor embedded in a flexible towed array. The developed mathematical model, based on long-wave analysis of the inner space of a cylindrical multipole source, predicts essential cancellation of flow noise of a vector sensor embedded in a underwater flexible towed array by means of intensimetric processing (cross-spectral density calculation of responses of oscillating velocity and sound pressure sensors). It is experimentally found that intensimetric processing results in flow noise cancellation by 12–25 dB on mean levels and by 10–30 dB on fluctuations *re* squared oscillating velocity channel. The effect of flow noise suppression in the intensimetry channel *re* squared sound pressure channel is observed only for frequencies above a threshold. These suppression values are 10–15 dB on mean noise levels and 3–6 dB on fluctuations. The threshold frequency on fluctuations is between 30 and 45 Hz under towing velocities 1.5–3 ms^{-1} and accumulation time 98.3 s.

WEDNESDAY MORNING, 25 MAY 2011 GRAND BALLROOM D, 8:25 TO 11:20 A.M.

Session 3aID

Interdisciplinary: Public Relations, Student Council, and Education in Acoustics: Effective Communication Between Scientists and the Media

Andrew A. Piacsek, Chair
Central Washington Univ., Dept. of Physics, 400 E. University Way, Ellensburg, WA 98926

Chair's Introduction—8:25

Invited Papers

8:30

3aID1. Science news in the 21st century. Devin C. Powell (Science News, 1719 N St., Washington, DC 20036)

Open a newspaper in 1980s, and youward likely find a science section. Open one in 2011, and you may not even find a science reporter. With the rise of the Internet and the decline of print, the news world is changing. Science journalism is increasingly relegated to pages of specialty publications. American science literacy remains poor. And the NSF is pushing scientists to make a greater effort to reach out to the public. The presenter will draw upon his experiences as a science reporter to illustrate and describe the state of science journalism, what the American public knows about science, and your place as an acoustic researcher in this ever-changing landscape.

8:45

3aID2. Tips and tools for talking to the press. Jason S Bardi (News Div., Univ. of California, San Francisco, 333 California St., San Francisco, CA 94143)

Your latest paper is about to be published in a major journal, your press officer has issued a news release to the media, the phone is starting to ring off the hook, and your inbox is filling up with emails ending with domain names such as bbc.co.uk and discovermagazine.com. Now what? How can you get the most for yourself and your field out of this latest 15 min of fame? Join Jason Bardi, veteran press officer at the American Institute of Physics, set for a discussion of what journalists expect and hope to get from an interview. Hear about your rights and responsibilities as a source. Learn how to make the most of your time with the press, including best practices for handling media inquiries and helpful hints for describing your work.

9:00

3aID3. A veteran reporter's perspective on science. Alan Boyle (MSNBC on the Internet, Bldg. 25/N2, 1 Microsoft Way, Redmond, WA 98052, alan.boyle@msnbc.com)

As MSNBC.com's science editor, Alan Boyle runs a virtual curiosity shop of the physical sciences and space exploration, plus paleontology, archaeology, and other ologies that strike his fancy. During his 34 years of daily journalism in Cincinnati, Spokane and Seattle, he survived a hurricane, a volcanic eruption, a total solar eclipse, and an earthquake. Join Boyle for an insider's take on the mind of the reporter—how journalists approach complex scientific topics and complex scientists, and what information they are looking for.

9:15

3aID4. Talking to the media: Lessons in crossing the great divide. Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195-7920, pkkuhl@u.washington.edu)

Most scientists consider talking to the media akin to crossing a minefield with no protection. You not only have to worry about potentially making a fool of yourself or failing to describe your work in language a lay audience can digest. You also have to worry about properly highlighting the role played by your institution, your home department, your funding agency, your donors, your co-authors, and your colleagues, who (no matter what you say), will believe you focused unduly on your own work to the exclusion of theirs. The press will not allow you to review the to-be-published story so you will not be able (even with exquisite attention to detail) solve the issues raised above. What's a scientist to do? For tales, pitfalls, and some suggestions (not necessarily solutions) regarding crossing the science-media divide, attend this talk!

9:30

3aID5. Talking to journalists: And knowing they never get it right!. Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Communicating with journalists poses a major dilemma: We have an obligation to explain science to the public because there are so many voices that denigrate scientists and their work; yet, in order for the journalists to effectively communicate, they often dumb it down so much that they get it wrong, and then we are embarrassed in front of our scientific colleagues. This dilemma can be mitigated somewhat by speaking only to those journalists who understand the science and to whom will allow you to edit their work. Of course, it is often difficult to determine, in the middle of an interview, the scientific capability of an interviewer, and often journalists do not like to have their work edited. In the end, a greater understanding of science by the general public is in our best interests and interviews with journalists should (nearly) always be granted. Some reflections on my experiences with such interviews will be given.

9:45

3aID6. Finding my voice: My expericences communicating research with the media. Kelly Benoit-Bird (College of Oceanic and Atmospheric Sci., Oregon State Univ., 104 COAS Admin Bldg., Corvallis, OR 97331, kbenoit@coas.oregonstate.edu)

Talking with the media can be intimidating but it can also provide a great platform for reaching the public, increasing the impact of your research. Recently, I have had the opportunity to interact with television, radio, magazine, and newspaper outlets about my work. These experiences have ranged from frustrating to inspiring. So, how do you make sure that coverage of your work is at least accurate and hopefully engaging for all concerned? First, do not be afraid to ask for help. If your institution has an outreach, communications, or press office, talk with them before the need to interact with the press comes up and each time you are contacted by the media. Second, before agreeing to an interview, be sure you know who you are talking to. Determine what the focus of the story they are working on is, if the interviewer has an agenda, and why they have chosen to talk to you. Third, prepare your key messages in advance and repeat them often during the interview. These messages should be clear and concise so they are easy to quote or turn into sound bites. Finally, practice saying what you want heard and answering questions.

Contributed Paper

10:00

3aID7. When "keep it simple" turns into dangerous misconceptions: A Brazilian case study. Stephan Paul (Acoust. Eng. Program, DECC, Fed. Univ. of Santa Maria, Av. Roraima 1000, Camobi, Santa Maria 97060-900, RS, Brazil)

Communicating acoustics to naive subjects by means of the news media requires a good balance between "keep it simple" and correctness. Unfortunately in Brazil, the balance is usually not maintained and "keep it simple" is often even substituted by erratic use of terms and concepts, not only in news media material [Paul, S., Proceedings of the XXIII Encontro da Soc. Brasileira de Acstica (2010)]. The analysis of Brazilian news media material, but also folders, brochures, etc., issued by equipment suppliers, shows terrific errors regarding even simples concepts such as the definition of sound. Unfortunately it can be even shown that erratic use of terms and concepts in news media, brochures, etc., negatively affects the Brazilian legislative, causing erratic noise pollution laws. To change this specific situation in Brazil, a joint effort of the Brazilian acoustics community is required, e.g., on (1) issuing a guide on "communicating acoustics correctly," (2) introducing a Brazilian standard that defines terms and concepts in acoustics (first proposals made by Souza and Paul: [Ferramentas para uma padronização dos termos utilizados em acxfa;stica e vibrações, Acstica&Vibrações, 42, (2010)], and (3) thorough revision of Brazilian print/online material.

10:15—10:30 Break

Invited Paper

10:30

3aID8. The cold call: How to face an interview. Alan Boyle (MSNBC on the Internet, Bldg. 25/N2, 1 Microsoft Way, Redmond, WA 98052, alan.boyle@msnbc.com)

Talking to a reporter can be an exciting or a harrowing experience. In the second part of his talk, veteran science reporter Alan Boyle will conduct a mock interview of scientist Steven Garrett of the Acoustical Society of America. Watch the 10 min interview unfold to experience firsthand the potential pitfalls and opportunities involved in this experience.

10:50—11:20 Panel Discussion

WEDNESDAY MORNING, 25 MAY 2011 — ASPEN, 8:00 A.M. TO 12:05 P.M.

Session 3aMU

Musical Acoustics: Materials in Musical Instruments

Uwe J. Hansen, Chair
Indiana State Univ., Dept. of Physics, Terre Haute, IN 47809

Invited Papers

8:00

3aMU1. Materials in musical instruments. Uwe J. Hansen (Dept. of Chemistry & Phys., Indiana State Univ., Terre Haute, IN 47809)

Materials play an essential role in determining tonal characteristics and tone quality of string and percussion instruments. Their role in wind instruments is a matter of ardent debate among scientists, instrument makers, and performers. For instruments in which the structural elements are primarily responsible for the radiated sound this importance is self-evident. That significance is less obvious for instruments where the structure primarily serves to contain the vibrating air column. A few general concerns will be discussed and specific examples will be given.

8:20

3aMU2. Material properties of steel in the steelpan. Andrew C. Morrison (Dept. of Phys., DePaul Univ., 2219 N. Kenmore Ave., Chicago, IL 60614)

The Caribbean steelpan is one of the most recently developed tuned percussion instruments and has been the subject of much scientific study in recent years. The tuning of a steelpan is a complex process involving sinking the bowl of the pan, marking the note areas, heating the pan to relieve stress, and hammering notes into tune. Throughout the tuning process the material properties of the steel change measurably. After the pan is tuned, it will need periodic retuning on roughly an annual basis, depending on how often the instrument is played. A process of enriching the steel with nitrogen has been developed which increases the surface hardness of the steel significantly more than traditional methods of tuning. This nitriding process lengthens the time needed between retuning the instrument.

8:40

3aMU3. Materials in percussion instrument. Garry M. Kvistad (Woodstock Percussion, Inc., 167 DuBois Rd., Shokan, NY 12481, garry@chimes.com)

As ideophones, the sounds of percussion instruments are very dependent on the materials of which they are made. The physical properties are important as are the methods of processing the materials that determine the dryness, temperament, and speed of sound. In this presentation, the focus will be from a performer's point of view. The choice of mallets used to activate the instrument is as important as the instrument itself, and mallets are made of many varieties of materials as well. In addition to being a performing musician, the presenter is an instrument builder, which yields a unique relationship to the understanding of sound production. His orchestral training and instrument building experience work together, creating an understanding of how to produce different sounds with different materials and how sound is produced from a acoustical standpoint. Many different percussion instruments will be used to demonstrate the effect materials and methods of manufacturing have on the timbre of the instrument.

9:00

3aMU4. Wood for guitars. Trevor Gore (Trevor Gore Guitars, P.O. Box 469, Terrey Hills, NSW, 2084, Australia)

Numerous famous luthiers have used low grade salvaged timber and non-wood products to demonstrate that how a guitar is designed to exploit available materials is more important than using prime tonewoods. The material properties of timber are highly variable and are not the single figures frequently quoted in reference books. Within species material properties can vary by a factor of 2. Consequently, there is significant overlap of the material properties of one species with others, implying that wood species substitution is possible with little acoustical impact if the component is designed and built to acoustical tolerances rather than dimensional tolerances. However, species selection remains a significant factor in designing guitar components, primarily for structural rather than acoustical reasons. The woods chosen have to survive long-term loading without excessive distortion over time while still allowing the radiating surfaces to vibrate freely. Important parameters include Young's modulus, density, stability with humidity variation, heat bendability, and hardness. The author considers wood for soundboards, braces, backs, sides, necks, fretboards, and bridges. Guitars designed to acoustical criteria (rather than dimensional criteria), where the effects of different stiffnesses and densities of species are minimized, sound very similar.

9:20

3aMU5. The choice of materials for "brass" instruments. Robert Pyle (11 Holworthy Pl., Cambridge, MA 02138, rpyle@post.harvard.edu)

It is generally accepted by players and builders of brass instruments that instrument response and timbre are significantly affected by what would appear to an "outsider" to be minor changes in the materials used to construct the instrument. This talk will present the opinions of players and makers as learned through personal interviews and from advertising literature. Particular attention will be paid to the process of refining a trumpet design to tailor the instrument to the wishes of an individual player.

9:40

3aMU6. Evolution of music wire and its impact on the development of the piano. N Giordano (Dept. of Phys., Purdue Univ., 525 Northwestern Ave., West Lafayette, IN 47907, giordano@purdue.edu)

The earliest pianos were strung with brass and iron strings, and the material properties of these strings placed limitations on the design of these instruments. Over time, improvements in technology produced wire with improved tensile strength, which allowed important changes in piano design. This work culminated in the mid-to-late 1800s with the availability of steel music wire, and piano design has changed remarkably little since that time. This talk reviews how and why improvements in music wire influenced the development of the piano and speculates on how further improvements might (or might not) impact piano design. Our analysis will include spectral data from instruments from several different eras.

10:00—10:15 Break

10:15

3aMU7. Zinc for organ pipe building. Judit Angster, Zlatko Dubovski, and Andras Miklos (Fraunhofer Institut fur Bauphysik, Nobelstr. 12, 70569 Stuttgart, Germany)

Zinc is being used only rarely for organ pipes, since high quality material has not been available. The slightly harder metal makes manual processing of the pipes more complicated for the voicer. Nevertheless zinc is much more cost-efficient. Grillo Werke (Germany) has developed a "soft zinc strip" which is significantly better, as far as workability is concerned, than materials available in the past. This material has been tested to determine to what extent zinc modifies pipe wall vibrations, and how essential the impact of the material would be on the sound. Various pairs of pipes made of plain metal and of zinc have been voiced for the same sound, if possible, according to subjective assessment. The experiments prove that the wall vibration spectra of flue pipes made of various materials clearly show differences. Total velocity spectra recorded by means of 3D scanning Doppler laser vibrometry vary considerably. The results of analyzing the stationary sound spectra and the attack transient, however, allow the voicing of these pipes to the same sound. Recording by a microphone array system also proved that the wall vibration does not have any direct influence on the sound radiation of the pipe.

10:35

3aMU8. Aluminum handbells for enhanced bass sound. Thomas D. Rossing (Stanford Univ., 26464 Taaffe Rd., Los Altos Hills, CA 94022, rossing@ccrma.stanford.edu)

In order to obtain a higher radiation efficiency and thereby enhance the sound of bass bells, the Malmark company makes a bass handbell of aluminum. [Rossing, *et al.*, MRS Bull. **20**, 40 (1995)]. These aluminum bells are larger in diameter than the corresponding bronze bells, and they have lower coincidence frequencies, both of which lead to more efficient radiation of bass notes. In addition they are considerably lighter in weight, and thus they are more easily handled by bell ringers.

10:55

3aMU9. Bamboo in Asian musical instruments. James P. Cottingham (Phys. Dept., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

The acoustical properties of bamboo, along with its widespread availability, have made it one of the most commonly used materials for the construction of musical instruments worldwide. Among its properties important for acoustics, bamboo has a relatively low density and elastic modulus and a significant difference between the elastic moduli parallel to and perpendicular to the bamboo fibers. An extensive general review of the acoustically relevant properties of bamboo has recently been published [U. G. K. Wegst, Annu. Rev. Mater. Res. **38**, 323–349 (2008)]. This paper presents some recent results on measured physical properties of bamboo as used in Asian wind and percussion instruments. The mechanical properties of the bamboo reeds used in some free reed wind instruments have been studied. Since a wide variety of flutes and reed wind instruments employ bamboo pipes the elastic moduli of bamboo pipes, as well as the non-uniformity of the pipe wall material are of current interest, in particular, as related to pipe wall vibrations. The consequences of the non-uniformity in composition and elastic modulus of quasi-cylindrical bamboo pipes are also relevant when these pipe sections are used in percussion instruments.

11:15

3aMU10. Material properties of pipes and reeds from the Southeast Asian khaen. Kyle W. Hershey and James P. Cottingham (Phys. Dept., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, kwhershey@coe.edu)

The khaen is constructed with free reeds mounted in bamboo pipes and enclosed in a carved hardwood wind chamber. These instruments have traditionally been constructed with simple tools using natural materials, primarily bamboo and wood. Unlike some Asian traditional instruments, the khaen has not undergone modernization, and high quality instruments are still made in the traditional manner by village craftsmen. In the past bamboo could have been used for the reeds as well as the pipes, but metal reeds have been used extensively for some time. Measurements have been made on reeds and pipes from two khaen: one a high quality instrument made in northeastern Thailand and one an inexpensive khaen purchased at an import shop. Analysis and comparison have been made of the physical properties of the bamboo pipes, the composition of the alloys used for the metal reeds, and the mechanical properties of the reed such as the elastic modulus. Comparisons can be made with an earlier published analysis of the properties of the bamboo pipes and metal reeds from a traditionally made khaen [Picken *et al.*, Musica Asiatica **4**, 117–154 (1984)].

Contributed Papers

11:35

3aMU11. Soundbox behavior in environments with varying sound speed. Chris Waltham, Mo Chen, Andrzej Kotlicki, Nathan Wolfe, Jing Fei Yu, and Chengchong Zhu (Dept. of Phys. and Astronomy, Univ. of BC, Vancouver, BC, V6T 1Z1, Canada, cew@phas.ubc.ca)

The radiation produced by the wooden soundbox of a string instrument is largely the result of the coupling of oscillations in the wood and those of the enclosed air. The traditional way of studying this interaction is to make modifications to the instrument such as adding masses to the soundboard and chimneys to the soundhole. Presented here is an alternative approach that changes the sound speed of the environment using a tent filled with different gases. The resulting admittance data, expressed as contour plots, show clearly the relationship between air and wood modes. The effect of these modes on the radiation of the instrument is demonstrated by measurements of monopole radiativity for a guitar (with and without the hole blocked) and these data are compared to the output of a simple model. Other instruments studied with the gas tent include violins (regular and balsa), viola, setar, and sound.

11:50

3aMU12. Designing soundboards with flexural disk models. Evan B. Davis (Brugh Davis Acoust., 8556 Burke Ave. N., Seattle, WA)

Musical instruments are designed to be efficient sound radiators that can survive the rigors of the handling and use by musicians. A flexural disk soundboard and ported box system is developed to explore various structural acoustic design strategies. The flexural disk is used to link the structural and acoustic properties of the soundboard. Musical instruments are designed to reproduce a range of frequencies. The playing frequency range of an instrument is defined as being from the frequency of lowest open string to the frequency an octave above the frequency of the highest open string. The fundamental or main wood mode of a string musical instrument, plucked or bowed, is observed to be at the center of the instrument's playing range. The main air mode is placed approximately an octave below the main wood mode. Soundboards are sized to be approximately a quarter of the acoustic wavelength in diameter at the main wood frequency. The simple flexural disk models with their linked structural acoustic properties demonstrate a solid structural acoustic logic to the empirically developed instruments and why these designs have been so resistant to change by inventive and creative contemporary instrument makers.

Session 3aNSa

Noise, Committee on Standards, Physical Acoustics, Animal Bioacoustics, and Underwater Acoustics: Wind Turbine Noise

Edward C. Duncan, Cochair
Resource Systems Group Inc., 55 Railroad Row, White River Junction, VT 05001

Kenneth H. Kaliski, Cochair
Resource Systems Group Inc., 55 Railroad Row, White River Junction, VT 05001

Invited Papers

8:00

3aNSa1. Philosophical viewpoints on noise from wind power projects. Edward C. D. Duncan (RSG, 55 Railroad Row, White River Junction, VT 05001, eduncan@rsginc.com)

Many people are involved with evaluating, engineering, developing, and opposing wind power projects. While there are many worldviews on global warming, renewable energy, and green technologies, this paper focuses on the philosophical viewpoints surrounding noise from wind power projects. After identifying various stakeholders, the paper investigates how some great historical and current minds have contributed, likely unknowingly, to this field including Gifford Pinchot, John Muir, Jonathan Edwards, Mark Sagoff, Bryan Norton, and Cass Sunstein. A brief review of the history of noise in the rural environment comparing transportation noise with wind turbine noise will be provided. The relation between cost-benefit analysis, the precautionary principle, and wind power noise ordinances will be discussed. The paper will wrap up with some thoughts of what Sagoff may have to offer regarding NIMBYism and why Nortons call for convergence is also a call for acousticians to practice good science.

8:20

3aNSa2. Collecting data on wind turbine sound to identify causes of identified concerns. William K. Palmer (TRI-LEA-EM, 76 Sideroad 33-34 Saugeen, RR 5, Paisley, ON N0G 2N0, Canada, trileaem@bmts.com)

Regulations for wind turbines are generally based on *A*-weighted sound levels, and typical sound spectrums found in the community often from localized sources, such as an exhaust stack. The regulatory limits are generally at levels known to cause little annoyance in the community. Large industrial wind turbines with a sound emitter from 50 to 150 m overhead, and based on a source perhaps 100 m in diameter pose a relatively new source of sound to communities, particularly the rural communities where they are located. Community experience shows that the same *A*-weighted sound limits that are acceptable for many typical sound spectrums and localized sources and pose a considerable level of annoyance for wind turbines. This paper sets out to identify the differences in the sound levels found at locations 500–600 m from wind turbines (about one-third mile) considered acceptable by regulators near wind turbines, in both spectrum, intensity, duration, and special characteristics, such as tonality or amplitude modulation (cyclical pattern) compared to the sound levels at control sites distant by at least 5000 m (about 3 miles) from wind turbines. An explanation of the data collection method is given, as well as an analysis of extensive sound samples gathered.

8:40

3aNSa3. Audible low-frequency wind turbine sound. Harrison, F. Richarz (Univ. of Leicester, Leicester, United Kingdom, hricharz@gmail.com) and Werner G. Richarz (Aercoustics Eng. Ltd., 50 Ronson Dr., Toronto, ON, Canada)

To date a large fraction of research on wind turbine sound has been directed at the audible spectrum. Rotation of wind turbine blades is a primary cause of the cyclic broad-band sound that is characteristic of wind turbines. Auto-correlations of measured wind turbine sounds exhibit distinct, periodic “low frequency” pulses. The pulse duration is but a fraction of the blade passage period, with energy content of the order of 20–40% of the overall mean square pressure. Analytical decomposition of the low-frequency pulses reveals line spectra whose components diminish relatively slowly. The spectrum levels of the frequency components above 20 Hz are sufficiently high to be audible under favorable ambient conditions. This hypothesis was tested by generating and listening to various possible pressure time histories that share the shape of the low-frequency pulses as measured by auto-correlations. When propagation effects such as scattering are accounted for, the relative phase shifts suffered by the higher frequency components result in an audible swoosh.

9:00

3aNSa4. A community noise law for wind turbines. Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL, 61821, schomer@schomerandassoc.com)

The noise generated by wind turbines, termed wind energy conversion systems (WECSs), appears to be a growing problem worldwide. The WECSs seem to be especially a problem in quiet rural areas, areas where there is known to be a greater expectation for peace and quiet than in other areas. This paper presents an ordinance for use in such areas when appropriate vis-à-vis state and/or local laws and regulations. The ordinance includes metrics, criteria, background determination, WECS noise prediction, and enforcement. The ordinance is universally applicable to communities experiencing WECS noise, except for the criteria which are geared toward quiet rural areas, but are easily adjusted to less noise sensitive areas on the basis of the ambient in these areas and on the basis of applicable national and international standards.

Contributed Papers

9:20

3aNSa5. Wind turbine noise ordinances: A review of selected state and local regulations. Isaac H. C. Old, Emily J. Eros, and Eddie C. D. Duncan (Resource Systems Group, 55 Railroad Row, White River Junction, VT 05001, iold@rsginc.com)

Noise regulations can often have a great impact on the siting of wind turbines. In the United States, there are no national wind turbine noise regulations. State standards are often not in place, leaving wind turbine siting regulations up to local authorities. This presentation discusses an assortment of both state and local standards from the United States, their implications, and from what basis the ordinances are developed.

9:35

3aNSa6. Planning for small wind turbines in California. John C. Bennett (Dept. of Planning and Land Use, County of San Diego, 5201 Ruffin Rd., Ste. B, San Diego, CA 92123)

A case study in the development of regulatory noise standards for wind energy projects is presented for a California county. The discussion will focus on small wind turbines, their setbacks for safety and noise effects, acceptable heights, and related installation issues. A summary of possible guidelines and field studies points out the ongoing concerns that exist over this topic in California and the rest of the United States.

9:50

3aNSa7. Prediction of flow induced noise near a rigid body at low Mach numbers. Yiping Wang, Jun Chen, Kai Ming Li (Dept. of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031), Yiping Wang, and Zhengqi Gu (Hunan Univ., Changsha, Hunan 410082, China)

For low Mach number flow around a solid body, the far-field sound pressure is dominated by dipole sources comparing with the monopole and quadruple components. The dipole sources are created by the unsteady surface pressure distributions due to the presence of a turbulent boundary layer in the vicinity of the solid body. The far-field sound pressure can be calculated by integrating the time derivative of the wall-pressure fluctuations at the surface of the rigid body. Recent development in computational fluid dynamics (CFD) brings a powerful tool for predicting unsteady turbulent flow fields and the generation mechanism of aerodynamic noise. The current study investigates the use of improved CFD method to accurately simulate the flow fields near a solid body. The information of the flow field is then used to calculate the far field sound pressures. Extensive computations have been conducted to calculate the noise induced by the flow over a circular cylinder and an automobile's rear view mirror. The calculated sound pressure levels have shown to agree well with published experimental results. [Work sponsored by the China Scholarship Council.]

WEDNESDAY MORNING, 25 MAY 2011 WILLOW B, 10:25 A.M. TO 12:00 NOON

Session 3aNSb

Noise and Architectural Acoustics: Second Life Structures: Issues in the Repurposing of Buildings

Norman H. Philipp, Chair
Univ. of Kansas, School of Architecture, Lawrence, KS 66045

Chair's Introduction—10:25

Invited Papers

10:30

3aNSb1. Isolation strategies and results in repurposed spaces for theater and music. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Three repurposed buildings are examined with attention to noise isolation approaches and structural issues. Two black box theaters, one inside a brick powerhouse, and a second inside an early 1900s dorm that includes new classrooms, are observed. A third project is an early 1900s music building gutted and redesigned for integrated recording and improved isolation. Additional items of interest in regard to mechanical noise, acoustics, and historical commission limitations are presented.

10:50

3aNSb2. Case study: 1920's elementary school gymnasium to church sanctuary. Norman H. Philipp (School of Architecture, Design & Planning, Univ. of Kansas, 1465 Jawhawk Blvd., Lawrence, KS 66045, nphilipp@ku.edu) and Robert C. Coffeen (Univ. of Kansas, Lawrence, KS 66045)

Case study of the repurposing of Hawthorne Elementary School in Ottawa Kansas to a house of worship. The school was built and opened in 1926, served as an elementary school for 81 years, was closed in 2007, and sold to Grace Community Fellowship. The focus for this paper will be on the conversion of the existing gymnasium and attached stage platform to the main worhip space for Grace Community Fellowship. Aspects discussed include the architectural acoustics, mechanical noise control, and sound system integration.

11:10

3aNSb3. Case studies of adaptive acoustical reuse of three historic buildings. Gary W. Siebein and Hyun Paek (Siebein Assoc., Inc., Consultants in Architectural Acoust., 625 NW 60th St., Ste. C, Gainesville, FL 32607)

Case studies of the acoustical reuse of three historic buildings all included on the National Register of Historic Places will be presented. Case study 1 is a university library space that was reused as a small performance hall and meeting room. Acoustical interventions in the renovation were extremely limited. Innovative diffusing panels were custom designed and fabricated to fit within existing wood panel dimensions on the walls of the performance area. Sound absorbing materials that had to match the color and dimensions of existing materials were also strategically added to the space. Case study 2 is the conversion of a medium sized church to a performance hall. Impulse response measurements showed sound focusing difficulties in the existing building that had to be reduced through the strategic insertion of sound diffusing systems. Case study 3 was the conversion of old university buildings into state of the art classrooms and faculty offices. The integration of modern air-conditioning systems as well as airborne and impact sound control systems within the historic fabric of the building, was a major design issue.

Contributed Papers

11:30

3aNSb4. Contribution of non-verbal and non-ventilation noise sources to background noise levels in elementary school classrooms. Alex Hornecker, Clothilde Giacomoni, Michelle C. Vigeant, and Robert D. Celmer (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu)

The goal of this project was to determine the effect of hard versus soft flooring on overall speech and activity noise levels in elementary classrooms. Long-term calibrated sound recordings were measured in second and fifth grade classrooms. Within each grade level, two different classrooms were used: one with vinyl composition tile (VCT) flooring and one with short-pile rubber-backed commercial carpeting. The same students circulated between these rooms by grade level, providing a means of comparing sound levels generated by the same population on different floor surfaces. The VCT and carpeted classrooms had similar floor area, layout, and room volume. Recordings were edited to parse the calibrated WAV files into separate segments of (a) teacher/student speech (without other activity) and (b) classroom activity noise including footfalls, chair scrapes, and impacts (no speech). It was found that the type of flooring made a significant difference in measured activity noise. Increased levels found in the VCT classrooms were attributed more to interactive effects with the floor types than to differences in total room acoustic absorption. The third-octave bands where significant activity noise levels occurred varied by age group. Implications for future floor treatments are discussed. [Work supported by Paul S. Veneklasen Research Foundation.]

11:45

3aNSb5. Design of an anechoic termination for automotive applications. Kyle R. Myers and Koorosh Naghshineh (Dept. of MAE, Western Michigan Univ., 4601 Campus Dr., Kalamazoo, MI 49008, kyle.r.myers@wmich.edu)

In various automotive applications, the acoustic performance of tuning devices is assessed by measuring the transmission loss (TL) using the decomposition method [as described by Tao and Seybert (2003)]. The decomposition method, which "decomposes" the pressure wave into its incident and transmitted components, assumes an anechoic termination. However, the reliability of measurements using the decomposition method is compromised in the absence of a good anechoic termination. Thus, there is a need to accurately predict the performance of anechoic terminations. This paper constructs a finite element model in ANSYS of a typical anechoic termination consisting of a horn terminating into a pipe filled with absorbing material in order to predict the reflection coefficient across a broad range of frequencies. The model is compared to analytical predictions and experimental measurements of a prototype with similar geometry. This study could lead to further work towards optimization of horn geometry and the bodies they terminate into.

Session 3aPP

Psychological and Physiological Acoustics and Engineering Acoustics: Audio-Tactile Interactions: Recent Anatomical, Physiological, and Perceptual Research

Charlotte M. Reed, Cochair
Massachusetts Inst. of Technology, Cambridge, MA 02139

Louis D. Braida, Cochair
Massachusetts Inst. of Technology, Research Lab. of Electronics, Cambridge, MA 02139

Chair's Introduction—8:30

Invited Papers

8:35

3aPP1. Auditory-somatosensory integration in the auditory brainstem and its alteration after cochlear damage. Susan E. Shore (Kresge Hearing Res. Inst., Dept. Otolaryngol., Univ. of Michigan, 1150 W. Medical Ctr. Dr., Ann Arbor, MI 48109)

In addition to auditory-nerve inputs, the cochlear nucleus (CN) receives extensive non-auditory inputs from somatosensory ganglia and brainstem nuclei [Shore *et al.*, JCN (2000); Zhou and Shore JCN, (2007)]. The effects of stimulating these pathways on CN physiology are examined using multichannel technology to measure single- and multi-unit activities across populations of neurons in both the dorsal and ventral divisions of the guinea pig CN. Neurons in both CN divisions show evidence of multisensory integration by exhibiting non-linear summation of auditory nerve and somatosensory inputs. Thus, somatosensory neurons can suppress or enhance acoustically evoked response rates as well as altering their temporal patterns. These changes may explain why some patients with tinnitus can modulate the loudness and pitch of their tinnitus by somatic maneuvers of the head and neck. Exploring how these pathways change after various forms of deafness allows us to delve into the realm of neural plasticity, hyperacusis, and tinnitus. Noise induced deafness renders CN neurons more sensitive to somatosensory stimulation as well as increasing the spontaneous rates of those neurons responsive to these inputs [Shore *et al.*, EJN (2008)], findings that can be explained by cross-modal reinnervation [Zeng *et al.*, J. Neurosci. (2009)].

9:00

3aPP2. Neuronal mechanisms and perceptual significance of somatoauditory interactions in primate A1. Charles Schroeder (Nathan S. Kline Inst. for Psychiatric Res., 140 Old Orangeburg Rd., Orangeburg, NY 10962, schrod@nki.rfmh.org)

Recent anatomical, physiological, and neuroimaging findings indicate multisensory convergence at early, putatively unisensory stages of cortical processing. We sought to confirm somatosensory-auditory interaction in AI, and to define both its physiological mechanisms and its consequences for auditory information processing. Laminar current source density and multiunit activity were sampled during multielectrode penetrations of primary auditory area AI in awake macaques revealed clear somato-auditory interactions, with a novel mechanism: Somatosensory inputs appear to reset the phase of ongoing neuronal oscillations, so that accompanying auditory inputs arrive during an ideal, high excitability phase, and produce amplified neuronal responses. In contrast, responses to auditory inputs arriving during the opposing low-excitability phase tend to be suppressed. We speculate that these effects are likely mirrored by corresponding changes in the gain of the perceptual representation. Our findings underscore the instrumental role of neuronal oscillations in sensory operations. The timing and laminar profile of the multisensory interactions in AI indicate that nonspecific thalamic systems may play a key role in the effect.

9:25

3aPP3. Examining influences of deafness on behavioral and cortical responses to vibrotactile speech stimuli. Edward T. Auer, Jr. (Dept. of Speech-Lang.-Hearing, Univ. of Kansas, 1000 Sunnyside Ave., Lawrence, KS 66045, auer@ku.edu)

Deaf individuals can rely on somatosensation for the perception of acoustic stimuli. Research into both natural and artificial approaches to support this sensory substitution has led to the observation of enhanced vibrotactile processing in some deaf individuals. Two fundamental questions arise related to this observation. First, what in the experience of deafness could have led to the enhancement of somatosensory processing? Second, what, if any, cortical changes are associated with the processing of somatosensory stimuli in deaf individuals? With regard to the first question, evidence will be presented that suggests that the enhancement may arise as a result of the significant amounts of vibrotactile stimulation some deaf individuals receive from wearing high-powered hearing aids. With regard to the second question, evidence will be presented that suggests that life-long experience with high-powered hearing aids is also associated with the observation of auditory cortex activation by somatosensory stimuli in deaf individuals. Taken together these results support experience driven changes in somatosensory processing of acoustic speech stimuli and associate auditory cortex activation with these changes. These results will be discussed within the context of implications for multisensory processing following cochlear implantation.

9:50

3aPP4. Pitch and loudness interactions between audition and touch. Jeffrey M. Yau (Dept. of Neurology, Johns Hopkins Univ. School of Medicine, 1620 McElderry St., Reed Hall 2E-2218A, Baltimore, MD 21205, yau@jhu.edu), J. Bryce Olenczak (Univ. of Maryland School of Medicine, Baltimore, MD 21201), Alison I. Weber, J. Frank Dammann, and Sliman J. Bensmaia (Univ. of Chicago, Chicago, IL 60637)

Sensory signals combine and interact to produce stable representations of our environment. Audition and touch both convey information about environmental oscillations. We hypothesized that the two modalities interact in the perception of sounds and vibrations. We conducted a series of psychophysical experiments to explore how auditory and tactile signals interact in frequency and intensity perception. We found that audio-tactile pitch interactions are bidirectional, as distractors influence frequency perception regardless of the attended modality. In contrast, we found that loudness interactions depend on the attended modality: Tactile distractors influence judgments of auditory loudness, but judgments of tactile intensity are impervious to auditory distractions. We also found that audio-tactile pitch and loudness interactions differ in their sensitivity to stimulus timing. These results reveal that auditory and tactile inputs combine differently depending on the perceptual task. That distinct rules govern the integration of auditory and tactile signals in pitch and loudness perception implies that the two rely on separate neural mechanisms.

10:15—10:30 Break

10:30

3aPP5. Feeling sounds: Auditory influences on touch perception. Tony Ro (Dept. of Psych., NAC 7/120, 160 Convent Ave., New York, NY 10031, tro@ccny.cuny.edu)

Events occurring in one sensory modality can often affect how we experience sensations in a different sensory modality. For instance, the sound of a mosquito flying near us seemingly enhances sensitivity to touch on our skin. In this talk, I will describe a series of experiments that examines the effects of audition on touch perception, as well as some of the neural substrates underlying them. In addition to describing psychophysical experiments that have revealed systematic ways in which audition influences touch perception, I will discuss a set of experiments using diffusion tensor imaging and functional magnetic resonance imaging that were conducted in neurologically normal subjects as well as on a patient who has a unique form of acquired auditory-tactile synesthesia. Together, these studies illustrate some of the dynamic interactions between our different sensory systems, the neural mechanisms underlying these interactions, and also highlight the plastic properties of sensory processing in the human brain.

Contributed Paper

10:55

3aPP6. Auditory feedback from tactile interrogation of two dimensional scene for visually impaired. Pubudu Madhawa Silva, Thrasyvoulos N. Pappas (Dept. of Elec. Eng. and Comput. Sci., Northwestern Univ., Evanston, IL 60208-3118), Josh Atkins, and James West (Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218)

With the ever increasing availability of the internet and electronic media rich in graphical and pictorial information (for communication, commerce, entertainment, art, and education), it has been hard for the visually impaired community to keep up. We propose a non-invasive system that can be used to convey graphical and pictorial information via hearing and touch. The main idea is that the user actively explores a two-dimensional layout on a touch screen (with or without tactile overlay) with the finger while listening to spatially conditioned auditory feedback with unique sound assigned and each region representing an object. The proposed approach was successful in a range of tasks, from basic shape identification to perceiving a scene with several objects. While this can represent a revolution for the visually impaired population, sighted can also benefit from the technology in situations they are unable to utilize vision and in areas such as virtual reality, immersive environments, and medicine.

11:10—11:40 Panel Discussion

Session 3aSC

Speech Communication: Modeling and Measurements of Atypical and Typical Speech (Poster Session)

Kelly L. Tremblay, Chair
Univ. of Washington, Dept. of Speech and Hearing Sciences, 1417 N.E. 42nd St., Seattle, WA 98015-6246

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

3aSC1. The effect of atypical nasal resonance on intelligibility in sentence-length stimuli. Peter Watson and Bartek Plichta (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, pjwatson@umn.edu)

Kent and Rosenbek [(1989)] hypothesized that abnormally high nasal resonance (HNRS) due to a paretic or paralyzed velopharyngeal mechanism in speakers with dysarthria could worsen intelligibility. Contrastingly, Whitehill's [(2000)] review of literature found few that believe that HNRS associated with cleft palate contributes very little to a worsening of intelligibility. Determining the effect of atypical nasality on intelligibility is difficult to determine, because HNRS in disordered speech populations is almost always accompanied by other speech/voice abnormalities. A method of synthesizing different degrees of nasal resonance to sentence-length stimuli has been developed and successfully tested for perceptual adequacy. This method was used to synthesize three degrees of nasal resonance (none, mild, and moderate) in 30 sentences spoken by a healthy speaker. The sentences contained only oral phonemes. The sentences were normalized for intensity and mixed with noise and presented to 30 listeners over headphones. The listeners were instructed to orthographically transcribe what they heard. The sentences without nasal resonance were 10%–15% more intelligible than those sentences with increased nasal resonance.

3aSC2. The effect of the frequency of deep-brain stimulation on fine-grained acoustic features of speech. Yu-Wen Chen, Peter Watson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55447, pjwatson@umn.edu), Erwin B. Montgomery, Jr. (Dept. of Neurology, SC 360A, 1720 7th Ave S, Birmingham, AL 35294-0017, emontgom@uab.edu), and Harrison Walker (Dept. of Neurology, Birmingham, AL 35294-0017, harrison.walker@gmail.com)

Deep-brain stimulation (DBS) has been used for approximately 1 decade to ameliorate the symptoms of Parkinson's disease (PD). DBS often ameliorates many of the atypical motor symptoms of PD—with the exception of speech production. Clinically significant improvement of speech is not realized or in some cases is worsened [Tornquist *et al.* (2005). One potential explanation of this paradox is that the DBS settings used to improve limb and truncal control are set for force generation, but not for the complex agonist-antagonist interplay that is required for speech [Montgomery, (2008)]. The data from six patients with PD who have implanted DBS systems will be presented. Each patient produced sustained vowels, sustained fricatives, and oral reading of sentences under four different DBS frequency settings. Data show that for some settings patients were able to increase loudness only, but for other settings loudness is slightly reduced but vowel formant space is increased and spectral kurtosis of fricatives is decreased. The experimenters performing the acoustic analysis were blind to the frequency settings.

3aSC3. Temporal structure in the speech of a person with dementia: A longitudinal case study. Linda Carozza, Margaret Quinn, Julia Mack, and Fredericka Bell-Berti (Dept. of Commun. Sci. and Disord., St. John's Univ., 300 Howard Ave., Staten Island, NY 10301)

Cognitive and language processes in dementia have been studied extensively, but motor speech degeneration in the course of dementing illness has been relatively unexplored. The potential for early dissociation of motor functions of language at the level of speech production has not been explored; an interaction between motor speech and language production and perception changes should inform our understanding of the deterioration in dementia. We have previously reported results of a preliminary study of the temporal structure in the speech of persons with dementia. In that study, one of the participants showed inconsistent final lengthening and effects of final consonant voicing on vowel duration, as well as a VOT pattern that suggested a reduced distinction between American English /b/ and /p/. This study explores changes in the temporal structure of speech of that participant, who was recorded 6 months after the first session. He produced a series of short phrases containing a target word occurring in phrase-medial or -final position; the target word began with a fricative or voiced or voiceless stop consonant and ended with /t/ or /d/. We will report on temporal pattern changes in VOT, phrase-final lengthening, and the effect of final consonant voicing on vowel duration.

3aSC4. Relative fundamental frequency as an acoustic correlate of vocal effort in spasmodic dysphonia. Cara E. Stepp (Dept. of Comput. Sci. Eng. and Rehabilitation, Univ. of Washington, Box 352350, Seattle, WA 98195) and Tanya L. Eadie (Univ. of Washington, Box 354875, Seattle, WA 98195)

Individuals with vocal hyperfunction have lowered relative fundamental frequency (RFF) in vowels immediately preceding and following a voiceless consonant, but the relationship between RFF and perceived voice quality has yet to be studied in neurogenic voice disorders. The purpose of this study was to determine the relationship between RFF with listeners' perceptions of 19 individuals with adductor spasmodic dysphonia. Speech recordings were presented to 20 inexperienced listeners who evaluated effort and overall severity using visual analog scales. The correlation coefficient (R^2) between average effort and severity measures and an RFF measure was calculated as a function of the number of acoustic instances used for the RFF estimate (1–9, of a total of nine voiced-voiceless-voiced instances). Increases in the number of acoustic instances used for the RFF average led to increases in the correlation with perceptual measures, with use of six or more instances resulting in a stable estimate. Correlation coefficient values were highest between effort and RFF (range R^2=0.06–0.43), with slightly lower correlation coefficients between severity and RFF (range R^2=0.06–0.35). This study indicates that RFF measures are correlated with perceived voice quality and that guidelines should be developed for calculating reliable estimates of RFF in future research.

3aSC5. Envelope modulation spectrum: Exploring the challenges to intelligibility of dysarthric speech. Rene L. Utianski, Kaitlin Lansford (Dept. of Speech Hearing Sci., Arizona State Univ., Coor Hall, P.O. Box 870102, Tempe, AZ 85287, rutiansk@asu.edu), Andrew Lotto, and Julie M. Liss (Arizona State Univ., Tempe, AZ 85287)

This study explores whether information obtained from the envelope modulation spectrum (EMS) can predict the type of intelligibility decrement experienced by listeners of dysarthric speech. Previously, Liss *et al.* [J. Speech Lang. Hear. Res. **53**, 1246–1255 (2010)] demonstrated that spectral

information obtained via EMS can classify individuals with different patterns of degraded speech into their appropriate clusters. Variables noted as significant in discriminating dysarthria types, such as relative energy above and below the region of 4 Hz, are of particular interest. While the analyzes showed variations of energy in certain octaves of the acoustic signal were critical for this analysis, it is unclear which perceptual components map onto these measures. Using performance measures that capture listeners' strategies to understand speech at the segmental, suprasegmental, and phonetic levels (intelligibility, lexical boundary errors, and word substitutions), regressions will be performed to determine which information contained within the EMS metrics can account for the variability in listener performance at various levels. Ultimately, this will allow us to look at trends in the EMS output, identify the source of intelligibility decrement, and predict listener performance on outcome measures. Results will bear on models of connected speech perception as well as clinical practice.

3aSC6. Acoustic variability of speakers with dysarthria on word repetition tasks. Yunjung Kim (Dept. of Commun. Sci. Disord., LSU, Baton Rouge, LA 70803, ykim6@lsu.edu)

Although large variability in acoustic characteristics "across" speakers with dysarthria has been reported in previous studies in the process of describing characteristics of dysarthrias and identifying speech severity, relatively less attention has been paid to acoustic variability "within" speakers with dysarthria [Kim *et al.*, JSLHR (in press)]. This is a theoretically and clinically important question regarding validity and reliability of acoustic parameters as markers of speech disorders or severity, especially given the difficulty in data collection due to the physical condition of speakers and the relatively small number of subjects and trials (repetitions of given items) in previous literature. As a follow-up analysis of Kim *et al.* (in press), this study aims to examine the degree to which speakers with dysarthria exhibit consistency in repeated articulatory behaviors by measuring variability of selected acoustic measures on word repetition tasks. Acoustic measures include F2 slope for vocalic nuclei and moment analysis for fricatives /s/ and /ʃ/ produced by 108 speakers with dysarthria and ten healthy controls. These results will be interpreted in terms of dysarthria in general and the different types of dysarthria that are said to be associated with different types of neurological disease.

3aSC7. Prosodic characteristics of speakers with schizophrenia disorders: Preliminary data. Ali E. Beslin (2153 Wisteria St., Baton Rouge, LA 70806, abesli1@tigers.lsu.edu), Yunjung Kim, and Alex S. Cohen (Louisiana State Univ., Baton Rouge, LA 70803)

This study aims to provide preliminary data on prosodic characteristics of speakers with schizophrenia as the first phase of a larger study that explores acoustic vocal markers of schizophrenia for the ultimate purpose of differential diagnosis of this disorder. Only a few studies have reported acoustic characteristics of individuals with schizophrenia including flattened intonation [Covington *et al.* (2005)] and an increase in pause duration [Rapcan *et al.* (2010)] compared to healthy speakers. For this presentation, 100-s-long spontaneous speech samples produced by three female speakers with schizophrenia were analyzed and results were compared to three female speakers with bipolar disorder as a psychiatric control group. Speech recordings were first segmented into a total of 165 breath groups, and then from each breath group, acoustic measures were made including (1) mean and range of fundamental frequency contour, (2) mean and range of intensity contour, (3) duration and frequency of pauses, and (4) speaking/articulation rate. It is expected that the results from these preliminary data will further motivate investigations of negative symptoms of speakers with schizophrenia and the development of automated tools for differential diagnosis of schizophrenia from other psychological conditions known to exhibit symptoms similar to those in schizophrenia such as depression and anxiety.

3aSC8. The interaction between tone and intonation in Mandarin aphasics speech. Shang-Chen Chiu, Naomi Ogasawara (Dept. of English, Natl. Taiwan Normal Univ., No. 162, Sec. 1, He-Ping E. Rd., Taipei 106, Taiwan, Republic of China, 496211295@ntnu.edu.tw), and Yuh-Mei Chung (Taipei Veterans General Hospital, Taipei City 11217, Taiwan, Republic of China)

This study addresses two issues: (1) the interaction of tone and intonation in Mandarin and (2) whether nonfluent aphasic patients show the same pattern of the interaction of these two prosodic features as normal speakers do. Both tone and intonation use pitch change at different phonological levels. Intonation is superimposed on a phrase or a sentence, which must affect the pitch change of tones. In order to examine the interaction of tone and intonation, the study focuses on Mandarin disyllabic words with tone 2-2 (rising) and tone 4-4 (falling) in declarative (falling intonation) and interrogative (rising intonation) forms. An acoustic analysis tests how the F0 of the rhyme is affected by the two prosodic features. The investigation involves two groups of Taiwan Mandarin speakers: normal people and aphasic patients. The previous studies have tested non-tonic language speakers and found that the Broca's aphasia, a nonfluent aphasia subtype, usually shows motor speech disorder, which may cause the phonetic-motoric impairment. In order to provide findings from a tonic language, this study investigates whether aphasic patients of a tone language, Mandarin, show the similar dysprosodic features.

3aSC9. Contribution of vowel distinctiveness to intelligibility and vowel identification accuracy of dysarthric speech. Kaitlin L. Lansford, Julie M. Liss, Rene L. Utianski, Tamiko Azuma, Michael F. Dorman (Dept. of Speech and Hearing Sci., Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287, kaitlin.lansford@asu.edu), and Andrew J. Lotto (Univ. of Arizona, Tucson, AZ 85721)

Distorted vowel production is a hallmark characteristic of dysarthric speech and often is characterized by spectral and temporal degradation, flattening of spectral change formants, and vowel space distortions that may differentially affect high versus low or front versus back contrasts. Because acoustic information critical to accurate speech perception is contained in vowels, it is important to ask how such degradations influence the resulting percept. To date, attempts to characterize this relationship have met with mixed results. In the present investigation, a variety vowel space metrics were derived from vowels produced by 48 speakers with dysarthria to assess their ability to predict perceptual outcome measures (e.g., intelligibility and vowel identification accuracy). Novel metrics capturing distinctiveness of spectral neighbors (i.e., dispersion of vowels) were included in this analysis, as recent evidence suggests that vowel distinctiveness, not vowel space area, better predicts vowel intelligibility [Neel, J. Speech Lang. Hear. Res. **51**, 574–585 (2008)]. Results of several stepwise multiple regressions support these findings as measures of vowel dispersion both novel and established were included in models that significantly predicted both intelligibility and vowel accuracy. Results of the present analysis provide compelling support of inclusion of such dispersion metrics in the study of dysarthric vowel perception.

3aSC10. A portable, real-time vocoder: Technology and preliminary perceptual learning findings. Elizabeth Casserly (Dept. of Linguist., Indiana Univ., Memorial Hall, Rm 322, Bloomington, IN 47405, casserly@indiana.edu), David B. Pisoni (Indiana Univ., Bloomington, IN 47405), Christopher Smalt, and Thomas Talavage (Purdue Univ., West Lafayette, IN 47907)

Transformed or vocoded acoustic signals paralleling the processing of cochlear implants have proven extremely useful for simulating aural perception in deaf CI users with normal hearing participants. The benefits of current perceptual studies with vocoded stimuli are limited, however, in their ecological validity and direct applicability to the everyday challenges faced by CI users. Normal hearing subjects listen for relatively short periods of time to isolated, prerecorded words or sentences; therefore, their perceptual learning occurs without the semantic context, visual support, and conversational interplay that normally accompanies spoken language use. In this paper, the development of a new device, which enables vocoded speech research to overcome many of these limitations, is described. This technology carries out vocoding signal transformations in real-time, with very short delays, and is small enough to fit in a participant's pocket. The portable, real-time vocoder (PRTV) therefore allows subjects to experience vocoded acoustics while freely interacting with their environment and the people in it, potentially for extended periods of time. Preliminary investigation of perceptual learning with the PRTV has demonstrated its efficacy: improvements of up to 15% in open-set word recognition accuracy were found in 3 NH listeners after 1.5 h of interactive use.

3aSC11. Prosodic characteristics of speech directed to adults and to infants with and without hearing loss. Laura C. Dilley (Dept. of Communicative Sci. and Disord., Michigan State Univ., 116 Oyer, East Lansing, MI 48824, ldilley@msu.edu), Evamarie Cropsey (Michigan State Univ., East Lansing, MI 48824), Maria V. Kondaurova, and Tonya R. Bergeson (Indiana Univ. School of Medicine, Indianapolis, IN 46202)

Studies have shown that infant-direct speech (IDS) is distinguished from adult-directed speech (ADS) via a variety of acoustic-prosodic characteristics, including fundamental frequency and rate. However, little is known about how IDS and ADS may be distinguished in terms of linguistically relevant prosodic constructs, such as pitch accents, nor how these constructs map onto acoustic differences previously identified for IDS vs ADS. To investigate these issues, longitudinal recordings of mothers reading a storybook to their typically developing infants and to an experimenter were made when the infants were approximately 3, 6, or 9 months old. Trained analysts used an annotation system to code prosodic characteristics, including rhythmic prominences, pitch accents, and intonation phrase boundaries. Preliminary analyzes of labeled corpora revealed that mothers produced significantly more prominences in IDS than in ADS. In addition, mothers showed substantial individual differences in rates of prominence production in IDS and ADS. Future work will include acoustic analyzes of labeled prosodic constructs, as well as investigations of how prosodic variation in mothers' speech may predict speech-language outcomes in both normal-hearing children as well as children with hearing loss. [Work supported by NIH-NIDCD Grant No. R01DC008581.]

3aSC12. The intelligibility of clear speech at normal and slow rates for older hearing-impaired listeners. Jean C. Krause, R. Ann Siapno, and Ari B. Hansen (Dept. of Comm. Sci. and Dis., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, jeankrause@usf.edu)

Sentences spoken "clearly" are significantly more intelligible than those spoken "conversationally" for normal-hearing and hearing-impaired listeners in a variety of listening environments [e.g., Payton *et al.*, J. Acoust. Soc. Am. **95**, 1581–1592 (1994)]. Typically, clear speech is also spoken more slowly than conversational speech [Picheny *et al.*, J. Speech Hear. Res. **29**, 434–446 (1986); Uchanski *et al.*, *ibid.* **39**, 494–509 (1996)]. However, talkers can produce a form of clear speech at normal rates that provides a large intelligibility advantage to young normal-hearing listeners in noise [Krause and Braida, J. Acoust. Soc. Am. **112**, 2165–2172 (2002)]. To determine whether this form of clear speech provides a similar benefit to older hearing-impaired (OHI) listeners, this experiment examined the intelligibility of conversational and clear speech at normal and slow rates (conv/normal, clear /normal, conv/slow, and clear/slow) for 11 OHI listeners with symmetric, sloping, moderate SNHL. Although clear speech at normal rates provided only a very small benefit on average, large benefits (comparable to clear /slow speech) were obtained in certain talker/listener combinations. For two of the four talkers, the intelligibility advantage was consistent across listener, suggesting that the benefit of clear speech at normal rates for this population may be talker-dependent.

3aSC13. An investigation of speech production and intelligibility of a post-lingually deaf sequential bilingual adult cochlear implant user, pre- and post-implantation for English and Twi. Stephanie J. Scibilia (Dept. of Commun. Sci. and Disord., Long Island Univ.-Brooklyn Campus, 1 University Plaza, Brooklyn, NY 11201, sscibilia637@gmail.com)

This study investigates changes in speech production for a sequential bilingual adult male who is postlingually deaf (9 years of hearing) following cochlear implantation (CI). Recordings of words in carrier phrases took place at pre-implantation in English (L2) and at the 6-month, 1-year, 1-year-6-month, and 2-year post-implantation marks in L2 and Twi (L1). Normative data were recorded from a typically hearing adult male with a similar linguistic history. Acoustic measurements for L2 include f0, F1, and F2 of vowels, VOT, and spectral measures of s-S. Voice quality measurements for sustained /a/ were also obtained. Acoustic measurements for L1 include F1 of vowels that undergo vowel harmony and f0 of tonal contrasts. Questionnaires about CI usage and self-reported changes in production/perception supplement the acoustic data. Lastly, forced-choice speech intelligibility ratings in L1 and L2 were obtained utilizing inexperienced monolingual listeners of each language separately. Current results suggest preservation of L1 following 16 years of auditory deprivation, phonetic influence of L1 on L2, and fluctuating suprasegmental parameters. These acoustic data will be compared to intelligibility ratings and information from the self-report questionnaires.

3aSC14. Female voice age classification based on acoustic characteristics of speech. Eun-Min Lim and Jeung-Yoon Choi (Dept. of Elec. and Electron. Eng., Yonsei Univ., 134 Shinchon-dong, Seodaemun-gu, Seoul 120-749, Korea, vision8611@dsp.yonsei.ac.kr)

As there are many changes in vocal apparatus caused by aging, many acoustic differences also can be found in speech. These differences show different tendencies for male and female speakers and are more pronounced for females. Therefore, this paper focuses on female speaker age analysis and classification. To analyze acoustic differences, global measurements related to the vocal source such as band energy, jitter, shimmer, open quotient, and spectral tilt are used. Additionally, local features that are extracted near vowel landmark regions are also used. Appropriate age boundaries are selected based on the results of analysis, and feature selection is carried out according to the results of analysis of variance (ANOVA) test for classification. The proposed age classification system is modeled using Gaussian mixture models (GMMs), and the sequential feature selection method is used. Experiments are performed on the aGender corpus provided in the INTERSPEECH 2010 Paralinguistic Challenge.

3aSC15. Comparing voice quality measures from steady-state phonation and continuous speech. Bruce R. Gerratt and Jody Kreiman (Div. of Head /Neck Surgery, Univ. of California, Los Angeles, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, bgerratt@ucla.edu)

The debate over the equivalence of voice quality measures derived from continuous speech vs steady state vowels is founded in the various definitions of voice quality and the resulting uncertainty about which aspects of speech measures of quality are most important. Measures derived from steady-state phonation correspond to narrow definitions of voice quality as the perceptual consequence of vocal fold vibration. Measures from continuous speech derive from broader conceptions of voice quality as essentially synonymous with speech encompassing articulation, unvoiced portions of utterances, and sentential prosody. To complicate this matter, quality measures from continuous speech may reflect gestures related to linguistic voicing contrasts (e.g., breathiness near /h/). We will argue that measures from continuous speech depend on the measurement of vocal quality within the vowels: Because vowels form the largest voiced portion of continuous speech, any measure of quality must quantify their sound. To assess the perceptual equivalence of continuous speech vs steady-state phonation, we therefore used analysis by synthesis to compare steady-state vowels to vowels extracted from continuous speech, for 20 speakers with vocal pathology. Results will be discussed with respect to the overall manner in which quality should be quantified. [Work supported by NIH.]

3aSC16. Dependence of long-term-average spectrum curves of voices as function of equivalent sound level. Johan Sundberg and Svante Granqvist (Dept. of Speech Music Hearing, School of Comp. Sci. Commun., KTH, Lindstedsvxe4;gen 24, 10044 Stockholm, Sweden, pjohan@speech.kth.se)

Long-term-average spectrum (LTAS) is a quick and simple analysis method, which is frequently used in voice analysis. It typically reaches a stable curve after about 30 or 40 s of speech or singing and reflects both voice source and formant frequency characteristics of a voice. However, it is quite sensitive to changes of vocal loudness. In two previous investigations a set of recordings were analyzed in which 15 male and 16 female adults read a standard text aloud at seven different degrees of vocal loudness. The results showed that the LTAS curve changed linearly but in a frequency dependent manner with the equivalent sound level Leq. Furthermore, the alpha ratio between the summed energy below and above 1 kHz was found to be a quadratic function of Leq. In the present investigation we study the relation between Leq and (1) the overall LTAS slope between 0.75 and 4 kHz and (2) the average increase in LTAS level, in dB, in the same range. While

slope varied considerably between speakers, average LTAS level increase could be accurately described by a linear function of Leq which is similar between speakers. Possible applications of these findings will be discussed.

3aSC17. The influence of flow model simplification on phonation onset. Syed F. Rehman and Zhaoyan Zhang (School of Medicine, Univ. of California Los Angeles, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, chronolz@ucla.edu)

In previous studies, phonation onset characteristics were investigated using linear stability analysis in a continuum vocal fold model. Although this approach allowed efficient calculation of phonation onset, a one-dimensional simplified glottal flow model was used in these previous studies, and it is unclear how phonation onset prediction was affected by this flow model simplification. In this study, to understand the effects of glottal flow simplification on phonation, phonation threshold pressure and onset frequency as a function of vocal fold geometry and stiffness were calculated using a two-dimensional continuum vocal fold model coupled with two-dimensional incompressible Navier–Stokes flow equations. These phonation threshold pressures and onset frequencies were compared to those predicted from a linear stability analysis of the same vocal fold model coupled with a one-dimensional glottal flow model, as in previous studies. The results show that phonation threshold pressures and onset frequencies predicted from these two approaches compared well for large glottal gaps. The difference increased as the glottal gap was constricted. Hence, the results suggest that beyond a certain glottal gap, viscous effects (except flow separation) are negligible and phonation characteristics can be predicted using a simplified and thus computationally efficient glottal flow model. [Work supported by NIH.]

3aSC18. Perceptual sensitivity to changes in vocal fold geometry and stiffness. Zhaoyan Zhang, Jody Kreiman, and Bruce R. Gerratt (UCLA School of Medicine, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794)

Control of vocal quality is a primary goal of speech and the focus of clinical management of voice disorders. Therefore, it can be argued that auditory-perceptual sensitivity to the acoustic changes that occur as a laryngeal parameter is manipulated should be considered in the development of physiologic models of phonation. Listener perception of the effects of change in vocal fold geometry (medial surface thickness and vocal fold depth) and stiffness will be investigated by measuring the just-noticeable difference (JND) to acoustic stimuli generated by physical models differing in these three model parameters. The acoustic pressure produced by physical models differing in small steps in each model parameter will be recorded and normalized for F0. Listeners will hear pairs of these acoustic stimuli in a same/difference task and JNDs will be calculated for each parameter. Data will be interpreted by comparing the JND for each model parameter to the range of values for that parameter. Results will be discussed with regard to which vocal fold parameters are most important for generating perceptually relevant and thus clinically meaningful changes in quality, thus validating their inclusion in physical models of phonation. [Work supported by NIH.]

3aSC19. Acoustic and aerodynamic analyses of transgender voice. Adrienne B. Hancock, Rachael M. Harrington, James Mahshie, and Alicia Lennon (Dept. of Speech and Hearing Sci., George Washington Univ., 2115 G St., NW 201, Washington, DC 20052, hancock@gwu.edu)

Acoustic and aerodynamic data are needed to describe how the vocal mechanism of a transgender person operates to achieve desired gender perception. Early evidence-based practice for transgender voice therapy was limited to changing vocal pitch—at first aiming for the normative values of the desired gender and more recently aiming for more feasible targets of pitch within a "gender-neutral range" established by perceptual research. 20 years later, research is discovering that pitch may be a major influence in gender perception, but is almost certainly not the only influence. In this presentation, the physiology of 20 voices (male-to-female and female-to male) will be illuminated by reporting the glottal closure patterns (e.g., electroglottography), patterns of airflow through the vocal mechanism (e.g., subglottal pressure estimated from intraoral pressure and derived glottal airflow), and resonant qualities of the vocal tract (e.g., formants). Preliminary data suggest that transgender values are not equivalent to a simple average of female and male normative data. Once we learn how the transgender vocal tract can safely overcome constraints of biological anatomy and which parameters contribute most to listeners' perceptions of gender, speech-language pathologists can better guide their clients to speak in an authentic voice.

3aSC20. Curvature of the medial surfaces of the vocal fold, the surface wave model, and phonation threshold pressure. Lewis P. Fulcher and Ronald C. Scherer (Bowling Green State Univ., Bowling Green, OH 43403)

Focusing on characteristics of waves propagating along the medial surfaces of the vocal folds, Titze derived an analytic formula for the phonation threshold pressure (PTP). Among the relationships made explicit by this formula is that converging glottal shapes should have a higher PTP than diverging glottal shapes, for a given glottal width. Although several aspects of Titze's formula have found experimental support, measurements of the angle dependence of the PTP [Chan *et al.*, "Further studies of phonation threshold pressure in a physical model of the vocal fold mucosa," J. Acoust. S. Am. **101**, 3722–3727 (1997)] do not follow the expected trend. Introducing the bulge parameter allows one to examine the role of glottal curvature and thus to explore the effects of more realistic prephonatory glottal shapes. Preliminary calculations for the modified rectangular glottis show that the bulge leads to somewhat higher PTP values for larger bulges, but no distinctive qualitative features. The effects of introducing a bulge for the converging and diverging glottal shapes will be also explored for biomechanical parameters appropriate for human phonation and for the model of Chan *et al.* [Work supported by NIH Grant No. R01DC03577.]

3aSC21. Three-dimensional whole-field measurements of pulsatile glottal jets using synthetic aperture particle image velocimetry. David J. Daily, Tadd Truscott, and Scott L Thomson (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT 84602, thomson@byu.edu)

The use of synthetic aperture particle image velocimetry (SAPIV) to measure the three-dimensional (3-D) jet velocity field generated by the human vocal folds during speech is described. SAPIV uses an array of cameras to acquire 3D flow velocity data throughout a volume of interest. In the present application, eight high-speed cameras are synchronized with a volume-illuminating pulsed laser to image the supraglottal jet exiting from self-oscillating synthetic vocal fold models. The images from the eight cameras are transformed and reconstructed to generate different image planes within the flow field, to which traditional particle image velocimetry (PIV) techniques are applied to calculate 3-D velocity vectors throughout the flow field. Two-dimensional PIV using thin laser sheets has been previously used to study the supraglottal jet, but 3-D measurements have been typically limited to either (1) quasisteady measurements as the laser sheet traverses the glottal jet volume, or (2) 3-D measurements over a narrow subvolume of the glottal jet. SAPIV enables 3-D data to be acquired simultaneously throughout the flow field and over a volume enclosing a large region of the jet. The SAPIV implementation is explained and 3-D data of the glottal jet are presented.

3aSC22. Modeling overall voice quality with a small set of acoustic parameters. Jody Kreiman and Bruce R. Gerratt (Div. of Head/Neck Surgery, Univ. of California, Los Angeles, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Despite the great need for valid measures of voice quality, we are still unable to adequately quantify what a person sounds like. Drawing on recent theoretical and experimental work, we propose assessment of overall quality as a whole, rather than using individual rating scales. This paper describes experiments evaluating a psychoacoustic model that includes vocal tract formant frequencies and bandwidths, four source spectral slope parameters (H1–H2, the slope of the harmonic spectrum from 2–4 kHz and from 450–452 kHz, and the overall spectral slope from H2 to the highest harmonic), the noise-to-harmonics ratio, F0, and slow variability in F0 and amplitude (tremor). To assess the adequacy of this model, we will copy-synthesize 30 pathological voices while manipulating only these parameters. To the extent that listeners judge that the natural and synthetic tokens match exactly, the psychoacoustic model will be considered valid. Discussion will include analysis of mismatches to determine what parameters should be

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added to or subtracted from the model and its application for use in evaluation of disordered voice quality and modes of phonation. [Work supported by NIH.]

3aSC23. Biomechanical modeling of non-stationary vibrations of the pharyngo-esophageal segment. Bjoern Huettner, Sven Herkenhoff, Ulrich Eysholdt (Dept. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Bohlenplatz 21, 91054 Erlangen, Germany, bjoern.huettner@ukerlangen.de), Joerg Lohscheller (Univ. of Appl. Sci. Trier, P.O.Box 1826, 54208 Trier, Germany), and Michael Doellinger (Univ. Hospital Erlangen, Bohlenplatz 21, 91054 Erlangen, Germany)

After total laryngectomy, a substitute voice is generated by means of the tissue vibrations of the pharyngo-esophageal (PE) segment. The quality of the substitute voice significantly depends on the vibration characteristics of the PE segment. A numerical model is presented that allows for the simulation of non-stationary vibrations of the PE segment, i.e. the fundamental frequency varies with time. The introduced time dependent multi-mass model for the PE segment (PE-MMM(t)) has been developed from the non-stationary two-mass model of the vocal folds. The pseudo-glottal opening and vibrations are modeled by 6×2 coupled mass-spring-oscillators placed in two horizontal planes. The masses are arranged to describe enclosed shapes. The model parameters masses, their rest positions, the sub-glottal pressure and the anchor spring couplings are all variable with time. The PE vibrations are described by the time dependent quantities frequency, opening area and pseudo-glottal shape. We will present the PE-MMM(t) that is appropriate to generate self-sustained, non-stationary vibrations. Furthermore, we show dynamics of different model configurations, particular circular and elliptical arranged masses, to simulate different shapes of the PE segment.

3aSC24. Effects of left-right asymmetry in vocal fold geometry and stiffness on phonation onset in a physical model. Abie H. Mendelsohn and Zhaoyan Zhang (Div. of Head and Neck Surgery, Univ. California Los Angeles, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794, amendelsohn@mednet.ucla.edu)

Vocal fold paralysis or paresis often leads to left-right asymmetry in vocal fold geometry and stiffness. An improved understanding of the influence of such left-right asymmetry on phonation would allow clinicians to better diagnose and treat these voice disorders. In this study, the effects of such left-right asymmetry on phonation were investigated using a two-layer physical vocal fold model. Specific vocal fold parameters investigated included body layer stiffness, cover layer stiffness, cover layer thickness, and cover layer depth. Results showed that a unilateral decrease in vocal fold body layer or cover layer stiffness reduced the phonation threshold pressure and onset frequency to a value that was closer to that of the less stiff fold in symmetric conditions. Unilateral thinning of the cover layer depth caused an increase in phonation threshold pressure and onset frequency, whereas unilateral shortening of cover layer thickness led to a threshold pressure close to the averaged threshold pressures of the two unique geometries in symmetric conditions. The clinical implications of the results were also discussed. [Work supported by the NIH.]

WEDNESDAY MORNING, 25 MAY 2011 METROPOLITAN A, 8:00 A.M. TO 12:00 NOON

Session 3aSP

Signal Processing in Acoustics, Physical Acoustics, Underwater Acoustics, and Biomedical Acoustics: Source Localization and Characterization Using Time Reversal Methods

Brian E. Anderson, Cochair
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Sean K. Lehman, Cochair
Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550

Chair's Introduction—8:00

Invited Papers

8:05

3aSP1. Matched signals: The beginnings of time reversal. Clarence S. Clay (Dept. of Geoscience, Univ. of Wisconsin, 1215 W Dayton St., Madison, WI 53706, clay@geology.wisc.edu) and Brian E. Anderson (Brigham Young Univ., N283 ESC, Provo, UT 84602)

This paper discusses the original time reversal experiments as conducted by Parvulescu and Clay. The idea of conducting the first time reversal experiment came about from measurements of the reproducibility of sound transmissions in the ocean in the late 1950s and early 1960s. Then it was believed by many that long range acoustic signal transmissions in the ocean were not reproducible or stable. During a coffee hour in 1960, Parvulescu proposed a simple experiment to create an acoustic matched signal through the use of time reversal, in their coffee room/laboratory area. This represents the first known record of a time reversal experiment. This paper will then discuss a series of experiments that were conducted underwater to communicate between two ships through the use of this matched signal technique. Details are in the theoretical-experimental monograph of Tolstoy and Clay [*Ocean Acoustics* (McGraw-Hill, New York, (1966); J. Acoust. Soc. Am. (1987)]. This paper calls attention to these early experiments and theory as the first demonstrations of the now popular technique termed time reversal.

8:25

3aSP2. Lessons learned from time reversal experiments: Applications to target detection. Hee-Chun Song, William A. Kuperman, William S. Hodgkiss (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238), Tuncay Akal (SUASIS Underwater Systems, Kocaeli, Turkey), and Mark Stevenson (Spawar Systems Ctr., San Diego, CA 92152-5000)

Over a decade beginning in 1996 we carried out a series of time reversal experiments jointly with the NATO Undersea Research Centre in coastal water at various frequencies (450 Hz, 850 Hz, 3.5 kHz, and 15 kHz). These experiments and follow-up analysis suggested potential applications to active sonar. Time reversal focuses acoustic energy on a target enhancing the target echo while minimizing the reverberation from the boundaries below and above the focus, thereby resulting in echo-to-reverberation enhancement. These ideas have been confirmed under limited circumstances in our shallow-water acoustic experiments. This talk will discuss acoustic and signal processing lessons learned from these experiments including an acoustic barrier concept and both active and passive reverberation nulling.

8:45

3aSP3. Time-reversal in seismology. Carene Larmat, Paul Johnson, and Robert Guyer (Los Alamos Natl. Lab., MS D443, Los Alamos, NM 87545)

This talk is a review of the use and history of time-reversal in seismology. time-reversal has been developed independently in the field of acoustics and in geophysics exploration, without the two communities being aware of the fact, as testifies the lack of the cross-references in early papers. The elegance of time-reversal is that it thrives with complexity. Seismologists have to deal with complex signal transmitted through the earth. Moreover, earthquakes are complex sources, involving different episodes of slip along the fault. Early in the development of time-reversal in seismology, emphasis has been put on the need of accurate numerical schemes to back-propagate the wavefield and the need of relatively dense data. In addition to presenting several of the landmark applications of time-reversal in seismology, several of our results will be outlined. In the last decade, our group has studied the potential of TR with the unique approach of combining laboratory experiments with application to seismology problems. Several applications will be presented: location and characterization of the rupture of the 2004 Parkfield earthquake, location, and retrieval of the sliding motion of glaciers in Greenland, finally imaging of the source location of a seismic signal of particularly emergent nature, the tremor.

9:05

3aSP4. Ultrasound focusing to a predefined area using principles of time reversal acoustics. Alexander Sutin (Stevens Inst. of Technologies, Hoboken, NJ 07030), Yegor Sinelnikov (Sound Interventions, Inc., Stony Brook, NY), Gaurav Gandhi (Artann Labs., Trenton, NJ 08618), Natalya Rapoport (Univ. of Utah, Salt Lake City, UT 84112), and Armen Sarvazyan (Artann Labs., Trenton, NJ 08618)

There are several applications of ultrasound that require accurate ultrasound focusing to a predefined area without affecting the surrounding medium. We present two methods of generating high intensity ultrasound field accurately tailored to the shape of the predefined area using the time reversal acoustics (TRA) principles. The first method is based on TRA focusing to the volume distribution of nonlinear sources—ultrasound contract agents (UCAs) placed to sonicated volume. The second method is based on the sensor measurements of a set of impulse responses from a number of points in the sonicated volume. Impulse responses for some of these points can be measured directly and some of them can be based on mathematical interpolation of the measured points. The suggested methods were modeled and the second was tested in the laboratory experiments and is currently being explored for the ultrasound-assisted treatment of pancreatic adenocarcinoma performed by direct injection of a drug into the tumor. The tip of the injection needle includes an ultrasonic sensor providing measurements of impulse responses required for the formation of the desired focal structure. The suggested approach may be applied for other TRA applications including nondestructive evaluation (NDE) and seismic exploration.

9:25

3aSP5. Toward a high power non-contact acoustic source using time reversal. T. J. Ulrich, Pierre-Yves Le Bas, Brian Anderson, and J. James Esplin (Geophys. Group EES-17, Los Alamos Natl. Lab., Los Alamos, NM 87545, tju@lanl.gov)

Over the last decades, nonlinear acoustic techniques have been developed to detect mechanical damage in solids. They have been proven to be far more sensitive to early damage than standard linear acoustic techniques such as C-scans, time of flight, etc. Unfortunately, nonlinear techniques often require high amplitude waves to propagate within the sample. To be practical in industrial applications, signals have to be generated without contact. Currently available non-contact transducers are usually not powerful enough to apply nonlinear techniques. Well describe a first step toward the creation of a high amplitude non-contact acoustic source. This source is based on the principle of focusing energy on the surface of a sample using time reversal in a hollow cavity. By using a laser vibrometer for the necessary calibration of the system, we are able to use a full non-contact process. This source is proven to be much more energetic than current off the shelf non-contact transducers. It is a broad band source with an adjustable standoff distance. This has the potential to open new perspectives for acoustic processes that require high amplitude, including nonlinear techniques for non-destructive evaluation.

9:45

3aSP6. Localization of a nonlinear source in the bulk of a solid. Pierre-Yves Le Bas, TJ Ulrich, Paul Johnson, and Brian Anderson (Geophys. Group EES-17, Los Alamos Natl. Lab., Los Alamos, NM 87545, pylb@lanl.gov)

Time reversal (TR) has the potential to become a powerful tool in non-destructive evaluation. Coupled with nonlinear properties of cracks in a group of techniques known as Time reverse nonlinear elastic wave spectroscopy, it provides the means to detect and image mechanical damage in complex solids. In this prototype experimental study, we show that we can excite a buried nonlinear feature applying TR. The feature scatters energy that is detected on the sample perimeter. The time signals are filtered about the nonlinear-generated components and are broadcast back, focusing on their source, the buried nonlinear feature. The current challenge is introducing sufficient energy in order to excite the buried feature and produce nonlinear scattering that can be detected on the edges of the sample. This was achieved using a 30 channel system. Two features were localized: a part of the interface between two blocks submitted to a high amplitude signal, and a defect on the same interface (a cured drop of glue creating hard contact between the two solids). Results are visualized using the energy flux quantity.

10:05—10:20 Break

10:20

3aSP7. Ultrasonic defect localization in pipes using time reversal. Nicholas A. O'Donoughue, Joel Harley, José M. F. Moura, and Jun Shi (Dept. of Elec. & Comput. Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, nodonoug@andrew.cmu.edu)

Time reversal mirrors have been shown to effectively focus acoustic waves for structural health monitoring in thin plates. This approach achieves a high degree of accuracy, despite the complex propagation environment inherent to thin plates. We consider ultrasonic defect localization via time reversal in thin-walled pipes. The different boundary conditions in pipes give rise to new wave modes and dispersive effects. We use an array of six transducers arranged around the circumference of a 10-cm-diameter steel pipe to excite guided ultrasonic wave modes, and apply iterative time reversal to focus these waves on a defect. We apply maximum likelihood to detect the presence of a defect, and then correlate the response from the defect with a library of known signatures to locate the defect. Results are provided from a simulation based on experimental data collected from a stainless steel laboratory pipe specimen 11.5 cm in diameter and 1.5 m in length. We simulated our defect with a grease-coupled commercial transducer and measured waves from two arrays of six transducers affixed to the pipe. Our initial excitation is a wideband sinc with 400 kHz bandwidth.

10:40

3aSP8. Optimizing time reversal mirrors in a half-space environment with application to localization of jet noise. Blaine M. Harker and Brian E. Anderson (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602)

Time reversal (TR) is a technique of mapping acoustical sources using a time reversal mirror (TRM) and is especially useful in reverberant environments. Studies of using TRMs in chaotic, reverberant environments have shown that the mirror elements should be spaced with no smaller than a half-wavelength spacing [IEEE Trans. Ultrason. Ferroelectr. Freq. Control **39**(5), 555–566 (1992)]. In a half-space environment, TR cannot exploit multiple reflections to enhance source localization. Thus the question arose as to whether half-wavelength spacing of TRM elements was optimal or not when using TR in a half space. Using simple point source propagation theoretical computer modeling and experimental measurements, it is shown that the TRM elements need not be spaced as close as a half wavelength to optimize focusing. The purpose of this study is to optimize TR localization of an acoustical source using as few TRM elements as possible, thereby increasing the efficiency of the technique. [This work is supported by the Acoustical Society of America's 2010 Robert Young Award.]

Contributed Papers

11:00

3aSP9. Compressing the computational burden of matched-field processing for sound source localization. Karim G. Sabra (School of Mech. Eng., Georgia Tech, Atlanta, GA 30032-0405, karim.sabra@me.gatech.edu), William Mantzel, and Justin Romberg (Georgia Tech, Atlanta, GA, 30032)

Sound source localization in shallow water environment is done using matched field processing (MFP), also referred to as time-reversal imaging or backprojection method. Standard MFP is usually implemented by matching (using a cross-correlation operation) the received acoustic data from the sought source with modeled data (or replicas) for point source located at multiple test locations over the *a priori* search area. Consequently, a direct implementation of MFP (i.e., brute force search) over a large search area can be highly computationally demanding especially when attempting to locate repeatedly several stationary or moving unknown sources in a complex environment. We formulated instead a compressive MFP approach allowing for significant computation time savings with a computational cost that scales with the logarithm of the number of independent-or uncorrelated- test locations. This approach leverages the key concepts behind group testing and compressed sensing by computing instead the expected value of multiple ambiguity surface realizations obtained by backprojection of orthogonal random signals from the receiver locations. Thus compressive MFP allows to estimate the actual ambiguity function using few replica computations by evaluating hypothetical random groups of test locations instead of individual test locations (as done in conventional MFP).

11:15

3aSP10. Using the impulse response from blind deconvolution for elementary source localization. Shima H. Abadi (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109), Daniel Rouseff (Univ. of Washington, Seattle WA 98105), and David R. Dowling (Univ. of Michigan, Ann Arbor, MI 48109)

Artificial time reversal (ATR) is a technique for blind deconvolution in an unknown multipath environment that relies on generic features of underwater sound propagation. Ray-based ATR uses a beam-former-determined reference-ray arrival direction to construct a frequency-dependent phase correction at the receiving array that allows the source-to-array impulse response of the sound channel and the original source waveform to be separately estimated. Although absolute timing and amplitude information is not recovered by ATR, the relative arrival timing of the various signal propagation paths can be determined from the temporal spacing of peaks in the ATR-determined impulse response. With this relative timing information, elementary source localization may be possible when some environmental information is available at the array, and ray-path arrivals can be separated by beamforming at the array. This presentation describes results from underwater experiments involving source-array ranges of 100–500 m in 60-m-deep water and 50-ms chirp signals with a bandwidth of 1.5–4.0 kHz. The ATR-based localization results are found to be comparable with those from coherent and incoherent Bartlett matched field processing. [Work supported by ONR.]

11:30

3aSP11. Two-dimensional speed and depth filtering applied to shallow water matched-field detection. Ryan R. Pitre, Donald R. DelBalzo, and James H. Leclere (QinetiQ North America, 40201 Hwy., 190 East, Slidell, LA 70461)

Passive acoustic sonars have difficulty in detecting quiet sources in noisy shallow-water environments. The standard approach to improve detection is to use a large aperture linear array of horizontally distributed hydrophones and azimuthal processing to increase the signal-to-noise ratio (SNR) through beamforming. A newer approach is to exploit the acoustic interference of normal modes in shallow water and apply matched-field (model-to-data) correlation techniques to estimate source locations in depth and range using signals received on a linear vertical array. These localization techniques are not designed for detection of quiet targets and some applications preclude the use of vertical arrays. This work examines the utility of matched-field processing on linear and planar horizontal arrays on the ocean bottom in shallow water. A speed and depth filter to eliminate high-speed surface targets is applied. A track-before-detect strategy to convert matched-field localizations into source detections in various physical and acoustic environments is utilized. We search for submerged targets at constant speed, constant depth, and linearly changing depth. We show the advantages of planar compared to linear arrays to discriminate submerged from surface sources as a function of SNR. [Work sponsored by QNA.]

11:45

3aSP12. Array design for matched field processing: Role of extending aperture to two dimensions. Paul Hursky, Ahmad T. Abawi, and Michael B. Porter (HLS Res. Inc., 3366 North Torrey Pines Court, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

Most work on MFP has been done with vertical line arrays, and indeed, using the near-orthogonality of the normal modes over the water column provides better resolution of range and depth than a horizontal array of comparable size. However, when we put an array into an environment where there are multiple sources, some much louder than others, the horizontal aperture's ability to form nulls in azimuth starts to trump the vertical aperture. We will present simulation results comparing various combinations of horizontal and vertical apertures and evaluate their ability to both resolve source range and depth and to cope with a loud interfering source in an adaptive MFP context, where the loud source hides other sources if conventional Bartlett (i.e., nonadaptive) MFP is used.

WEDNESDAY MORNING, 25 MAY 2011 QUEEN ANNE, 9:00 TO 10:30 A.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C. A. Champlin, Chair ASC S3
University of Texas, Department of Communication Sciences & Disorders, CMA 2-200, Austin, TX 78712

G. J. Frye, Vice Chair ASC S3
Frye Electronics, Inc., P.O. Box 23391, Tigard, OR 97281

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note — those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 24 May 2011.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

WEDNESDAY MORNING, 25 MAY 2011 QUEEN ANNE, 10:45 A.M. TO 12:00 NOON

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. K. Delaney, Chair ASC S3/SC 1
USA CERL, 2902 Newmark Drive, Champaign, IL 61822

M. C. Hastings, Vice Chair ASC S3/SC 1
Georgia Institute of Technology, G.W. Woodruff School of Mechanical Engineering,
126 Love Building, 771 Ferst Drive, Atlanta, GA 30332-0405

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

Session 3pAA

Architectural Acoustics: Room Acoustics II

Michael R. Yantis, Chair
Sparling, 720 Olive Way, Seattle, WA 98101-1833

Chair's Introduction—1:00

Contributed Papers

1:05

3pAA1. Experimental and numerical investigations on decay parameter distributions in coupled-volume systems. Angela Ostrowski (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, aostrowski303@gmail.com), Jose Escolano (Univ. of Jaén, Linares 23700, Spain), Ning Xiang, and Jonas Braasch (Rensselaer Polytechnic Inst., Troy, NY 12180)

The current interest of couple-volume systems in performing art spaces has motivated research to better understand and predict multiple-sloped sound energy decays. This work investigates sound energy decay characteristics of coupled spaces using scale model measurements and diffusion-equation modeling results. The experimental and numerical study is aimed to reveal possible ranges of decay parameters when investigating source and receiver positions in the primary room and aperture sizes. The receiver positions over large areas in the primary room are investigated to identify the presence of a spatial dependence and variations of decay parameters on the source position as it changes with respect to the aperture. Both scale model measurements and diffusion-equation model results are analyzed using Bayesian probability inference in order to estimate decay orders (number of multiple slopes) and to quantify multiple-slope decay parameters including individual decay levels, level differences, and decay times of individual slopes. This paper will also discuss dependence on the upper limit of backward integration on signal-to-noise ratios achieved from experimental measurements.

1:20

3pAA2. Using a spherical microphone array to analyze stage acoustics. Anne E. Guthrie (Arup, 155 Ave. of the Americas, New York, NY 10013, guthra2@rpi.edu), Terence J. Caulkins (Arup, New York, NY 10013), Samuel Clapp, and Jonas Braasch (Rensselaer Polytechnic Inst., Troy, NY 12180)

A spherical microphone array, capable of beamforming, has been constructed to measure sound fields using second-order ambisonics. The design of the microphone is based on the work by Duraiswami *et al.* using equations for scattering off a rigid sphere. The 16-channel array was used to measure nine different concert-hall stages around the state of New York, with multiple positions recorded to analyze their acoustical behavior. Additional measurements with an omnidirectional microphone were obtained to determine A.C. Gade's stage support parameter for comparison purposes. Based on the measurements, alternative methods will be discussed to obtain better descriptors to quantify stage acoustics based on the additional spatial information the spherical microphone provides. Additionally, the work by Dammerud in developing geometric parameters based on comparisons between omnidirectional measurements and architectural dimensions will be utilized. A comparison between parameters by Dammerud and information extracted from a beamforming analysis of the ambisonic stage measurements will be shown. In particular, the comparison will examine the spatial response over time to understand how this is affected by the dimensions of the stage.

1:35

3pAA3. Explorations of room acoustics using a second-order ambisonic microphone. Sam Clapp, Anne Guthrie, Jonas Braasch, and Ning Xiang (Grad. Prog. in Arch. Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

Most room acoustic parameters are calculated with data from omnidirectional or figure-of-eight microphones. Using an ambisonic microphone to record room impulse responses can open up several new areas of inquiry. It can yield much more information about the spatial characteristics of the sound field at the points of interest, including the diffuseness of the sound field and the directions of individual reflections. In this research, a 16-channel, second-order ambisonic microphone is designed, built, and tested with both simulations and simple, controlled sound events. Room impulse responses are then measured for a wide variety of different concert halls located throughout New York state. The results are analyzed using beamforming techniques to determine spatial information about the sound field. This method has allowed for determining the directions of individual early reflections and to characterize the spatial evolution of the sound field over the course of the impulse response. This new dimension of data can also help in understanding the differences between halls that may yield similar values in traditional metrics (such as reverberation time and clarity), but which are judged very differently by listeners.

1:50

3pAA4. Just-noticeable-difference of clarity index (C80) using real-time switching between test signals. Adam P. Wells, Caitlin I. Ormsbee, Robert D. Celmer, and Michelle C. Vigeant (Dept. of Mech. Eng., Acoust. Prog. and Lab., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, vigeant@hartford.edu)

The purpose of this study was to investigate the just noticeable different (JND), or the smallest detectable difference, for the clarity index for music (C80). This project is the third in a series of investigations on this topic. The first study showed that C80 JND is not close to 1 dB, as work from other researchers had previously found. The second study examined the effect of two different test methods. Test-method1 required the subjects to listen to all of signal A and all of signal B, while for Test-method2 subjects could switch in real-time between the two signals. The results of the second study, which were implemented in this current study, suggested adding an extensive training period with Test-method1 before using test-method2 for the actual test. Two C80 base-cases were used, with base-case1 having C80 at a 1 kHz set to -3 dB with a reverberation time (T30) of 1.9 s and base-case2 with a C80 of $+1$ dB with T30 of 1.5 s. Highly trained musicians were used as test subjects. The C80 JNDs for each base case, along with a comparison to the results from the first two studies, will be discussed. [Work is supported by the Paul S. Veneklasen Research Foundation.]

2:05

3pAA5. The measurement and evaluation of bespoke 3-D absorptive panels: A comparative analysis. John Zeman, Stephen Dance, and Georgia Zepidou (Dept. of Urban Eng., London South Bank Univ., Borough Rd., London SE1 0AA, United Kingdom, dances@lsbu.a.cuk)

A recent movement of Scandinavian design houses has brought about a new artistic sensitivity to porous sound absorbers in the world of architectural acoustics. These absorbers have vibrant colors and textures never before seen on a mass scale. However, as the design of these absorbers becomes more intricate, the problem arises on how to test such bespoke products in accordance with international standards. A particular problem is the arbitrary size of the products. Monetary and labor constraints keep these absorbers from being produced to a size that will satisfy international requirements. Therefore, a testing methodology and correction factor is needed in order to predict the acoustic performance of the designer products. Although this test methodology is not meant to replace current standards for total surface area criteria, this approach would work as a corollary, allowing materials that could not be easily reproduced to be measured in conjunction with standards despite their size, saving designers and manufacturers both time and money.

2:20—2:30 Break

2:30

3pAA6. Rating of the loudest college basketball arenas for Entertainment Sports Network (ESPN) magazine. Micah R. Shepherd, Stephen A. Hambric, Neal U. Evans, Dan J. Domme, Andrew W. Christian, Kieran Poulain, Michael D. Gardner, Andrew J. Orr, and Bryan P. Cranage (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802)

A recent ESPN magazine article [“These go to 11,” ESPN the Magazine 15 Nov (2010)] ranked the top collegiate basketball arenas according to “noise potential.” The top five arenas were listed as Kansas University, Duke University, New Mexico University, Kentucky University, and Florida University. The rankings were established by a team of Penn State Acoustics students using the theory for sound buildup in large rooms, since actual measurements were infeasible. Both diffuse field and direct field contributions of the sound pressure were estimated at center court for octave band frequencies from 125 Hz to 4 kHz. Seating geometries, materials, and other relevant information were collected for each arena and used with estimated absorption coefficients to determine the room constant and critical distance. The diffuse field contributions were then combined with approximate sources terms based on the seating capacity of the arena, the proximity of the fans to the court, and whether they were students. The sound pressures were then combined into a total A-weighted sound pressure level and used to determine the ranking. The direct and reverberant contributions of each arena were then compared to establish the positive and negative aspects of each arena in terms of noise potential. This comparison reveals how Kansas and Duke reached the top of the ranking, despite having drastically different arena geometry and capacity.

2:45

3pAA7. Acoustical behavior of colonial churches in Cusco, Peru. Carlos R. Jimenez-Dianderas (Dept. of Architecture, Pontificia Universidad Catolica del Peru, Av. Universitaria 1801, Lima 32, Peru, cjimene@pucp.edu.pe)

The paper searches for characterizing the acoustical behavior of more than ten colonial churches in Cusco, a southern city of Peru. Cusco was an important city through the Spaniard Colonial Era in Peru; several religious orders established in the area and fix the European styles and building process to the geography, geology and building materials. Through five typical acoustical objective parameters the acoustical environment in each church is defined. A comparison between several positions in each church analyzed and among churches will be presented.

3:00

3pAA8. Impact sound insulation predictions for light weight floors: Potential modal response of the floor and its effect on impedance. D.J. Griffin (Marshall Day Acoust., 84 Symonds St., Auckland 1010, New Zealand, dgriffin@marshallday.com)

While light weight flooring systems are commonly used in new apartment developments in many countries, they are generally regarded as providing only moderate impact sound insulation. Poor acoustic design of a light weight flooring system can result in non-compliance with applicable building code standards and poor perceived performance, particularly in comparison to massive floor systems which include a concrete slab. It is therefore useful to develop a reliable prediction routine, to engineering accuracy, for light weight floors. This paper briefly outlines existing theory for prediction of impact sound insulation for light weight floors. A development of the basic model is proposed, involving consideration of the modal response of the floor membrane and its potential effect on the impedance of the floor in the mid-frequency region.

WEDNESDAY AFTERNOON, 25 MAY 2011 ISSAQUAH, 1:30 TO 2:45 P.M.

Session 3pAB

Animal Bioacoustics: General Topics in Beaked Whale Detection

Holger Klinck, Chair

Oregon State Univ., CIMRS, 2030 S.E. Marine Science Dr., Newport, OR 97365

Contributed Papers

1:30

3pAB1. Are Blainville’s beaked whales echo-locating without a clock? Mark Fischer (Aguasonic Acoust., P.O. Box 1073, Rio Vista, CA 94571-3073, info@aguasonic.com)

Recent work using wavelets in the service of detection and classification of Ziphiid clicks indicates that a novel method of echo-location exists in the clicks of Blainville’s beaked whales /*Mesoplodon densirostris*/. Results will be presented that support the hypothesis that, instead of timing, they rely on the creation of a unique waveform that allows them to perceive round-trip distance based on changes in the wavelet power spectrum.

1:45

3pAB2. Real-time detection and tracking of beaked whales using a towed hydrophone array. Tina M. Yack, Jay Barlow (NOAA Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Ct., La Jolla, CA 92037, tina.yack@noaa.gov), and John Calambokidis (Cascadia Res. Collective, 218 1/2 W 4th Ave., Olympia, WA 98501)

Beaked whales spend the majority of their time at depth and are inconspicuous when they surface. Therefore, they are difficult to detect using only standard visual survey methods. However, recent advancements in acoustic detection have made passive acoustic monitoring from a towed array of hydrophones a viable alternative to visual survey methods for beaked whales. Beaked whales can be discriminated from other cetaceans by the unique characteristics of their echolocation clicks (duration >175 μs, center frequencies between 30 and 40 kHz, inter-click intervals between 0.2 and 0.4 s and frequency upsweeps). These unique characteristics make it possible to use passive acoustic monitoring (PAM) to detect and track beaked whales in real-time. In 2009 and 2010, we used PAM to identify areas of beaked whale habitat use in the Southern California Bight (SCB). This area is characterized by deep basins separated by islands and shallow ridges. We developed effective methods for detecting and tracking beaked whales in real-time and identified previously unknown areas of beaked whale habitat use in the SCB. This work will allow for beaked whale habitat relationships to be explored on a relatively small scale and will contribute to efforts for effective management and conservation of beaked whales in the SCB.

2:00

3pAB3. Complementary techniques for robust acoustic identification of individual sperm and beaked whales. George E. Ioup, Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, geioup@uno.edu), Christopher O. Tiemann (Univ. of Texas at Austin, Austin, TX), and Natalia A. Sidorovskaia (Univ. of Louisiana, Lafayette, LA)

Properties of individual clicks and characteristics of click trains provide clues for the passive acoustic identification of individual sperm whales via several complementary techniques. For example, recordings of individual clicks can be clustered according to their shape or spectrum, or entire trains of clicks can be exploited by monitoring the slow evolution of click shapes over time. Cadence analysis studies differences in click-train rhythms of simultaneously clicking whales. Localization uses bearing, and if possible range, to differentiate individuals. Manual click-train identification employs both individual click properties and sequence analysis to identify individual whales. When combined, these methods can provide robust identification of individual sperm whales. The application of these techniques to beaked whales is difficult because, compared to sperm whales, beaked whales have lower source level, a narrower beam, and a faster rate of turning. Consequently, click trains on a single sensor have frequent drop-outs. The above techniques require more sophistication and benefit from closely spaced sensors to fill in drop-outs. Previous Littoral Acoustic Demonstration Center (LADC) multi-mooring Environmental Acoustic Recording System (EARS) data are analyzed for beaked whales and an experiment is proposed with greater mooring density to improve individual beaked whale identification. [Research supported by SPAWAR and ONR.]

2:15

3pAB4. Noise reduction for better detection of beaked whale clicks. Yang Lu, Holger Klinck, and Dave Mellinger (CIMRS, 2030 SE Marine Sci. Dr., Newport, OR 97365, luya@onid.orst.edu)

A least mean square (LMS)-based filter with a forgetting factor has been designed to improve the signal-to-noise ratio (SNR) of clicks recorded from Blainville's beaked whales (Mesoplodon densirostris). A suboptimal stepsize of the LMS filter is obtained through searching. The trained filter is proven to be environmentally robust; i.e., it can be used in beaked whale recordings from other areas, which is an advantage for applications such as autonomous gliders for which accurate click detection in untested locations is beneficial. It can also be used to improve the SNR of other marine mammal acoustic signals. A detector using filter output is designed to detect beaked whale calls. Through simulations, the proposed detector is shown to be capable of detecting most of the desired clicks, but is not able to differentiate other coexisting species such as Risso's dolphins and pilot whales. By combining the proposed detector with the energy ratio mapping algorithm (ERMA) [Klinck and Mellinger (in preparation)], which measures energy differences between different species, higher detection accuracy for beaked whale clicks can be achieved.

2:30

3pAB5. Passive-acoustic monitoring of odontocetes using a Seaglider: First results of a field test in Hawaiian waters. Holger Klinck (CIMRS Bioacoustics Lab, Hatfield Marine Sci. Ctr., Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, holger.klinck@oregonstate.edu), David K. Mellinger (Oregon State Univ., Newport, OR 97365), Marie A. Roch (San Diego State Univ., San Diego, CA 92182), Karolin Klinck (Oregon State Univ., Newport, OR 97365), Neil M. Bogue, Jim C. Luby, William A. Jump, John M. Pyle, Geoff B. Shilling, Trina Litchendorf, and Angela S. Wood (Univ. of Washington, Seattle, WA 98105)

In fall 2009 the University of Washington, Applied Physics Laboratory conducted in collaboration with the Oregon State University, a comprehensive field test of a passive-acoustic Seaglider along the western shelf-break of the island of Hawaii. During the 3 week mission, a total of approximately 170 h of broadband acoustic data [194 kHz sampling rate] were collected. The recordings were manually analyzed by an experienced analyst for beaked whale (Ziphiidae), dolphin (Delphinidae), and sperm whale (Physeter macrocephalus) echolocation clicks as well as echo sounder pings emitted by boats in the area. Here we present and discuss first results of these data analysis, which revealed that more than 50% of the recorded files (each of 1-minute duration) contain bioacoustic signals. Furthermore the recorded data and the results of the manual analysis are used to validate and optimize an automated classifier for odontocete echolocation clicks, which was developed in a collaborative effort with San Diego State University. The algorithm is intended to be implemented on the Seaglider to enable species identification by classifying detected echolocation clicks in (near) real-time during sea trials. [This work is funded by the Office of Naval Research and the U.S. Navy's Environmental Readiness Division.]

Session 3pBA

Biomedical Acoustics: Best Student Paper Award (Poster Session)

Tyrone M. Porter
Boston Univ., Biomedical Engineering, Boston, MA 02115

Kevin Haworth
Univ. of Cincinnati, Dept. of Internal Medicine, Cincinnati, OH 45267

Todd Hay
Univ. of Texas, Applied Research Lab., Austin, TX 78713

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with $500 for first prize, $300 for second prize, and $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with their abstract number and title listed. Full abstracts can be found in the oral sessions associated with the abstract numbers.

All entries will be on display and all authors will be at their posters from 1:30 p.m. to 3:00 p.m.

1pBA11. *In-vitro* validation of three-dimensional cavitation-based pressure mapping for quality assessment of clinical high intensity focussed ultrasound devices. Student Author: Stuart Faragher

2aBA12. Quantitative ultrasound assessment of thermal therapy in liver. Student Author: Jeremy Kemmerer

2aBA13. The application of phase-shift nanoemulsion in high intensity focused ultrasound-mediated heating and its potential in monitoring lesion formation. Student Author: Peng Zhang

2pBA10. Miniature acousic fountain mechanism for tissue emulsification during millisecond boiling in high intensity focused ultrasound fields. Student Author: Julianna Simon

2pBA12. Ultrasound-induced fragmentation of connective tissue guided by passive acoustic mapping. Student Author: Michael Molinari

2pBA13. Modeling single-bubble dynamics in histotripsy. Student Author: Changyun Hua

3aBA1. Theoretical considerations for ultrasound contrast agent amplitude modualtion techniques at high frequencies. Student Author: Amin Jafari Sojahrood

3aBA12. Subharmonic response from ultrasound contrast microbubbles for noninvasive blood pressure estimation. Student Author: Amit Katiyar

4aBA1. Vulnerable atherosclerotic plaque in human artery sections and apolipoprotein E-deficient mouse aortas. Student Author: Pavlos Anastasiadis

4aBA2. Smoking effects on cyclic variation of harmonic echogenicity in carotid artery. Student Author: Ying Li

4aBA7. Effect of low intensity pulsed ultrasound on mesenchymal stem cells. Student Author: Jia-Ling Ruan

4aBA9. Bubble dependence of the mechanism FUS-induced blood–brain barrier opening in mice *in vivo*. Student Author: Yao-Sheng Tung

4aBA10. Ultrasound-induced permeability variations in cell gap junctions for drug delivery. Student Author: Pavlos Anastasiadis

4aBA13. Miniature kilohertz-frequency ultrasound implant for brain drug delivery. Student Author: John Foo

4pBA8. Complex wavelet analysis of high frequency ultrasound backscatter from low echogenic biospecimen. Student Author: Sushma Srinivas

4pBA13. Numerical modeling of photoacoustic imaging of brain tumors. Student Author: Kamyar Firouzi

Session 3pED

Education in Acoustics and Committee on Women in Acoustics: Listen Up and Get Involved

Marcia J. Isakson, Cochair
Univ. of Texas at Austin, Applied Research Lab., 10000 Burnet Rd., Austin, TX 78713

Tracianne B. Neilsen, Cochair
Brigham Young Univ., Dept. of Physics and Astronomy, Provo, UT 84602

Chair's Introduction—5:30

Invited Papers

3pED1. Integrating interactive simulations into demo sessions to help students visualize the invisible. Wendy K. Adams (Dept. of Phys., Univ. of Northern Colorado, CB 127, Greeley, CO 80639, wendy.adams@colorado.edu)

The hands-on demos included with the ASA demo-kit or "traveling road show" are excellent demonstrations of physical phenomena such as standing wave & resonance (spring, string vibrater, pitch fork, air columns, and wine glass), the force of sound (ultrasonic levitation), and spectrum analysis just to name a few. Most of these demos work with either waves that move so quickly the eye cannot follow (spring appears to be in two places at once) or sound waves in air, which are invisible leaving it up to the students to visualize on their own. Integrating in some interactive simulations such as PhET http://phet.colorado.edu could provide students with the ability to slow time while observing standing waves "Wave on a String" or see the sound waves travel from a speaker to the ear "Sound" and "Wave Interference" or see how individual harmonics (sines and cosines) can create a more complicated sound wave, "Fourier: Making Waves." These simulations will be demonstrated along with ideas on how to fit them into a future demo show.

3pED2. Marine mammal vocalizations: Spectrum analysis. Melanie E. Austin (JASCO Appl. Sci., 4464 Markham St., Victoria, BC V8Z 7X8, Canada, melanie.austin@jasco.com)

Marine mammals communicate underwater using sounds at different frequencies. Some whales speak using low frequency moans, and others communicate using high frequency whistles and clicks. Some sing notes that sweep over different frequencies. Come hear some of the amazing sounds that whales, seals, and walrus make! Look at spectrum plots to see which frequencies make up the sounds. See if you can match pictures of spectrogram plots with the sounds you hear. Also explore the sounds made by common man-made noise sources in the ocean such as ships, sonars, and geophysical survey equipment. In this demonstration, you will learn how to distinguish low frequency and high frequency sounds, both by ear and by eye, through a computer based matching game. You will click on an image of a marine mammal, or a man-made noise source, to hear an audio clip of the sound that each makes. After hearing the sound, try to guess which of the presented spectrum plots matches the sound you heard.

3pED3. The effects of underwater noise on marine mammals. Christine Erbe (JASCO Appl. Sci., Brisbane Technol. Park, 1 Clunies Ross Ct., Eight Mile Plains, Queensland 4113, Australia)

The marine soundscape is made up of natural ambient sounds, biological sounds, and anthropogenic sounds. Natural ambient sound sources include wind, waves, rain, and ice. Biological sounds include calls emitted by marine mammals, fish, and crustaceans. Water conducts light very poorly but sound very well, which is why marine animals rely heavily on acoustics for communication and navigation. Did you know that fish talk underwater? Anthropogenic, i.e., man-made, sources of sound include ships, harbor construction, petroleum exploration, naval sonar, fish finders, etc. Acoustic ecology encompasses the relationships—mediated through sound—between organisms and their environment. An overview of marine acoustic ecology will be presented with focus on the effects of man-made noise on marine mammals. You will be able to immerse yourself in the underwater soundscape and listen to both man-made and animal sounds. The interference of noise with animal signals will be explored through masked hearing experiments. Come and test if you can hear whale calls in ship noise better than a trained beluga whale from Vancouver Aquarium.

3pED4. Killer whale scouting: Listen live for troops J, K, and L. Scott R. Veirs (Beam Reach Marine Sci. and Sustainability School, 7044 17th Ave. NE, Seattle, WA 98115, scott@beamreach.org), Val R. Veirs (Colorado College, Friday Harbor, WA 98250), and Jason D. Wood (SMRU, Friday Harbor, WA 98250)

The southern resident killer whales are highly vocal, sharing almost 40 distinct calls. The endangered population of about 87 orcas live in matriarchal groups (like troops) known as J, K, and L pods. Each pod has a favorite call that is used about half the time pod members vocalize. If you hear a favorite call you can guess (or infer) which pod is present. A new way to listen for orcas is with hydrophones (underwater microphones) that are connected to the internet. The Northeast Pacific Hydrophone Network—http://orcasound.net—includes five hydrophones located near Seattle, WA within the summertime range of the orcas. You can listen live through iTunes to discover which pods are calling or to monitor the environment for noise pollution that could be a problem for the whales. This hands-on activity teaches you how to listen, identify the three pods based on their favorite calls, and share your observations with other listeners. High school students can use free software to record orca calls, whistles, and clicks and analyze them, guided by a suite of citizen science projects.

3pED5. Helmholtz resonator: An empty bottle. Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Helmholtz resonance is the phenomenon of air resonance in a cavity, such as when one blows across the top of an empty bottle. In this demonstration, the sound radiating from the bottle will be described in terms of a lumped acoustic system. Such an explanation applies the long wavelength limit, which allows acoustic devices to be modeled as a harmonic oscillator with two degrees of freedom. For the case of an empty bottle, the air inside the neck of the bottle behaves as the mass element. The air contained in the remainder of the bottle makes up an elastic cushion, which acts as the spring element. Together they form an oscillating system with a specific resonance frequency that can be easily excited, as is well known, by blowing across the opening of the bottle. The relative effects of the mass and stiffness of the system on the pitch of the sound radiated from the bottle will be demonstrated by changing the volume of air in the neck and cavity of the bottle.

3pED6. Hands-on demonstrations for Project Listen Up: Education outreach part II. Colleen E. Carberry, Anna E. Carpenter, and Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402)

Midshipmen will be getting involved in an ASA education outreach effort by presenting or video taping a number of acoustical demonstrations geared to promote a hands-on learning experience for middle- and high-school age girl Scouts. This is an extension of the demonstration effort and outreach presented at the ASA Meeting held in Baltimore in 2009. The demos are designed to visualize certain wave effects that will be explained by the Midshipmen "live" or by video. The participants will be free to explore and then control the apparatus and make their own scientific discoveries. The hands-on demonstrations will include (1) a ripple tank with an electronic wave height transducer for wave visualizations or making simple oscilloscope measurements, (2) a torsion wave machine for demonstrating reflection and interference effects, and (3) a forced damped harmonic oscillator apparatus that consists of a subwoofer driver connected to the bottom of a vertical spring that is attached at the top to a hanging mass on a string that is suspended over a pulley. Resonance effects can be easily observed watching the pulley oscillate around 1 Hz. The driver frequency is controlled by adjusting a single knob and looking at the digital display.

3pED7. Zombie classification demonstration. Jennifer L. Cooper (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723, jennifer.cooper@jhuapl.edu)

Many tasks in the real world call for identifying or grouping things by the sounds they make. Some examples include identifying species of whale or even individual animals using recordings of their clicking sounds or finding the cause of an engine failure by the sound it makes. The same methods can be used to determine whether a sound is human or animal in nature or even human vs zombie. In this demonstration, students' voices are recorded and subsequently classified as human or zombie using a neural network. Using a graph called a spectrogram, we can look at the spectral (time and pitch) features of each voice and discuss how those features were used to identify it as human or zombie.

3pED8. Standing waves on piano wire. Sarah Hargus Ferguson (Dept. of Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

This demonstration will use a piano wire stretched between two pieces of aluminum exactly 100 cm apart. Plucking this wire will set it into vibration, generating sound waves that we can hear. Touching the wire at different locations along its length will change the way the wire vibrates and, therefore, change the sound we hear.

3pED9. The vibrating string. Helen Hanson (Dept. Elec. and Comput. Eng., Union College, 807 Union St., Schenectady, NY 12308)

In this experiment, we will observe standing waves on a long elastic band. A sine wave generator will be used to make the band vibrate at different frequencies. The band will vibrate particularly well at certain frequencies, during which distinct loops will appear on the band. Observers will note the frequency that results in a single loop on the band, then the frequency that results in two loops, and so on up to five loops. The frequencies that result in these loops should be integer multiples of the single-loop frequency.

3pED10. Listen up and perceive speech: Demonstrating the flexibility of speech perception. Rachel M. Miller (Dept. of Psych., Univ. of California, Riverside, 900 University Ave, Riverside, CA 92521, rmiller.ucr.grad@gmail.com)

Individuals encounter many different types of speech information on a daily basis. Some of this information comes in forms that are not even auditory in nature (e.g., visual speech). Other types of information contain variations that should make them difficult to understand (e.g., speech-in-noise and accented speech). However, individuals are able to perceive all sorts of speech quite easily. The present demonstrations will show the flexibility of speech perception by providing girl scouts with examples of various types of speech. The demonstrations will be (1) a lipreading choice task that presents silent videos of talkers producing words, (2) silent videos of talkers producing sentences, (3) silent, point-light videos that show how speech can be perceived simply by seeing the movement of talkers' faces, (4) sentences produced in three levels of noise, and (5) words produced by native Chinese and Spanish talkers.

3pED11. How musical instruments make sounds: An exploration of standing waves and resonance frequencies. Tracianne B. Neilsen and Jessica Morgan (Dept. of Phys., Brigham Young Univ., 283 ESC, Provo, UT 84602, tbn@byu.edu)

A basic understanding of the production of sound waves can be obtained by studying standing waves and resonance frequencies. A vibrating string settles into a steady, standing wave pattern when it is driven at one of its resonance frequencies. The resonance frequencies depend on the string's length and the tension applied to the string. Similarly the resonance frequencies of tubes are determined by the tube's length and whether the ends are open or closed. In a series of hands-on demos, we explore the factors that influence the creation of standing waves by exciting the resonances of strings, tubes, rods, a metal plate, slinky, and a wine glass. These simple models provide insight into how musical instruments produce sound.

3pED12. Singing wine glasses and whistling bottles. Trudy L. Philip (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723)

These are two experiments that are easy to do at home. The first is with a wine glass, it does not have to be fancy, but it must be a glass on a stem. Put some water in the glass and hold the glass firmly on the counter by the stem. Wet one of your fingers and run it around the rim of the glass. Eventually, the glass will sing. Change the amount of water in the glass, the pitch will change. A bottle is needed for the second experiment. It does not have to be a specific type of bottle or a certain height, it does not even need to be glass;

however, it may be easiest with a bottle with a narrow neck and smaller opening like a soda bottle. With about half the bottle full of water, blow across the top of the opening of the bottle, trying not to blow directly into the bottle. You may have to adjust the angle at which you are blowing, but eventually the bottle will whistle. Now, change the amount of water in the bottle and the pitch will change. These singing household items use two different acoustic principles!

3pED13. Building a sound and breaking it down. Tetjana Ross (Dept. of Oceanogr., Dalhousie Univ., Halifax NS B3H 4J1, Canada, tetjana@dal.ca)

The sounds we hear everyday are made up of acoustic waves of many different frequencies. When a sound is dominated by waves of a particular frequency, it is referred to as having a certain pitch. However, even single musical notes are often made up of various different frequencies (usually harmonics of the first). This is part of what gives different musical instruments their distinctive sounds. Our ears analyze sounds, with different nerve endings being activated by waves of different frequencies. This breaks sounds down so that we can distinguish different pitches or even chords. This breaking down of sound, showing its loudness as a function of frequency, is called finding its spectrum. In two hands-on experiments, we will use a computer to visualize the building and breaking down of sounds. First, we will use a keyboard as a wave generator to incrementally build up a sound from waves of different frequencies, all the while using a computer to show us the spectrum of the resultant sound. Second, we will record you singing an "ah" or "oo" sound and examine its spectrum to show all the frequencies involved.

3pED14. Longitudinal waves demonstrations. Michelle E. Swearingen (US Army ERDC-CERL, P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil)

In a wave, something always moves back and forth. A longitudinal wave is one that travels along the direction of propagation. In this demonstration, two different devices will be used to illustrate longitudinal propagation. An aluminum bar is used to produce transverse waves. A slinky will be used to visually show longitudinal propagation along the slinky. Finally, the aluminum bar is again used to show differences in longitudinal and transverse wave propagation.

3pED15. Discover sounds in the sea. Kathleen J. Vigness-Raposa (Marine Acoust., Inc., 809 Aquidneck Ave., Middletown, RI 02842, kathleen.vigness@marineacoustics.com), Gail Scowcroft, Christopher Knowlton, Holly Morin (Univ. of Rhode Island, Narragansett, RI 02882), Peter F. Worcester (Univ. of California-San Diego, La Jolla, CA 92093), James H. Miller (Univ. of Rhode Island, Narragansett, RI 02882), and Darlene R. Ketten (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Investigate the various sounds that exist in our oceans in the Name That Sound interactive game. Sounds are presented and contestants are challenged to select the correct source of the sound from four possible options. This hands-on activity is based on the Audio Gallery, with a wide range of sounds produced by animals, people, and natural processes like waves and rain, at the "Discovery of Sound in the Sea" website (http://www.dosits.org). The DOSITS website explains the physical science of underwater sound and how people and animals use sound for many purposes, like communication and exploration. It also has three major resource sections: media, teacher, and student. The media resources include a Facts & Myths quiz, Frequently Asked Questions, and PDF downloads of a tri-fold pamphlet and 12-page educational brochure. The teacher and student resources include structured tutorials, classroom activities, and educational games, including the Name That Sound game. The DOSITS team recently launched a significantly redesigned website. The look and feel has been refreshed without losing functionality and content. The redesign includes an interactive front page and Audio Gallery, and a redesigned Scientist Gallery. Participants at ASA will receive free CD-ROMS. Come explore the science of underwater sound with DOSITS!

3pED16. Pitch perception. Lynne Werner, Angela Garinis, Bonnie Lau, and Louise Yeager (Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105)

Pitch is the characteristic of sound that varies from low to high. For example, the pitch of a man's voice is low, while the pitch of a child's voice is high. Sounds produced by different sources, such as a musical instrument or a person speaking, sound different, but still share the same pitch. These demonstrations will show how different sound waves can have the same pitch.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Barbara G. Shinn-Cunningham, Chair
Boston Univ., Biomedical Engineering, 677 Beacon St., Boston, MA 02215

Invited Papers

1:20

3pID1. Bilateral cochlear implants in young children. Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin, Madison, WI 53705, litovsky@waisman.wisc.edu)

Cochlear implants (CIs) are provided to a growing number of people who are deaf. Single CIs offer opportunity for language acquisition, but there exist remaining challenges. In normal-hearing individuals binaural hearing provides cues that are important for sound localization and speech understanding in noisy environments. Provision of bilateral CIs to many children occurs with the expectation that they will develop spatial hearing abilities. Outcomes in these children are mixed; while some are able to localize sound significantly better with bilateral CIs than with a single CI, other children demonstrate poor spatial hearing skills. Factors that seem to contribute to performance include amount of auditory deprivation prior to implantation and auditory experience following bilateral activation. Hardware/software limitations in implanted devices result in binaural cues being distorted or discarded. Studies using research processors show that these cues can be transmitted with fidelity to the patients, but that perceptual sensitivity to these cues may depend on the period of auditory deprivation that listeners have experienced. This talk will focus on progress and outcomes in children who are deaf and use bilateral CIs, with suggestions for future development that will depend on improved audio signal processing and signal delivery to the implantable devices.

1:45

3pID2. Hot topics of signal processing in acoustics. David H. Chambers (Lawrence Livermore Natl. Lab., P.O. Box 808, L-154, Livermore, CA 94551)

Signal processing is used to some extent in all areas of acoustics such as extracting relevant information from acoustic measurements made either in the laboratory or in the field, processing signals, and/or synthesizing data to cope with demanding tasks raised in acoustics. Techniques range from simple classical approaches based on Fourier transforms and Gaussian noise to sophisticated model-based techniques that incorporate physical/parametric models of the acoustical system. In this talk we highlight new approaches to signal processing that could be applied to a broad variety of acoustical problems. Examples of each approach will be shown to illustrate the class of problems addressed and the performance. [This work performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.]

2:10

3pID3. Hot topics in underwater acoustics. Lisa Zurk (Dept. ECE, Portland State Univ., 1900 SW 4th FAB 160-17, Portland, OR 97201, zurkl@pdx.edu)

Research in underwater acoustics continues to evolve with the development of more capable acoustic systems, the introduction of new applications for underwater sensing, and with the emergence of integrated physics-based processing approaches, which provide robustness in variable ocean environments. This talk provides an overview and examples of some of the active research areas in underwater acoustics, with particular emphasis on three topics. The first topic is the characterization and exploitation of underwater noise. This topic has research activities in both environmental monitoring (and potentially mitigation) of noise produced from anthropogenic activities, as well as the creative utilization of ambient noise structure for passive underwater sensing. The second topic area is the application of the waveguide invariant in active and passive sonar systems for enhanced detection, localization, classification. The invariance structure is being explored for deep water environments, with application to different sensor geometries and potentially new beamforming constructs. The final research area is underwater acoustic communications and networking. These systems are evolving to higher data rates (both single and multi-carrier) with adaptive and self-organizing algorithms to fully exploit the capabilities of distributed undersea networks.

2:35

3pID4. Building the oceanic concert hall: Multidisciplinary applications of acoustics for ocean observing systems. Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105, dushaw@apl.washington.edu)

Over the past decade, the oceanographic community has been constructing integrated multidisciplinary observational systems to serve the immediate and long-term needs of the society (www.ioos.gov). The overarching goal is to provide practical products and information to society as a return on its long-term investment in oceanographic research. Demand is increasing for information about the state of our oceans to address a myriad of issues ranging from climate variability to fisheries management to public education. Customers for ocean observing systems range from government agencies to commercial shipping companies to the science projects of high school students. Two developing “observing systems” in the Pacific Northwest are the Northwest Association of Networked Ocean

Observing Systems (www.nanoos.org) and the Neptune Undersea Cable Network (www.interactiveoceans.washington.edu). The oceans are largely transparent to sound; hence, oceanographic, biological, and signal-processing acoustic techniques are primary tools for ocean observation and engineering. The opportunities and value of acoustical observations and techniques within these systems are boundless, yet incorporation of these techniques has been opportunistic and *ad hoc*. Coordination of the acoustical applications is essential. Organizations advocating acoustical observations face enormous challenges of planning, implementation, and data management to bring acoustical tools to fruition for ocean observing systems.

WEDNESDAY AFTERNOON, 25 MAY 2011 ASPEN, 1:00 TO 2:45 P.M.

Session 3pMU

Musical Acoustics: Acoustics of Keyboard Instruments

Antoine Chaigne, Chair
ENSTA, Chemin de la Huniere, 91761, Palaiseau, France

Invited Papers

1:00

3pMU1. Influence of ribs on the radiation of a piano soundboard. Antoine Chaigne, Romain Vuillez, and Roberto Viggiano (Unit of Mech. Eng. (UME), ENSTA ParisTech, Chemin de la Huniere, 91761 Palaiseau, France, antoine.chaigne@ensta.fr)

The presence of ribs attached to the soundboard of a piano influences the transfer of energy from strings to soundboard through the bridge. Measurements and simulations show that the input admittance at the bridge depends on the distance between string end and ribs. A second effect of ribs is related to the periodicity of rib spacing. Simulations of soundboard driving-point mobility were conducted for three different spacing rules: periodic, almost periodic with small random fluctuations, and random distribution of spacing within a given interval. The results show that aperiodicity reduces the emergence of strong peaks in the mobility spectrum. Finally, measurements and simulations were made on the radiated sound in an anechoic chamber. The primary effect of ribs here is due to the additional stiffness, compared to a flat plate, which reduces the critical frequency and thus increases the frequency range where the radiated sound power is significant. In addition, the presence of ribs influences the directivity of the sound field: strong differences exist in a plane parallel to ribs compared to ribs perpendicular to them. These results also serve as references for comparison with time-domain simulations of the complete piano including strings, soundboard vibrations, and radiation.

1:20

3pMU2. Modeling and numerical simulation of a piano. Juliette Chabassier and Antoine Chaigne (Unit of Mech. Eng., ENSTA ParisTech, Chemin de la Huniere, 91761 Palaiseau, France, juliette.chabassier@inria.fr)

A complete model of a piano is built, in order to account for the acoustical behavior of the instrument from excitation to sound. A nonlinear hammer strikes the strings. The precursor at the bridge can be explained by the presence of a longitudinal vibration in the string, which is nonlinearly coupled to the transversal vibration using the geometrically exact model. A nonstandard condition must be written for the bridge model, in order to transmit transversal and longitudinal vibrations of the strings to the soundboard. A Reissner Mindlin plate model is used for the soundboard, which radiates in the air. The coupling must be reciprocal so that a global energy is preserved. A numerical discretization is proposed for the whole system. A first difficulty is due to the nonlinearity of both strings and hammer. Another one arises from the different couplings of the system: hammer/string, string/soundboard, and soundboard/air. An innovating, energy preserving scheme is built for the nonlinear system of string equations, and an energy technique is adopted for the whole problem to ensure numerical stability. The resulting complete numerical scheme conserves a discrete and consistent global energy. Numerical results are presented and compared to measurements.

1:40

3pMU3. Experimental study of nonlinear phenomena in an upright piano. Antoine Chaigne and Juliette Chabassier (Unit of Mech. Eng. (UME), ENSTA ParisTech, Chemin de la Huniere, 91761 Palaiseau Cedex, France, antoine.chaigne@ensta.fr)

Measurements are conducted on an upright piano in order to investigate the nonlinear dynamics of the instrument from the blow of the hammer to the radiated sound in the nearfield. The experimental setup consists in synchronous recording of eight channels: hammer force, string displacement at various points, velocity and force at bridges, soundboard acceleration, and sound pressure. From these measurements, the transmission of vibrations through the instrument is followed in both the time and frequency domains. Excitation of strings at different amplitudes shows the influence of the geometrical nonlinear phenomena in the radiated sound. In the time domain, the most visible consequence of nonlinearity is the presence of a precursor in the force and acceleration waveforms at the bridge. To clearly discriminate between dispersion precursor, due to stiffness, and nonlinear precursor, experiments are conducted on two strings corresponding to limiting cases: the C2 string, where stiffness is dominant, and the E3 string where the geometrical nonlinear effects are preponderant. Fine spectral analysis shows that the transmission of energy from strings to soundboard at the bridge is crucial in enhancing the level of phantom partials in the emitted sound due to quadratic and cubic nonlinearities in the string force.

Contributed Papers

2:00

3pMU4. Physical model of the harpsichord plectrum-string interaction during slip-off. Chao-Yu J. Perng (Dept. of Phys., Stanford Univ., 382 Via Pueblo Mall, Stanford, CA 94305), Julius O. Smith, and Thomas D. Rossing (Stanford Univ., Stanford, CA 94305)

A harpsichord plectrum model and the plectrum-string interaction [C.-Y. J. Perng *et al.*, J. Acoust. Soc. Am. **127**, 1733(A) (2010)] with its subsequent implementation using a digital waveguide [C.-Y. J. Perng *et al.*, J. Acoust. Soc. Am. **128**, 2309(A), (2010)] have been proposed by us earlier. In this presentation we model the final moments of the string sliding past the plectrum, a critical stage of the pluck action that provides the distinctive excitation signal. In addition, by incorporating motion data of a real harpsichord jack, we present here a revised model on the harpsichord plucked string mechanism.

2:15

3pMU5. Vibroacoustics of a gothic harp. Chris Waltham, Shira Daltrop, Andrzej Kotlicki, Nathan Wolfe (Dept. of Phys. & Astronomy, Univ. of British Columbia, Vancouver, Br. Colombia V6T 1Z1, Canada, cew@phas.ubc.ca), Benjamin Elie, and François Gautier (LAUM-UMR CNRS 6613-Université du Maine, 72085 Le Mans, France)

The gothic style of harp was popular across most of Europe from the late medieval period to the Renaissance. To study the vibroacoustic behavior of gothic harps, one was constructed from plans created by the Boston Museum of Fine Arts from a late German model in their collection. The vibrational behaviors of the soundboard and soundbox were measured at various stages of construction of the instrument. Once complete, the instrument was subjected to modal analysis and radiativity measurements using Weinreich's method. The sound radiation of this harp is dominated by two breathing modes at 188 and 273 Hz and higher modes around 350 Hz, which together function like the A0/T1 resonance pairs seen in the soundboxes of many other instruments including modern concert harps. As the frequency increases, radiation is emitted from higher up the soundboard and from higher soundholes, as has been observed in other harps. Unlike modern instruments, the gothic harp's thin back plays a large role in sound production.

2:30

3pMU6. Musical controller for wind instrument using Max/MSP software and ATmega128. Anuva Chowdhury, Sangjin Cho, and Uipil Chong (School of Elec. Eng., Univ. of Ulsan, 93 Daehak-ro, Nam-gu, Ulsan, Korea, upchong@ulsan.ac.kr)

The physical appearance of the proposed musical controller is composed of eight buttons, two potentiometers, and one for sensing resistor (FSR). The eight buttons each represent one octave scale. One of the two potentiometers selects kinds of instruments, while the other changes octave. FSR is for tremolo effects. For the real-time play, we used a Max/MSP software. All recorded sounds were sampled and quantized as 44 100 Hz and 16-bit, respectively. To use wavetable synthesis, samples of three to ten periods of the sounds were stored in each table. We also used an AVR board equipped with an ATMega128 8-bit microcontroller. This microcontroller acts as an interface circuit between the sensor and PC as well as transmits data about notes and playing style to the Max/MSP by using a serial communication under RS-232. Considering that the controller is a prototype of the stand-alone musical instrument and is worked on the PC, the latency caused by RS-232 communication can be easily solved embedding the program onto the microprocessor such as digital signal processor. [Work supported by Basic Science Research Program through the National Research Foundation of Korea (NRF) funded by the Ministry of Education, Science and Technology (No. 2010-0005092).]

WEDNESDAY AFTERNOON, 25 MAY 2011 — WILLOW B, 1:00 TO 3:30 P.M.

3p WED. PM

Session 3pNS

Noise, Architectural Acoustics, and Musical Acoustics: Community Noise Impact, Criteria, Mitigation from Outdoor Concert Venues

Joel A. Lewitz, Cochair
Rosen Goldberg Der & Lewitz, 1100 Larkspur Landing, Larkspur, CA 94939

Steven D. Pettyjohn, Cochair
Acoustics and Vibration Group, Inc., 5700 Broadway, Sacramento, CA 95820-1852

Chair's Introduction—1:00

Invited Papers

1:05

3pNS1. Dealing with sound generated by a multi-use stadium when background sound exceeds the limits: Political issues and field results. Steven D. Pettyjohn (The Acoust. & Vib. Group, Inc., 5700 Broadway, Sacramento, CA 95820, spettyjohn @acousticsandvibration.com)

Raley Field is home to the River Cats baseball team, a Triple -A minor league affiliate of the Oakland Athletics. This stadium hosts a variety of events besides the baseball games, including concerts. The sound generated by all of these events is governed by the City of West Sacramentos Noise Control Ordinance and the non-transportation performance standards in the General Plan. However, these regulations make no provision with how to deal with situations where background sound levels exceed the sound limits. Nor do the standards deal appropriately with the difference in the frequencies generated by background sound levels compared with those generated by either baseball games or concerts. An agreement was reached to set a limit based on background sound levels that required the events in Raley Field to generate less sound than the background to limit the increase in the total or ambient sound level. The standards are based on hourly L_{eq} and L_{max} sound level. Examples of community and concert sound measurements made during concerts and baseball games are presented and compared with standards. The proposed development around the stadium will be explained and compared with potential noise impacts at elevated sites.

1:25

3pNS2. Environmental noise study and follow-up measurements at the White River Amphitheatre in Auburn, Washington. Ioana Park and Jeanette Hesedahl (BRC Acoustics & Tech. Consulting, 1741 1st Ave S., Ste. 401, Seattle, WA 98134, ipark@brcacoustics.com)

The 20 000-seat White River Amphitheatre is located in Auburn, WA, approximately 35 miles southeast of Seattle. Built by the Muckleshoot Indian Tribe on reservation property, the project was not subject to jurisdictional noise limits during the permitting phase. However, after the basic structure and roof of the amphitheater were standing, the tribe learned of the Environmental Impact Statement requirement. Utilizing noise limit requirements from the two neighboring Washington State counties, criteria for identifying and evaluating noise impacts were developed for the study purposes. Original source level measurements from Shoreline Amphitheatre in Mount View, CA were reconciled with modeling results to produce sound level contours and predictions for sound levels in the 63-Hz and 125-Hz octave bands. Noise mitigation measures included constructing rear and side walls to the performance and audience area, installation of acoustically absorptive material to the underside of the roof structure, and separate directional time-delayed speakers for lawn seating coverage. This presentation summarizes the modeling process and results, noise mitigation measures employed, design challenges, and sound level monitoring results from the fully operational venue.

1:45

3pNS3. Potential noise impacts associated with concert sound from a proposed football stadium. Richard Carman, Carlos Reyes, and Silas Bensing (Wilson, Ihrig & Assoc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608)

A proposed new stadium for a national league football team was the subject of a recent Environmental Impact Report prepared for a major redevelopment of a former industrial site in San Francisco, CA. The development will include commercial and residential components, which would be adjacent to existing residential and industrial land. Consequently, potential noise impacts were evaluated for existing and future residential receptors. San Francisco recently revised its noise ordinance adopting relatively low limits for community noise, thus posing interesting challenges for new noise sources. The concept for the new outdoor stadium included a fixed sound system for football games that could result in noise impacts as well as those from crowd noise. This paper, however, will focus on the potential for community noise impacts from outdoor concerts that could be held in the stadium using the performer's sound system as is typical in this situation. A three-dimensional acoustical model, which included the local topography, stadium geometry, and representative loudspeaker characteristics, was developed to project sound levels in the residential community. The implications of exceeding the noise ordinance limits during a concert and possible mitigation measures are discussed as are the main details and results of the model.

2:05

3pNS4. Developing an amplified sound policy for San Francisco's Golden Gate Park. Alan T. Rosen (Rosen Goldberg Der & Lewitz, Inc., 1100 Larkspur Landing Cir., Ste. 375, Larkspur, CA 94939, arosen@rgdlacoustics.com)

Golden Gate Park in San Francisco hosts many community events, some with amplified sound. Sharon Meadows, near the east end of the Park, hosts events with amplified sound and is in relative close proximity to residential areas. Over the years, neighbors have expressed concern about the level of amplified sound from events at Sharon Meadows. In order to address neighbors' concerns, a program was initiated by the San Francisco Recreation and Parks Department to develop policies and procedures for monitoring and controlling amplified sound to reasonable levels. The program involved a review of existing city policies for amplified sound, a meeting with affected neighbors to discuss concerns, acoustical measurements during events, and sound propagation testing. The final result was an amplified sound policy that provides limits for sound levels within the park that are based on ambient noise levels in the community. The paper will discuss the development of the policy and the challenges of monitoring and regulating amplified sound in a dynamic urban environment.

2:25

3pNS5. Chastain Park's 63-Hertz compliance criteria: 5 years on. Robert Berens (Acentech, 33 Moulton St., Cambridge, MA 02138)

Over the years, Atlanta's open-air Chastain Park Amphitheater has been the source of enormous friction between the city, the venue's owner, and the wealthy, politically connected residential community abutting the park. In 2005, a study was undertaken to identify the characteristics of concert-event sound to which neighbors were particularly sensitive: Sound levels were monitored throughout the neighborhood during 17 concerts, ranging from quiet jazz and classical performances to rock-and-roll and hip-hop; at a number of locations, measurements were made simultaneously inside and outside homes. The study team confirmed that low-frequency sound was the one feature of concert-related sound that community residents identified as most problematic. A new compliance metric was established to address low-frequency annoyance issues: a two-tiered exceedence threshold, based on 1-min LEQ levels in the 63 Hz octave band measured at the rear of the amphitheater, with a concert-event "exceedence" defined to be either a single 1 min LEQ (63 Hz) level greater than 95 dB or more than ten 1-min LEQ (63 Hz) levels greater than 90 dB. 5 years of data from Chastain's computer-based concert monitoring system are analyzed to show how effective the "new" compliance criteria have proven, and practical experience with compliance monitoring is discussed.

2:45

3pNS6. Locating a new outdoor performance venue amidst public concern and the resulting sound level impacts at the remote amphitheater site. Steven D. Pettyjohn (The Acoust. & Vib. Group, Inc., 5700 Broadway, Sacramento, CA 95820, spettyjohn@acousticsandvibration.com)

An existing small amphitheater was to be enlarged from its 7000 audience size to hold up to 22 000 people. A threatened law suit by the City of Sacramento forced the developer, Bill Graham Presents, to find another site. Several sites were reviewed and a location on the edge of Folsom was first selected then abandoned when many people showed up to protest. The final site selection for the amphitheater was in a remote, rural area where the land was zoned for agricultural use. The surrounding land is used for growing rice and for pasture land for cattle operations. The County has jurisdiction over the land and has very poor noise regulations and a tendency to not enforce what they have. One rancher living north of the amphitheater filed a law suit against the operation of the facility because of deleterious impacts on both her well being and her cows. Sound measurements were made during a single concert representing a hard rock band to quantify the sound levels and the tonal content of the sound during normal operations. This paper provides the history of this process and the results from the sound tests done at the site.

Contributed Paper

3:05

3pNS7. Acoustic characterization of vuvuzelas. Richard Ruhala (Div. of Engr., Southern Polytechnic St. Univ., 1100 S. Marietta Pkwy., Marietta, GA 30060, rruhala@spsu.edu), Kenneth Cunefare (Georgia Inst. of Technol., Atlanta, GA 30332), Tina Ortkiese (Arpeggio Acoust. Consulting, LLC, Hampton, GA 30228), and Laura Ruhala (Southern Polytechnic St. Univ., Marietta, GA 30060)

Vuvuzelas are inexpensive plastic horns that became a part of the American vocabulary during the 2010 World Cup held in South Africa. Spectators were allowed to play these horns during the game, causing major noise complaints from athletes and spectators alike, including television listeners from around the world. The vuvuzelas and similar small, plastic horns are now being sold in the US and other countries, and there is concern that they will create a related noise problem at future sporting events. Several noise measurements have been reported in the literature, but rigorous acoustical experimental methods were lacking. This work presents the measured sound power and spectra produced by several different vuvuzelas, one at a time, inside a hemianechoic chamber. A variety of volunteers have played the horns so that the measurements provide a realistic range of levels. Furthermore, a model is developed to predict noise levels within the stands and on the playing field.

3:20—3:35 Panel Discussion

WEDNESDAY AFTERNOON, 25 MAY 2011 WILLOW A, 1:15 TO 3:15 P.M.

Session 3pPA

Physical Acoustics: General Topics in Linear Acoustics

Andrew A. Piacsek, Chair

Central Washington Univ., Dept. of Physics, 400 E. University Way, Ellensburg, WA 98926

Contributed Papers

1:15

3pPA1. An approximate causal Green's function for the Stokes wave equation. Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2220 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

A causal analytical approximation is derived for the time-domain Green's function of the Stokes wave equation. This causal approximation, which contains a modified Bessel function of the first kind, is the exact analytical time-domain Green's function for a related wave equation that approaches the Stokes wave equation in the low frequency limit. An asymptotic expression is also obtained for this approximate Green's function. The causal approximation and the asymptotic expression are evaluated and compared to the numerically computed time-domain Green's function and the analytical frequency-domain transfer function of the Stokes wave equation and to other causal and noncausal approximations. The results show that this approximate Green's function closely matches the numerically computed Green's function and the analytical frequency domain transfer function of the Stokes wave equation near the source, and at greater distances, the asymptotic expression accurately represents the time- and frequency-domain behavior of the Stokes wave equation. Time-domain analysis demonstrates that the analytical approximation and the asymptotic expression are dispersive, whereas causality is verified in the time-domain and in the frequency-domain. The differences and similarities between these causal approximations and other causal and noncausal approximations are also identified and discussed. [Work supported in part by NIH Grant No. EB012079.]

1:30

3pPA2. Radiation by a subwavelength-sized acoustic monopole: Energy considerations and application to subwavelength time-reversal focusing. Emmanuel Bossy and Rmi Carminati (Institut Langevin, ESPCI ParisTech, CNRS UMR 7587, 10 rue Vauquelin, 75231 Paris Cedex 05, France, emmanuel.bossy@espci.fr)

In this work, the radiation of pulsed acoustic waves by an acoustic monopole is theoretically studied based on energy consideration. Specifically, the instantaneous acoustic intensity and the instantaneous acoustic energy contained in a spherically symmetrical pulsed radiated field are calculated. In the case of a subwavelength-sized acoustic monopole, it is shown that the radiation of a given amount of energy involves energy exchanges between the field and the source that are much larger than the radiated energy. It is also shown that even after the source has been turned off, the instantaneous acoustic intensity may be directed inwards, although the field cannot exchange energy with the source anymore. By time-reversing the radiated field, we demonstrate that subwavelength refocusing requires a subwavelength energy exchanger, which is not only needed to absorb the incoming energy as well-known, but importantly also to transiently provide

3p WED. PM

energy to the field before reabsorbing it. In other words, we demonstrate in a homogeneous medium that the acoustic sink needed to perform subwavelength time-reversal must transiently be an active source of energy.

1:45

3pPA3. Supersonic flows and their interaction with propagating acoustic signals: Acoustic black holes in the laboratory. David J. Goldstein, Gregory J. Orris, and William G. Szymczak (Acoust. Div. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

Work in particle physics and general relativity (GR) has established that deep connections exist between acoustics and GR. Most remarkable is the fact that acoustic wave propagation in fluids is governed by an effective Lorentzian spacetime geometry: Acoustic waves follow the geodesics of a (curved) *acoustic* metric. This provides an entirely new way of looking at acoustic propagation, and in principle provides valuable theoretical tools since much of the machinery developed by the GR community over the past several decades can be directly applied to acoustic systems expressed in this framework. Notably, supersonic liquid flows are predicted to have completely analogous properties to spacetime regions near (gravitational) black holes. We present the status of a developing research program at NRL, designed to begin exploring these connections via laboratory experiments, numerical simulations, and theoretical development. [Work supported by the Office of Naval Research.]

2:00

3pPA4. Coupling layer corrections in pulse echo time-of-flight measurements in solids revisited. Blake T. Sturtevant, Cristian Pantea, and Dipen N. Sinha (Los Alamos Natl. Lab., MPA-11: Sensors & Electrochemical Devices, MS D429, Los Alamos, NM 87545)

It is well known that to extract highly accurate phase velocities from pulse echo ultrasonic time-of-flight measurements, one must take into account the influence of the medium used to acoustically couple the transducer (or buffer rod) to the sample being measured. This correction is typically on the order of 0.1% and is usually achieved by fitting experimental data to a scalar transmission line model in which echoes undergo phase shifts upon transmission through or reflection from the various boundaries. Fitting data to such a model involves varying two free parameters, the thickness of the coupling bond layer and the speed of sound in the sample. When performing the fit, one must make a choice between multiple solutions corresponding to different bond lengths and sample sound speeds. This talk will quantify the importance of the couplant correction in selected cases and discuss the error introduced to the extracted velocities by selecting a "wrong" fit to the data. Additionally, criteria and methods for choosing the "correct fit" without making independent measurements of the bond length will be discussed. [Support for this work was provided by the U.S. DOE.]

2:15

3pPA5. Using relative amplitude and travel times from geometric acoustics to determine nocturnal effective sound speeds. Philip Blom and Roger Waxler (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, psblom @olemiss.edu)

On clear dry nights over flat land, a temperature inversion and stable nocturnal wind structure lead to an acoustic duct in the lowest few hundred meters of the atmosphere. An impulsive signal undergoes strong dispersion during propagation and is received at long ranges from the source as an extended wave train consisting of a series of distinct arrivals followed by a low frequency tail. The leading distinct arrivals have been shown to coincide with the direct and single reflection geometric ray paths. At a range of 2 Km from the source, these ray paths propagate through the lowest 50–70 m of the atmosphere while larger range arrivals reach higher into the atmosphere. Using the solutions of the eikonal and transport equations, travel times, amplitudes, and caustic structures of the arrivals can be determined. Arrival details can then be used to fit a low order Taylor series expansion of the effective sound speed profile. Approximations of the atmospheric conditions in the duct will be presented along with both predicted and measured arrival waveforms and meteorological data from the time of propagation.

2:30

3pPA6. The propagation of sound above a hard-backed porous layer. Sheng Liu and KaI Ming Li (Dept. of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031)

In a recent paper [J. F. Allard, J. Acoust. Soc. Am. **125**, 1864–1867 (2009)], the inaccuracy of the classical formulation for the acoustic field of a point source above a hard-back porous layer was addressed. The current study develops a set of closed form analytical solutions for predicting the propagation of sound due to a monopole near a hard backed layer. The current formulation avoids the use of the effective impedance approximation to obtain asymptotic solutions. The asymptotic formulas are derived by applying the method of saddle path integration supplemented by the method of pole subtraction. The analytical solutions are compared with accurate numerical integrations using the fast field formulation and other accurate numerical quadratures. Predicted results suggest that the analytical formulas agree well with exact numerical solutions. Indoor experimental measurements were also conducted to validate the analytical formulas.

2:45

3pPA7. Flow and geometry induced scattering of high frequency acoustic duct modes. Alex F. Smith, Nick C. Ovenden, and Robert I. Bowles (Dept. of Mathematics, Univ. College London, Gower St., London WC1E 6BT, United Kingdom)

Cut-on cut-off transition of acoustic modes in hard-walled ducts with irrotational mean flow is well understood for Helmholtz numbers of order unity, where previous analyzes have shown that the incident mode undergoes a total reflection with a phase shift of $\pi/2$. Finite-element simulations of this phenomenon, however, appear to indicate the possibility of energy scattering into neighboring modes at a moderately large Helmholtz number, which can then propagate beyond the transition point. In this work, we attempt to explain and predict such scattering phenomena in slowly varying aeroengine ducts using a multiple-scale approach. It is found that, for sufficiently high frequencies, two mechanisms exist whereby energy can be scattered into neighboring modes by an incident propagating mode. One mechanism occurs only when a mean flow is present in the duct and induces scattering at significantly lower frequencies than the other mechanism that remains present even without a mean flow. An efficient system of coupled ordinary differential equations is developed to obtain the corresponding transmitted and reflected amplitudes of the scattered modes as well as the overall acoustic pressure field. Moreover, the theory appears to demonstrate that some interaction and exchange of energy between the acoustic and mean flow fields occur during scattering.

3:00

3pPA8. Acoustic methods for data transmission across a room. David Hutchins, Chuan Li, and Roger Green (School of Eng., Univ. of Warwick, Coventry CV4 7AL, United Kingdom)

Acoustic signals have been used to transmit signals indoors across a room with a data rate which allows many applications. Frequencies in the range 100 kHz–1 MHz have been used, together with modulation techniques used in modern digital wireless communications. It is shown that signals can be transmitted across several meters distance using sources and receivers based on capacitive (electrostatic) designs, where the required bandwidth is avaialable. It is also demonstrated that demodulation techniques can be successfully applied to retrieve the original data from the received signal. The approach has many uses, especially for secure in-room digital communications.

Business Meeting, Plenary Session and Awards Ceremony

George V. Frisk, President

Acoustical Society of America

Business Meeting of the Acoustical Society of America

Presentation of Certificates to New Fellows

Brent W. Edwards
Peter J. Fitzgibbons
J. Brian Fowlkes
Barry M. Gibbs
Sandra Gordon-Salant
Michael G. Heinz
Ronald A. Kastelein
Thomas R. Moore
Benjamin R. Munson
James C. Preisig
Daniel Pressnitzer
Giora Rosenhouse
John J. Rosowski
Stephanie R. Shattuck-Hufnagel
Donal G. Sinex
Christopher W. Turner
Roger M. Waxler
Magdalena Wojtczak
Fan-Gang Zeng

Presentation of Acoustical Society Awards

Medwin Prize in Acoustical Oceanography to Aaron M. Thode

R. Bruce Lindsay Award to Karim G. Sabra

2010 Silver Medal in Speech Communication to David B. Pisoni

Helmholtz-Rayleigh Interdisciplinary Silver Medal to James E. Barger

Honorary Fellowship to Amar G. Bose

Gold Medal to Eric E. Ungar

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. On Tuesday and Thursday the meetings will be held starting immediately after the Social Hours at 8:00 p.m. On Wednesday, two technical committees will meet at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Grand Ballroom A
Signal Processing in Acoustics	Metropolitan A

Session 4aAAa

Architectural Acoustics, Noise, and Committee on Standards: Soundscape: Standardization and Implementation I

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square, Vernon, CT 06066

Gary Siebein, Cochair
Univ. of Florida, Dept. of Architecture, Gainesville, FL 32611-5702

Chair's Introduction—7:30

Invited Papers

7:35

4aAAa1. Harmonization of methods-a contribution to the holistic Soundscape. Brigitte Schulte-Fortkamp (TU Berlin, Einsteinufer 25 10587 Berlin, Germany, brigitte.schulte-fortkamp@tu-berlin.de) and Andre Fiebig (HEAD Acoust., Ebertstrasse 30a, 52134 Herzogenrath, Germany)

The Soundscape approach can provide meaningful information based on the local expertise and allows combining meaningful knowledge and diverse acoustical data about these respective environments. Qualitative methods, which are related to people's mind belong to a heterogeneous research field. Moreover, among them are different forms of observation, interviewing techniques and the collection of documents and acoustical data. At the same time methods are used that are based on various theoretical assumptions and methodological positions. But, those approaches all share common ground and agree on certain ideas about the nature of social reality, which is shaped by social meaning. Social reality is always a meaningful reality and depends on a certain point of view or perspective and is therefore tied to social locations. Based on earlier findings, methods have been proposed for comparing sonic environments based on the physical properties and memories of such environments. When it comes to standardization, there are still many open questions how to combine the different contributors to Soundscape regarding to an analysis that takes physical and perceptive evaluation into consideration. This paper will provide an insight into the evaluation process related to the holistic in Soundscapes.

7:55

4aAAa2. Soundscape analysis standardization, a proposed lexicon of descriptors for local expert interviews. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Sq., Vernon, CT 06066, bbrooks@brooksacoustics.com)

Soundscape analysis is a powerful technique that can be applied to the practical development of quality sonic environments. The combination of physical measurements and the corresponding perceptual responses to a sound climate provides illumination otherwise unavailable to development project designers. Interviews of local experts, residents, and other users of the environment can yield insights into both personal reactions and universal observations. These can include the identification of sound sources of which the project developers were, until then, unaware. The success of a new project may depend on the lessons learned from previous developments. The standardization of interview techniques then becomes increasingly important. Can a standard lexicon of terms for local expert interviews be developed? The answer to this open question is informed by the contributions of psychoacoustic and perceptual evaluations the subtleties of linguistics, the circumstances of the environment under observation, and the persons involved. An interview questionnaire lexicon of descriptors for field use is proposed, based on the experiences of recent project interviews, the results of ASA soundscape workshops, and other sources.

8:15

4aAAa3. Soundscape : Recording the infinite. Alex U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Lowell, MA, 01854, alex_case@uml.edu)

The essential elements of a soundscape are far from defined, but accurately documenting an ensemble of identifiable sound sources across a multitude of spaces is sure to require a wide dynamic range, full frequency response, and the successful capture of the perceptual cues for assessing localization, distance, source size, spaciousness, envelopment, etc. More than 1 century of research and development in sound, recording techniques and technologies can inform soundscape standardization and implementation. The challenge is not only to capture and analyze, but also to reproduce, recreate, and even pre-create the soundscape in an alternative environment. Music, film, and game sound recordists have an advanced set of capabilities for creating meaningful, evocative works of art—both fiction and non-fiction—from the relative comfort, predictability, and convenience of the recording studio production environment. An ambitious attempt to export the highest quality studio capabilities to the field is presented. A mobile, multichannel, high definition, digital recording apparatus has been designed and field-tested, and the most important insights, caveats, accomplishments, and failures so far are detailed.

8:35

4aAAa4. Soundscape analysis: Need for (automatic) source separation. Andre Fiebig and Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany)

The use of psychoacoustics for soundscape analysis is imperative to predict the soundscape from an acoustical point of view. Psychoacoustic parameters as functions of intensity, time structure, and spectral content play an important role with respect to several sensations and yield information with greater differentiation than usually applied indicators within the community noise context. But, with respect to a comprehensive analysis of a soundscape, the exact source constellation must be known. Based on the performance of the human hearing system including intelligent signal processing, a listener can easily focus on a certain source and suppress the noise of other sources, which considerably influences the general appreciation of the whole soundscape. For example, this ability is permanently used to improve speech intelligibility in noisy environments. Therefore, an acoustical soundscape analysis should not just deal with the overall noise, but rather the (psycho-)acoustical properties of the different sources present in a soundscape must be investigated separately. Different methods are conceivable in realizing source separation in complex noise scenarios. The paper will present a few case studies demonstrating the opportunities in separating the contributions of several sound sources within a soundscape.

8:55

4aAAa5. Individual soundwalk methodology for assessment of urban soundscape. Joo Young Hong, Pyoung Jik Lee, and Jin Yong Jeon (Dept. Architectural Eng., Hanyang Univ., Seoul 133-791, Korea, st200045@hanmail.net)

This paper describes the individual soundwalk as a methodology for assessment of urban soundscapes. Compared with previous studies which have adopted soundwalk methodology, the individual soundwalk has significant differences in evaluation and analysis procedure. In the present study, soundwalks are performed individually, and subjects are asked to select the sites which produce a negative or positive impression while walking within a specific area. At each site, the subjects evaluate the soundscape perception using a questionnaire and describe why the urban soundscape is perceived favorably or unfavorably in terms of contexts. Furthermore, audio-visual recordings are carried out and environmental factors such as day light, temperature, and humidity were measured simultaneously. From various responses of the subjects, the dominant contexts that influence the soundscape perception are derived, and design elements of urban spaces are extracted through qualitative analysis.

9:15

4aAAa6. Effects of contexts on perception of urban soundscape. Jin Yong Jeon, Pyoung Jik Lee, and Joo Youn Hong (Dept. Architectural Eng., Hanyang Univ., Seoul 133-791, Korea, jyjeon@hanyang.ac.kr)

In the present study, the effects of contexts on soundscape perception in urban space are investigated through social surveys and soundwalks. Acoustic comfort, visual image, day lighting, olfactory, and reverberance are utilized as main contexts which may have an effect on the soundscape perception. Preference for contexts and overall impressions are evaluated by using an 11-point. numerical scale in the social survey and soundwalk. For qualitative analysis, semantic differential tests are performed in the social survey while subjects are asked to describe their own impressions during soundwalks. The results show that urban soundscapes can be characterized by soundmarks, and soundscape perceptions are dominated by acoustic comfort, visual image, and day lighting. However, it is found that the subjective rating of reverberance in urban spaces is not appropriate. The categories extracted from the qualitative analysis reveal that spatial sensations such as openness and density emerge as primary contexts for soundscape perception.

9:35

4aAAa7. Modelling subjective evaluation of soundscape: Towards soundscape standardization. Jian Kang and Lei Yu (School of Architecture, Univ. of Sheffield, Sheffield S10 2TN United Kingtom)

Based on a series of large scale surveys, this paper first analyzes the effects of social, demographical, and behavioral factors as well as long-term sound experience on the subjective evaluation of soundscape in urban open public spaces. Te paper then explores the feasibility of using computer-based artificial neural network to build models for predicting the soundscape quality evaluation of potential users in urban open spaces at the design stage. It has been shown that for both subjective sound level and acoustic comfort evaluation, a general model for all the case study sites is less feasible due to the complex physical and social environments in urban open spaces, models based on individual case study sites perform well but the application range is limited, and specific models for certain types of location/function would be reliable and practical. It is expected that such a model framework would be useful for the soundscape standardization, in terms of factors to be considered, for example.

9:55—10:05 Break

10:05

4aAAa8. Automated analysis and interpretation of long-term soundscape audio recordings. Robert C. Maher (Dept. of Elec. & Comput. Eng., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717-3780)

Contemporary audio recording devices now provide the capability to obtain uninterrupted digital audio soundscape recordings lasting days, weeks, and months. While storage and transfer of sound file recordings of such extreme length is now possible, human listeners are generally unable to dedicate the time, effort, and cost required to review and interpret aurally all but a tiny fraction of such lengthy files. Instead, a variety of automated processing algorithms are under development to aid in the analysis and characterization of long-term soundscape studies and to present the analysis results to human listeners in an informative and perceptually meaningful manner. This paper reports on three methods for presenting long-term soundscape data: visual spectrographic display, event-based aural sampling, and time-compressed and/or time-lapse aural display.

10:25

4aAAa9. Developing standard sound measurement and analysis methods to support noise management in parks. Kurt M. Fristrup (Natl. Park Service, Natural Sounds and Night Skies Div., 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

Acoustical conditions in National Parks encompass some of the quietest outdoor conditions ever measured, powerful natural sounds, dense aggregations of visitors, and high traffic roads. Most sites exhibit strong seasonality in natural conditions and visitation. Accordingly, the National Park Service has developed monitoring systems that can accommodate a wide range of conditions, sustain extended periods of continuous data collection, and support a diverse array of metrics. Several years of acoustical monitoring have yielded sound level measurements from 189 sites in 42 units, and measures of noise audibility for 93 sites in 22 units. Comparative analysis of these data revealed substantial diel patterns that are shared across low- and high-noise sites, as well as conspicuous exceptions that can be explained by special conditions at some parks.

10:45

4aAAa10. Modeling soundscapes: A process of analysis by anticipated sonic content. Adam D. Bettcher (Sch. of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611, abettch@ufl.edu)

Through the process of constructing sound maps or soundscapes, the acoustician attempts to describe a particular aural condition using noise-related metrics. Typically, raw-number acoustical data are one means of describing aural conditions; A-weighted sound levels, time-sensitive day-night levels, and octave frequency band analysis are commonly used means of attempting to describe a particular environment by using measurements of sound energy. Many other studies suggest that additional observations of the content of the sonic environment are important when developing descriptors of a given condition. This paper describes a process by which the soundscape is measured and subsequently modeled. Raw acoustical data, in combination with listener observations, are used as a basis for the construction of a probabilistic model that contains both predictions of sonic energy and information on the aural content of a given condition. This combined model is then used to describe and predict aural conditions with a greater potential range of descriptors than those based on raw-number acoustical data alone.

11:05

4aAAa11. Documenting the soundscape of a historic working farm. Keely M. Siebein (Dept. of Architectural Acoust., Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611, ksiebein@siebeinacoustic.com)

A working 1880's farm is the focus of an exploration to provide an acoustical record of historical sounds that once comprised the soundscape. Long term acoustical measurements (including overall A-weighted sound levels, minimum and maximum levels taken once per minute, SEL, statistical metrics, and day/night sound levels), short term measurements (including overall A-weighted and 1/3 octave band sound levels of specific acoustic events), WAV recordings, and video footage are used to analyze typical historic sounds of the farm including a blacksmith forging metal, woodworkers using hand tools to carve wood, children playing period games, lumberjacks shaping logs for buildings, and ranchers cracking whips to drive animals. These historic sounds are documented, along with sounds of the 21st century, including automobile traffic from nearby roadways, aircraft overflights, and various modern machines used in the park to provide a detailed analysis of what a typical rural southern farm soundscape may have sounded like over 100 years ago.

11:25

4aAAa12. Measurement of preserved structures as a resource for historic soundscape reconstruction. Eric Reuter (Reuter Assoc., LLC, P.O. Box 4623, Portsmouth, NH 03802, ereuter@reuterassociates.com)

Acoustical attributes of preserved historic structures have been measured using contemporary techniques and metrics. These include outdoor-to-indoor and indoor-to-indoor impulse response and noise reduction measurements. Combined, these measurements allow for simulation, both quantitative and qualitative, of the indoor soundscapes of the past. These data will be presented, along with audio examples of historical noise sources, as experienced from indoors.

Session 4aAAb

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

Robert C. Coffeen, Cochair
Univ. of Kansas, School of Architecture and Urban Design, Lawrence, KS 66045

Andrew N. Miller, Cochair
BAi, LLC, 4006 Speedway, Austin, TX 78751

The Technical Committee on Architectural Acoustics of the Acoustical Society of America, The Robert Bradford Newman Student Award Fund, and the National Council of Acoustical Consultants are sponsoring the 2011 Student Design Competition that will be professionally judged at this meeting. The purpose of this design competition is to encourage students enrolled in architecture, engineering, physics, and other university curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significant importance.

The competition is open to undergraduate and graduate students from all nations. Submissions are poster presentations that demonstrate room acoustics, noise control, and acoustic isolation techniques in building planning and room design. The submitted designs will be displayed in this session and they will be judged by a panel of professional architects and acoustical consultants. An award of $1250.00 US will be made to the entry judged "First Honors." Four awards of $750.00 US will be made to each of four entries judged "Commendation."

THURSDAY MORNING, 26 MAY 2011 ISSAQUAH, 8:25 TO 11:55 A.M.

Session 4aAB

Animal Bioacoustics and Underwater Acoustics: Use of Sound Propagation Modeling as a Tool in Bioacoustic Studies

Elizabeth T. Kusel, Cochair
Oregon State Univ., NOAA/PMEL, 2030 S.E. Marine Science Dr., Newport, OR 97365

Jon M. Collis, Cochair
Colorado School of Mines, Mathematical and Computer Science, Golden, CO 80401

Chair's Introduction—8:25

Invited Papers

8:30

4aAB1. Sound propagation modeling in the ocean for bioacoustics. Michael B. Porter (HLS Res., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

A variety of models are available to predict how sound is affected by the ocean channel. The precise waveform may be propagated, leading to distortions within the ocean itself, as well as reflections from the ocean boundaries, and sub-bottom layers. Alternatively, the models can directly predict simpler measures of the waveforms, such as their loudness. The approaches will be surveyed with an ear to which are appropriate for different applications, including marine mammal tracking and modeling soundscapes.

8:50

4aAB2. Applications of environmental assessment models for bioacoustic research. Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201), David Mountain (Boston Univ., Boston, MA 02215), Michael Porter (HLS Res. Inc., La Jolla, CA 92037), and Dorian Houser (Biomimetica, Santee, CA 92071)

In recent years, the authors of this paper have been incorporating acoustic propagation models along with animal behavior models into an open source software package used to assess the risk of anthropogenic sound on marine life. This software package is referred to as the effects of sound on the marine environment (ESME). With little modification, these ESME tools can be applied to address ocean animal bioacoustic research problems. For example, a typical ESME simulation scenario might involve a sonar transmission, propagation, and reception on a field of simulated marine animals (or animats). However, by simply substituting the sonar transmission for an animal call, the receptions from an animal could be simulated on a passive acoustic monitoring (PAM) system. Trivial modifications can be made to include more complicated PAM systems with arrays of hydrophones. This "ESME-in-reverse" approach might be used to estimate important PAM performance metrics such as signal-excess and detection probability. These kinds of performance prediction tools are commonly used in decision aids for navy sonar systems. In this presentation, the ESME tools will be described along with a discussion of how these could be applied to develop PAM decision aids. [Work supported by the ONR.]

9:10

4aAB3. Using model-based tracking methods to reduce position uncertainty. Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawaii, 250 Dole St., Honolulu, HI 96816)

Many marine mammal tracking methods rely on the assumption of a homogeneous sound speed profile. This assumption greatly simplifies the localization problem since solutions can be re-cast as a set of linear equations with closed form solutions for well-defined problems. However, it also leads to increased uncertainties in position estimates. As long as they are accounted for and understood, these uncertainties are often acceptable and the benefits of homogeneous sound speed assumptions outweigh the drawbacks. However, in some cases and applications (such as in shallow water or over long ranges), it is beneficial to use model-based tracking methods that incorporate sound speed profiles. This talk explains some of the concepts and practical issues associated with model-based tracking. It uses simulations and real datasets to investigate and demonstrate the effects of homogeneous sound speed profile assumptions on animal position estimates. [Work supported by ONR.]

9:30

4aAB4. Augmenting probabilistic inversion for whale localization with propagation models. Scott D. Frank (Dept. of Mathematics, Marist College, 3399 North Ave., Poughkeepsie, NY 12601) and Aaron N. Ferris (Weston Geophysical Corp., Lexington, MA 02420)

Tracking of marine mammals is important for the evaluation of species geographic range and for investigating the relationship between vocalization and behavior patterns. The use of ocean bottom seismic and hydrophone arrays is gaining interest in marine mammal studies due to their relatively long deployments (6–12 months) and low environmental impact. A Bayesian inversion method generates a probability density function for the source location by comparing travel time differences observed at ocean bottom arrays with those from a theoretical model. The method also accounts for uncertainty arising from difficulties in evaluating arrival times, instrument location, or choice of theoretical model. This method can be augmented with a similar comparison of observed arrival amplitudes and transmission loss results from parabolic equation propagation modeling. Due to acoustic convergence zones, the amplitude data may specifically contain information regarding the source's range from the array. Uncertainty in the propagation modeling can also be included in the inversion. Results from travel-time only inversions and those that include propagation modeling results will be presented for comparison.

Contributed Papers

9:50

4aAB5. Using diffuse ocean ambient noise to synchronize independent elements of a horizontal array. Melania Guerra, Aaron M. Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr. La Jolla, CA 92093-0238, melania@mpl.ucsd.edu), Jorge Urban (Laboratorio de Mamiferos Marinos, Universidad Autonoma de Baja California Sur, Apartado Postal 19-B, 23080, La Paz, Mexico), and Steven Swartz (Cetacean Res. Assoc. 14700 Springfield Rd., Darnestown, MD 20874)

Autonomous acoustic recorder packages have been deployed as two horizontal arrays in San Ignacio Lagoon, Mexico, during February and March 2010, to record gray whale (Eschrichtius robustus) sounds. This underwater acoustic environment contains multiple directional anthropogenic sources (passing boats) and numerous biological sources (snapping shrimp) evenly distributed in azimuth around the array systems, in the 0.1–5 kHz frequency range. The local bathymetry surrounding the arrays is highly range-dependent, and the distribution of noise sources is not evenly distributed with azimuth, so extracting a Green's function estimate from the ambient noise is not feasible. However, extracting information about the relative timing between the independent recorders may be. Estimates of the time derivative of the time-averaged cross correlation between independent elements of each array were computed using both types of continuous ambient noise sources to solve for the recorders relative time offset, clock drift, and physical separation. Acoustic bearings obtained from each array were then used to estimate the two-dimensional position of signals of interest. The ability of the resulting synchronized array system to acoustically localize and track sources is tested using GPS tracks of passing tourist pangas, as well as tracks of a tagged whale carrying both bio-acoustic and GPS tags.

10:05—10:25 Break

10:25

4aAB6. Techniques in sound propagation modeling for noise impact assessment in three-dimensionally variable environments. Melanie E. Austin and Alexander O. MacGillivray (JASCO Appl. Sci., Victoria, BC V8R 6P3, Canada, melanie.austin@jasco.com)

Sound generated in the ocean by human activities has the potential to be disturbing or harmful to marine fauna. Environmental assessments of offshore industrial activities must often include noise impact studies for regulatory approval. Such noise impact studies examine the spatial extent of underwater noise using a computer model to estimate the propagation loss between a noise source and a grid of surrounding receiver points, providing the information necessary for the effective assessment and management of anthropogenic marine noise. Models based on the solution of the parabolic form of the wave equation can be used to generate very accurate acoustic field estimations in range-dependent environments. While parabolic equation (PE) codes can also be configured to accurately handle three-dimensional (3-D) propagation effects, the numerical techniques they require can be very computationally intensive. Pseudo-3-D PE solutions are typically computed to prevent prohibitively long model run times. The implications of applying different PE modeling techniques for environmental noise impact assessment in coastal environments are presented. The factors considered include the neglect of or inclusion of 3-D propagation effects in the model, and the consideration of directional noise sources in 3-D variable environments.

10:40

4aAB7. Estimating bowhead whale communications space using measured and modeled data. Jeff MacDonnell and Bruce Martin (JASCO Appl. Sci., 32 Troop Ave., Halifax, NS, B3B 1Z1, Canada, jeff.macdonnell @jasco.com)

In summer 2009, JASCO deployed ocean bottom hydrophones (OBHs) in the Chukchi Sea to measure natural and anthropogenic noise levels and monitor marine mammals acoustically. Acoustic data were collected from early August to mid October 2009. Ambient data were analyzed to compute noise levels for two sites: a relatively loud and relatively quiet one. Computer modeling was used to determine transmission losses at those sites. Manually acquired statistical data were used to determine average call length, and upper and lower frequencies for bowhead calls. Localization techniques were also used to determine source levels of the calls (Ref. Jonathan/Jeff paper). Using the above information and a modified version of the sonar equation, the space within which bowhead calls remain above ambient can be determined as an indication of maximum communication space. The noise conditions can be modified by simulating a ship passing through the area of an OBH and observing the change in communication space.

10:55

4aAB8. Two-dimensional localization of walruses in shallow water using a ray-tracing model. Xavier Mouy (JASCO Appl. Sci. Ltd., 2101-4464 Markham St., Victoria, BC V8Z 7X8, Canada), Mikhail Zykov, and Bruce S. Martin (JASCO Appl. Sci. Ltd., Halifax, NS B3J 1R9, Canada)

Pacific Walruses summering in the eastern Chukchi Sea produce underwater sound pulses called knocks. Chukchi Sea is a flat shallow environment that favors reflections of the sound on the surface and bottom interfaces. In acoustic data collected in the Chukchi Sea in 2007, the knocks emitted by walruses were recorded with up to seven bottom/surface reflections. Relative multipath arrival times (RMATs) provide very valuable information that can be used to define range and depth of the vocalizing animal. This paper demonstrates a method for localizing walruses based on a ray-tracing model. BELLHOP was used to simulate the RMATs received by a hydrophone for hypothetic acoustic sources located at different range and depth positions. The model assumed flat bathymetry, a source emitting at 1 kHz and a single range-independent sound speed profile per monitoring station. An index representing the match between the measured echoes and the model was calculated at each point of the model grid. The highest matching index indicated the depth and range of the walrus. Several walrus tracks were extracted from the data. Localization results allowed defining diving patterns, depths of bell-like calls, source levels estimates, knock production rate, and swimming speed of monitored walruses.

11:10

4aAB9. Reconstructing echolocation behavior using time difference of arrival localization and a distributed microphone array as a virtual Telemike. Jeffrey M. Knowles, Jason Gaudette, Jonathan R. Barchi, and James A. Simmons (Dept. of Neurosci., Brown Univ., 89 Waterman St., Providence RI 02912)

Bats use echolocation to probe their 3-D environment. New techniques in distributed microphone array processing allow us to record echolocation behavior, reconstruct the flight paths, and generate an acoustic record of individual bats flying the laboratory or the field. Probabilistic methods of time difference of arrival localization allow us to follow bats navigating in flight room mazes, foraging in the wild and using active sonar in the presence of possible jamming from conspecifics. By reversing the localization operation, we back out the changing propagation times of the sounds emitted by bats flying fast in three dimensions. Through beamforming, we then create an amplified acoustic track that follows the reference frame of the bat, amounting to a virtual Telemike based on consensus data from the microphone array. Spatial tracking and virtual on-board recording allow us to ask new questions about the spectral and temporal strategies employed by echolocating bats and to model the neural and perceptual natures of sonar. [Work supported by ONR and NSF.]

11:25

4aAB10. A simple autocorrelation method for multipath localization techniques. Robert D. Valtierra (Dept. Mech. Eng., Boston Univ., 110 Cummington St., Boston MA 02215, rvaltier@bu.edu), Sofie M. Van Parijs (Northeast Fisheries Sci. Ctr., NOAA Fisheries, 166 Water St., Woods Hole, MA 02543), and R. Glynn Holt (Boston Univ., 110 Cummington St. Boston, MA 02215)

A simple method was developed to extract time of arrival information from overlapping direct and reflected acoustic signals using an autocorrelation algorithm in order to localize marine mammals with the direct-reflected time difference of arrival (DRTD) technique. DRTD has up to now been limited to pulsed signals such as whale or dolphin clicks where individual direct and reflected signal arrival times are separated in time and thus easily identified. Using autocorrelation, longer duration signals such as a frequency sweep or whale up-call could be analyzed by DRTD, expanding its compatibility to a larger range of signal types. This correlation-modified DRTD technique was applied to simulated and previously acquired experimental data. Both the simulations and experiments represented localization cases involving bottom-mounted receivers with a single acoustic source at multiple known locations. Experimental data came from autonomous acoustic recording units deployed in the Stellwagen National Marine Sanctuary with an acoustic transducer used to broadcast frequency sweeps.

11:40

4aAB11. Tracking dolphins with hydrophone arrays deployed from the floating instrument platform R/P FLIP in the Southern California Bight. Martin Gassmann, E. Elizabeth Henderson (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0205), Marie A. Roch (San Diego State Univ., San Diego, CA 92182-7720), Sean M. Wiggins, and John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093-0205)

Dolphin movements were studied with hydrophone arrays and visual observations using the R/P FLIP moored North of San Clemente Island in Fall 2008. A total of 14 hydrophones distributed as two L-shaped arrays at 36 m depth and one vertical line array at 139 m depth were deployed from FLIP. The data were sampled at 192 kHz continuously for 4 weeks. Sound localizations were realized by estimating vertical and horizontal angles from automatically detected dolphin echolocation clicks and whistles to compute ranges, depths, and bearings. While angles to broadband clicks were estimated by cross-correlation, angles to narrow-band whistles were obtained by coherently frequency-averaging conventional frequency domain beamformer outputs. Sound refraction errors were corrected using Snell's law. The localization methods were groundtruthed by successfully tracking ships and by comparing acoustic and visual positions for dolphin groups. First results reveal continuous trajectories of common dolphin schools for as long as 15 min and suggest that common dolphin actively approaches the FLIP from at least 1 km distance. Common dolphins also appear to be capable of echolocating man-made seafloor instruments at 350 m depth by diving from the surface to approximately 139 m depth while discontinuing whistling.

Session 4aBA

Biomedical Acoustics: Biological Applications of Ultrasound

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Contributed Papers

8:00

4aBA1. Vulnerable atherosclerotic plaque in human artery sections and apolipoprotein E-deficient mouse aortas. Pavlos Anastasiadis and John S. Allen, III (Dept. of Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 302, HI 96822, pavlos@hawaii.edu)

Atherosclerosis is a systemic disease developing over the course of many years. Vulnerable atherosclerotic lesions may rupture without warning resulting in acute cardiac infarction. An outstanding need exists for accurately predicting the plaque sites most prone to rupture. However, few studies have investigated the mechanical properties of plaque in nonfixated human samples or the properties as a function of plaque development in animal models. The size and fragility of plaque in mouse models limit standard mechanical measurement techniques. Elastic and mechanical parameters of atherosclerotic lesions were determined by scanning acoustic microscopy at center frequencies of 50 and 100 MHz. The comparative study included human femoral, aortic and coronary sections, as well as aortic sections from normal and apolipoprotein E-deficient mice. Nonfixated samples were investigated and subsequently fixated for complementary fluorescence microscopy imaging with atherosclerosis-related biological markers. Elastic parameters show significant differences between normal and diseased tissues. Ultrasound velocities across the intima, media, and adventitia of the artery cross-sections vary in the range from 1626 ± 24, 1680 ± 20, and 1640 ± 18 m/s, respectively. Moreover, the plaque lesions show a higher velocity of 1890 ± 27 m/s. [Research supported by NIH/NCRR Grant No. 2G12RR0030161-21.]

8:15

4aBA2. Smoking effects on cyclic variation of harmonic echogenicity in carotid artery. Ying Li, Tae-Hoon Bok, Dong-Guk Paeng (Dept. of Ocean System Eng., Jeju Natl. Univ., Jeju, 690756, South Korea, li_ying740@hotmail.com), Min-Joo Choi (Jeju Natl. Univ., Jeju 690756, South Korea), and Jeong-Hwa Yang (Cheju Halla College, Jeju 690798, South Korea)

The dynamic changes of echogenicity in common carotid artery was observed from coded harmonic images by GE ultrasound systems. The observed variation in echogenicity was related with the dynamic change of red blood cell (RBC) aggregation during a cardiac cycle. In this research, the cyclic variation in echogenicity (CVE) was observed in the harmonic images of the carotid artery from 28 smokers before and after smoking using a GE Voluson e ultrasound system. The amplitude of CVE (A_{cve}) was significantly increased after smoking in the heavy smokers compared with the one in light smokers (4.98 ± 4.26 vs -0.3 ± 3.01, $P<0.05$). A_{cve} was also compared among 17 nonsmokers, 11 light smokers, and 17 heavy smokers. The effects of tissue attenuation on A_{cve} were compensated with body mass index (BMI). Compensated A_{cve} is significantly higher in heavy smokers and light smokers compared with that in nonsmokers (36.7 ± 9.5 and 35.1 ± 5.1 vs 29 $.3\pm7.9$, $P<0.05$). The effects of smoking on CVE were discussed with hemodynamic and hematological changes caused by smoking. The increased RBC aggregation due to the flow acceleration and hematological changes were suggested as interpretations. It demonstrated the potential of A_{cve} to estimate RBC aggregation *in vivo*. [Work supported by Grant No. NRF-2010-0014004.]

8:30

4aBA3. Correlation between the pulse wave and the blood echogenicity in the radial artery. Tae-Hoon Bok, Ying Li, Dong-Guk Paeng (Dept. of Ocean System Eng., Jeju Nat'l Univ., 66 Jejudaehakno, Jeju 690-756, Republic of Korea), and Hee-Jung Kang (Hanyang Univ., Gyeonggi-do 426-791, Republic of Korea)

Pulsatile flow generated by the cardiac motion results in the complicated hemorheological phenomena in the artery. The pulse wave in the radial artery is utilized for the measurement of tonometry and the diagnosis of various diseases by the oriental medical doctors. In this paper, the pulse wave obtained from the three-dimensional (3-D) pulse wave analyzer was compared with the blood echogenicity and vessel diameter in the radial artery using an ultrasound biomicroscopy system. The ultrasonic data were acquired for five subjects and the left wrist of each subject was put on the wrist-rest of the 3-D pulse wave analyzer measuring the pulse waves for 25 s. The repetitive measurement was performed eight times for one of the subjects. The blood vessel motion and the blood echogenicity were calculated from the cross sectional *B*-mode images of the radial artery. The cyclic variation of the vessel diameter and the blood echogenicity was measured during cardiac cycles. The phases between both of them were opposite each other, and the cyclic variation of the vessel diameter showed the similar pattern with that of the pulse wave. These correlations can be helpful for the diagnosis of the cardiovascular diseases. [Work was supported by Grant no. NRF-2010-0017078.]

8:45

4aBA4. High intensity focused ultrasound for cyclocoagulation: From transducer design to early clinical trials. Cyril Lafon (Inserm, U556, Lyon F-69003, France, cyril.lafon@inserm.fr), Florent Aptel (Edouard Herriot Hospital, Lyon F-69003, France), Thomas Charrel, Alain Birer, Françoise Chavrier, Jean-Yves Chapelon (Inserm, U556, Lyon F-69003, France), Fabrice Romano (EyeTechCare, Rillieux la Pape F-69140, France), and Philippe Denis (Croix Rousse Hospital, Lyon F-69317, France)

Many therapeutic procedures have been proposed for treating glaucoma, which remains refractory in many cases. Although HIFU treatments were successful [Lizzi *et al.*], laser-based approaches proved to be more clinically feasible. The new proposed ultrasonic device aims to perform a conformal, partial, and fast thermal ablation of the ciliary bodies for reducing the production of aqueous humor and intraocular pressure (IOP). The design of the device was based on numerical simulations and consideration of anatomical constraints. The device has six cylindrical 21 MHz transducers for modulating and producing sharp zones of ablation. The applicator was characterized *in vitro* and tested *in vivo* in rabbits. Exposure conditions were set to 6 W/cm^2 for 3 s applied sequentially on the transducers. A clinical pilot study was then started to assess the feasibility, the tolerance, and the efficacy of the treatment of refractory glaucoma. Results of *in vivo* experiments showed significant decrease of IOP and no adverse effects on histology. Eight patients were treated with similar exposure conditions. The absence of

side-effects and the significant decrease of IOP, from 36.2 to 25.2 mm Hg 6 months after treatment, proved the clinical relevance of this new device for treating refractory glaucoma. [Work funded by EyeTechCare SA.]

9:00

4aBA5. Effects of diffraction correction on local attenuation estimation. Goutam Ghoshal, Roberto J. Lavarello, and Michael L. Oelze (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews, Urbana, IL 61801)

Quantitative ultrasound (QUS) techniques have been widely used to estimate morphological and mechanical properties of tissue microstructure for specified regions of interest (ROIs). QUS parameters are obtained by fitting theoretical models to backscatter coefficient (BSC) estimates. To estimate BSCs accurately, precise estimation of attenuation coefficient is necessary. A common assumption when estimating attenuation coefficients using focused transducers is that diffraction and attenuation effects can be decoupled. In this work, the validity of this assumption is investigated for various focused transducer geometries. The transducer beam pattern is modeled using multi-Gaussian beam model and O'Neil's theory. The backscatter response is then modeled using the Wigner transform of the beam pattern in conjunction with the stochastic wave equation within the single scattering formalism. Experimental results are shown from tissue mimicking agar phantoms with embedded glass beads. The results indicate that the percentage error in estimating local attenuation increases when ROIs are away from the focal zone in the sample. The results obtained using the Multi-gaussian beam model were then compared with other algorithms generally used to estimate local attenuation. [This work is supported by NIH R01EB008992.]

9:15

4aBA6. Experimental measurements of ultrasonic scattering for characterization of porous media. Elaine S. Vejar (Dept. of Biomedical Eng. and Biotechnology, Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, elaine_vejar@student.uml.edu), Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MA 01854)

Experimental measurements of ultrasonic scattered field from porous media representative of biological tissue such as trabecular bone are discussed in this work. The experiment consists of a water tank containing degassed water at room temperature where the phantom to be analyzed is immersed. The sample is insonified using a 5-MHz unfocused transducer (ULTRAN WS37-5) driven by a pulser/receiver unit (Panametrics-NDT 5077PR) at a pulse repetition frequency of 200 Hz. The backscattered field is measured at a millimeter spatial resolution in the lateral direction and recorded using an oscilloscope (Agilent 54621S). In addition, the angular dependence of the scattered field at a fixed radius is also obtained. Measurements are compared to results from computational methods that provide estimates of the scattered pressure field under strong scattering conditions. The motivation of this work is to better understand trabecular bone scattering properties for clinical use in the diagnosis of osteoporosis.

9:30

4aBA7. Effect of low intensity pulsed ultrasound on mesenchymal stem cells. Jia-Ling Ruan, Yak-Nam Wang, Lawrence A. Crum, and Stuart B. Mitchell (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA 98125)

Low intensity pulsed ultrasound (LIPUS) has been used to accelerate tissue regeneration; however, the biological mechanisms of LIPUS induced regeneration is not completely understood. The aim for this study is to elucidate the mechanical effect generated by US for the stimulation of mesenchymal stem cells (MSCs). MSCs were cultured on flexible cell culture membranes and stimulated by US for 10 min daily with acoustic intensities of 0, 6, 13.5, and 22.5 W/cm^2. Cell proliferation and viability were evaluated by direct cell count and Alamar Blue assay. Morphological evaluation was performed and cell-matrix interactions were evaluated. Cell-matrix interaction was analyzed by immunochemical staining of focal adhesion proteins. LIPUS enhanced cell proliferation at higher intensities and there was an increase in cell viability after 4 consecutive days of US treatment. No morphological changes were observed in all treatments. Expression of focal adhesion protein, vinculin, was enhanced after 3 consecutive days of ultrasound treatment. Studies of media agitation did not show any enhancement effect in cell proliferation or focal adhesion protein expression. The results validates that US is able to influence the cell matrix interaction. Application of higher acoustic pressure on cell growth environment can stimulate MSC proliferation and focal adhesion.

9:45—10:00 Break

10:00

4aBA8. Cavitation damage during sonothrombolysis using high intensity focused ultrasound. Hope L. Weiss (Dept. of Mech. Eng., Univ. of California, 6112 Etcheverry Hall, Berkeley, CA 94720, hope@me.berkeley.edu), Golnaz Ahadi, Thilo Hoelscher (Univ. of California, San Diego, CA 92103), and Andrew J. Szeri (Univ. of California, Berkeley, CA 94720)

High intensity focused ultrasound (HIFU) has been shown to accelerate thrombolysis, *in vitro* and *in vivo*, for treatment of ischemic stroke. Stable and inertial cavitations are thought to play an important role in sonothrombolysis, even though the mechanisms are not fully understood. Possible mechanisms associated with both stable cavitation (i.e., microstreaming) and inertial cavitation (i.e., microjets) are thought to increase clot lysis by enhancing the delivery of a thrombolytic agent. The damage to a blood clot\textquoteright{}s fiber network from bubble collapses in an HIFU field is studied. The region of damage caused by a single bubble collapse on the fiber network of the blood clot exposed to HIFU is estimated and compared with experimental assessment of the damage. The mechanical damage to the network is estimated using two independent approaches: a fiber deformation based method and an energy based method. Whole human blood clots under flow conditions are exposed to 220 kHz ultrasound using the ExAblate 4000. During HIFU exposure, passive cavitation detection is performed using a wide band (10 kHz–15 MHz) hydrophone. Scanning electron microscopy is used to assess the region of damage experimentally. [Work supported in part by NSF Graduate Research Fellowship.]

10:15

4aBA9. Bubble dependence of the mechanism FUS-induced blood-brain barrier opening in mice *in vivo*. Yao-Sheng Tung, Vlachos Fotios, Kirsten Selert, and Elisa Konofagou (Dept. of Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave., New York, NY 10027)

The activation of bubbles by an acoustic field has been shown to temporarily open the blood-brain barrier (BBB) but the physical effects responsible for BBB opening remain unknown. In this study, the physical mechanism of the FUS-induced BBB opening with monodispersed microbubbles is unveiled. Sixty-seven mice were injected intravenously with bubbles of either 1–2, 4–5 or 6–8 μm in diameter. The right hippocampus of each mouse was then sonicated using focused ultrasound (1.5-MHz frequency, 100-cycle (67 μs) pulse length, 10-Hz pulse repetition frequency, and 1 min duration). Peak-rarefactional pressures of 0.15, 0.30, 0.45, or 0.60 MPa were applied to identify the threshold of BBB opening and inertial cavitation (IC). It was found that the BBB opens with nonlinear bubble oscillation when the bubble diameter is similar to the capillary diameter and with inertial cavitation when it is not. The bubble may thus have to be in contact with the capillary wall to induce BBB opening without IC. The BBB opening was induced safely with nonlinear bubble oscillation at the pressure threshold for opening while its volume was highly dependent on both the acoustic pressure and bubble diameter. [This work was supported by NIH Grant No. R01EB009014 and NSF Grant No. CAREER 0644713.]

10:30

4aBA10. Ultrasound-induced permeability variations in cell gap junctions for drug delivery. Pavlos Anastasiadis and John S. Allen, III (Dept. of Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 302, HI 96822, pavlos@hawaii.edu)

An improved understanding of endothelial cell permeability changes at the molecular level has the potential to significantly improve our insights into inflammation and treatment of associated cardiovascular diseases. The relationship between integrin-mediated variations in gap junction behavior

and the empirically observed ultrasound-induced pore formation on the surface of plasma membranes are not well understood. More rigorous experimental comparisons with statistics with regard to pore formation and pore size distributions are needed to discern the underlying molecular mechanisms in order to optimize drug delivery applications. Human coronary artery endothelial cells (HCAECs) were grown to confluency. Long-term electric cell-substrate impedance sensing measurements were conducted in real-time and under physiological conditions. The HCAECs were subsequently irradiated with ultrasound over a range of clinically relevant parameters. Complementary scanning electron microscopy was conducted to quantify pore formation. The antibodies to integrins $\beta 3$ and $\alpha 2\beta 1$ change endothelial permeability from the normal values of $3.8 \pm 0.5\ \Omega\,\mathrm{cm}^2$ to $2.4 \pm 0.6\ \Omega\,\mathrm{cm}^2$. For an acoustic pressure of 0.1 MPa, the mean pore size was found to be 0.019 $\mu\mathrm{m}^2$ while at a pressure level of 0.3 MPa the corresponding value increased to 0.024 $\mu\mathrm{m}^2$ with considerable differences in the respective size distributions.

10:45

4aBA11. Self-assembly of ultrasonically synthesized gold nanoparticles on protein microbubbles for multiple imaging applications. Francesca Cavalieri, Meifang Zhou, Boon Mian Teo, Franz Grieser, and Muthupandian Ashokkumar (School of Chemistry, The Univ. of Melbourne, Parkville, Victoria 3010 Melbourne, Australia)

Air-filled lysozyme microbubbles can be synthesized in an aqueous medium by emulsification followed by cross-linking of protein molecules under high-intensity ultrasound. The efficiency of formation, size distribution, and morphology of the microbubbles can be controlled by manipulating the experimental conditions, namely, power and length of sonication [Zhou *et al.*, Soft Matter (in press); Med Cavalieri *et al.*, Current Topics in Medicinal Chemistry, **12**, 1198–1210 (2010)]. Gold nanoparticles have been sonochemically synthesized and stabilized with bovine serum albumin and polyvinylpyrrolidone. Gold nanoparticles (NPs) have been assembled on the surface of microbubbles in a deposition process. The functionalization of the surface of bubbles endowed the microbubbles with gold plasmon absorption and fluorescence emission. The NP-coated bubbles showed enhanced response when irradiated by 1 MHz ultrasound. We envisage that these stable, multifunctional microbubbles may have important applications in medical diagnostics where multiple imaging methods using a single contrast agent can be advantageous.

11:00

4aBA12. Use of high intensity focused ultrasound for localized activation of thermosensitive liposomes for drug delivery. Christophoros Mannaris, Eleni Efthymiou (Dept. of Mech. and Manufacturing Eng., Univ. of Cyprus, 75 Kallipoleos, 1678 Nicosia, Cyprus), Jean-Michel Escoffre, Ayache Bouakaz (Universit François Rabelais, Tours, France), Vera A. Khokhlova, Sergey A. Ilyin (Moscow State Univ., Moscow, Russia), and Michalakis A. Averkiou (Univ. of Cyprus, 1678 Nicosia, Cyprus)

Localized drug delivery holds great promise in improving drug efficacy in cancer treatment. Newly developed temperature-sensitive liposomes (TSLs) loaded with doxorubicin have been shown to release their payload with mild hyperthermia near their phase transition temperature (T_m = 43–45 °C). In the present work, high intensity focused ultrasound is used to induce the required temperature elevation for the release of the drug from TSLs. A theoretical model based on Pennes' bioheat equation was initially used to calculate the conditions for temperature elevation in fluids and tissue phantoms under conditions for drug activation. Acoustic pressures of 1–2 MPa at the focus with varying duty cycle (typically 50%) at 1 MHz frequency were calculated. Measurements of temperature rise were found in good agreement with our theoretical predictions. Fluorescence measurements were used to assess the release of free doxorubicin that exhibits higher fluorescence intensity than the liposomal formulation. *In vitro* experiments of drug delivery using doxorubicin-loaded TSLs and HIFU were performed with BLM and U-87 MG cancer cells that were seeded in Opticell™ chambers or suspended in polystyrene cuvettes. Trypan blue cell viability assay was used to evaluate the drug release and uptake. Ultrasound-induced TSL activation increased cell mortality considerably.

11:15

4aBA13. Miniature kilohertz-frequency ultrasound implant for brain drug delivery. John C. Foo (Dept. of Biomedical Eng., Cornell Univ., B60 Weill Hall, Ithaca, NY 14853, jcf247@cornell.edu), Susan C. Pannullo (Weill Cornell Medical College, New York, NY 10065), William L. Olbricht, Michael L. Shuler, and George K. Lewis, Jr. (Cornell Univ., Ithaca, NY 14853)

Glioblastoma multiforme is a highly aggressive brain malignancy currently treated by surgical resection, radiotherapy, and local drug delivery. However, due to heterogeneities in neural tissue and tumor cell migration into surrounding healthy tissue, treatment strategies remain ineffective and patient prognosis poor. We have shown that low intensity focused and non-focused ultrasound may safely enhance the penetration and distribution of chemotherapeutics delivered locally via convection-enhanced infusions in rodent brain parenchyma *in vivo*. In scaling the ultrasound-assisted therapy from rodent to large animal and increasing its clinical translatability, we designed a miniature kilohertz (100 and 600 kHz) ultrasound implant that can be placed within the brain tumor resection cavity during patient recovery to assist chemotherapy. The miniature ultrasound system has been tested in a diffusion-based human-sized agarose brain phantom tumor model under various acoustic intensities and pulse sequences. Results indicate that ultrasound increases the volume of tracer distribution by a factor of 2–3 compared with controls. We present the effects of acoustic intensity and pulse interval from the implant on distribution of tracers *in vitro* and *ex vivo* in the phantom model and intact equine brain, respectively. The miniature ultrasound implant provides significant drug distribution gains that may improve patient outcome.

11:30

4aBA14. Scattering and attenuation from an emulsion of echogenic liposomes. Shirshendu Paul (Mech. Eng., Univ. of Delaware, Newark, DE 19716), Tapas Nandy, Sanku Mallik (North Dakota State Univ., Fargo, ND 58108), and Kausik Sarkar (Univ. of Delaware, Newark, DE 19716)

Liposomes are nanoparticles (diameter 50 nm–10 μm) formed by a self-assembling lipid-bilayer encapsulating an aqueous phase. Due to their structure similarity to cells, they show many advantages in medical applications such as lesser toxicity, prolonged circulation, and greater uptake in target organs or tumors. They are ideal as drug delivery or image enhancing agents. Recently, echogenic liposomes (strong scatterers of ultrasound) have been developed for contrast ultrasound imaging. However, the mechanisms for the observed echogenicity—unlike contrast gas microbubbles, liposomes contain mostly an aqueous phase inside and water outside—remain uncertain. In this talk, size distribution as well as acoustic properties such as attenuation of and scattering from an emulsion of echogenic liposomes will be reported. The attenuation depends linearly on the liposomal concentration indicating absence of multiple scattering. The scattered response from the liposomal solution contains a subharmonic component much weaker than the second harmonic component. Reduction in mannitol concentration in the preparation of echogenic liposomes leads to decrease in polydispersity of the emulsion and affects echogenicity. Encapsulating the dye 5(6)-carboxyfluorescein as a model for hydrophilic drugs reduces echogenicity by 10 dB. [Work supported by NSF.]

Session 4aEA

Engineering Acoustics: Transducer Design and Fabrication (Lecture/Poster Session)

R. Daniel Costley, Chair
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Contributed Papers

8:30

4aEA1. Transducer design synthesis. William J. Marshall (Acoust. Section, Raytheon IDS, 1847 W. Main Rd., Portsmouth, RI 02871, william_j_marshall@raytheon.com)

Traditionally underwater sound projectors are designed by repeatedly exercising an electroacoustic model of a proposed device. Given properties of the environment, chosen materials, and controlling dimensions of the device, one can prepare a model that will predict its mechanical and acoustic performance. Reaching the design goals involves iteration of the model to find that particular mix of dimensions which best satisfies all requirements. The inverse of this process, a design synthesis method (DSM), makes the design process more efficient. In a DSM the environment, materials, and desired mechanical and acoustical metrics determine critical dimensions of a device optimized for meeting those requirements. Following this step regular electroacoustic modeling is used to calculate remaining aspects of the optimized design's acoustic and mechanical performance. Inverting the design equations to develop a DSM for the flextensional class of transducers would appear to be a daunting problem; however, it can be solved rather easily using the nonlinear goal-seeking algorithms available in mathematical tools such as MATLAB and MATHCAD. This talk will present a brief description of the relevant math problem and its solution.

8:45

4aEA2. Electronic enhancement of the directional response of a micromachined biomimetic optical microphone. Baris Bicen, Levent Degertekin (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405), Quang Su, Weili Cui, and Ronald N. Miles (State Univ. of New York, P.O. Box 6000, Binghamton, NY 13902-6000)

A silicon micromachined biomimetic optical microphone has been recently demonstrated to have directional response and low noise [J. Acoust. Soc. Am. **125**(4), 2013–2026 (2009)]. In this microphone, the motion of one end of the pivoting rectangular diaphragm is detected using an integrated optical interferometer and each end can be actuated using separate electrostatic actuators. This configuration enables one to selectively enhance either the directional, anti-symmetric rocking mode or the omni-directional symmetrical second vibrational mode of the diaphragm in two separate active feedback loops. A circuit model with two electrical ports to actuate each side of the diaphragm, and two acoustic ports to drive each mode has been developed to illustrate the electronic method. The model includes the effect of the air medium and the backside cavity through mechanical impedances which are verified through measurements. With the two-sided active feedback scheme, the model predicts significant improvements in the directional response resulting in more than 10-dB improvement in the residual intensity index when the microphone is used for gradient measurement in an intensity probe.

9:00

4aEA3. Investigation of a balanced-armature transducer for vibrational energy harvesting. Nikolas T. Vitt and Stephen C. Thompson (Appl. Res. Lab., Penn. State Univ., P.O. Box 30, M.S. 2430A, State College, PA 16804, ntv105@psu.edu)

The ability to sustain or prolong the operation of devices which require electrical power by utilizing ambient energy is desirable in many applications. While a variety of transducers have been considered for harvesting vibrational energy, the balanced armature transducer has yet to be fully investigated for such purposes. Balanced armature transducers, such as those used in hearing aids, are promising as they offer a compact design and are commercially available. An analog circuit model for a commercial hearing aid transducer has been modified to include a vibration input and the results are compared with experiment. Using this model, the performance of the transducer can be assessed before and after possible design modifications are considered. [This work was sponsored by the Office of Naval Research.]

9:15

4aEA4. Piezoelectric poly-g-benzyl-L-glutamate polymethyl methacrylate composite disks fabricated using molds. Yonghwan Hwang, Yub Je (Dept. of Mech. Eng., Univ. of POSTECH, Pohang, South Korea, serenius@postech.ac.kr), James E. West, Seungju M. Yu (Univ. of Johns Hopkins, Baltimore, MD 21218), and Wonkyu Moon (Univ. of POSTECH, Pohang, South Korea)

Poly-g-benzyl-L-glutamety Polymethyl methacrylate (PBLG-PMMA) composite polymer, a new piezoelectric transduction material, was invented by Yu *et al.* In this study a method is developed to fabricate a disk using detachable molding. Unlike film-type piezoelectric polymers such as the polyvinylidenefluoride (PVDF), this new material can be used to form a body of various shapes and sizes such as conventional piezoelectric ceramics. The piezoelectric characteristics mainly come from the state of alignment of polygamma-benzyl alpha, L-glutamate molecules (PBLG), which have large electric dipole moment. The PBLG can be melted with methyl methacrylate by heating, and mixture with applying electric field can be polymerized as a piezoelectric PBLG-PMMA composite in the mold. The fabricated specimens have the dimensions of 20 mm in diameter and 3 mm in height. The piezoelectric coefficient ($d33$) depends on the concentration of PBLG. Measured values were 0.45 and 1.9 pC/N with concentrations of 10 and 30 wt %, respectively. Even though these values are lower than PVDF, PBLG-PMMA could be applied as a transduction material in various shapes of acoustic transducers because the results suggest that increasing the intensity of electric field and the PBLG concentration may improve the piezoelectricity. Moreover, it is also a great potential to consider new matrix materials instead of PMMA. [Work supported by Grant No. ADD-UD080004DD.]

9:30

4aEA5. Capacitive micromachined ultrasound Doppler velocity sensor. Minchul Shin, Joshua S. Krause (Dept. of Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, minchul.shin@tufts.edu), Paul A. DeBitetto (Draper Labs., Cambridge, MA 02139), and Robert D. White (Tufts Univ., Medford, MA 02155)

An acoustic Doppler velocity measurement system using microelectromechanical systems (MEMS) capacitive micromachined ultrasound transducer (cMUT) array technology is introduced. This low power acoustic velocity measurement system operates in both transmit and receive modes. The device consists of 64 0.6 mm diameter polysilicon and gold diaphragms, and operates at approximately 180 kHz with a quality factor of 30. Computational predictions suggest that in transmit mode the system will deliver a sound pressure level (SPL) of approximately 66 dB SPL at 1 m from

the source with an 11 deg, −3 dB beamwidth when driven with a 12 V excitation. In receive mode, the predicted sensitivity is 2.2 mV/Pa at the preamplifier output. Based on experimentally determined electronic noise densities of approximately 4×10^{-16} V^2/Hz, resolution may be as good as 0 dB SPL in a 5 Hz band. This suggests that with good reflection, a range of approximately 8 m (4 m out and 4 m back) is achievable. Velocity resolution is expected to be on the order of 1 cm/s with a 5 m/s maximum measurable velocity. At this point, all predictions are computational. Fabrication will be complete in the next month and experimental characterization of system performance will be presented at the meeting.

9:45

4aEA6. Synthesize rugged acoustic sensors using poly(vinylidene fluoride)/carbon nanotube composite films. Qin Zhang (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, qzhang@olemiss.edu), Rasheed A. Adebisi, and Joseph R. Gladden (Univ. of MS, University, MS 38677)

A procedure is proposed to prepare poly(vinylidene fluoride) (PVDF) /multi-walled carbon nanotube (MWCNT) nanocomposite thin films with improved mechanical properties compared to the pure PVDF films. The PVDF/MWCNT mixture was formed using solution blending and the ultrasonic dispersion method and then spin-coated on a rotating glass substrate to produce films nearly 20 μm thick. The mechanical properties of the resulting films were examined and compared to the pure PVDF films through the mechanical tensile tests. The elastic modulus and fracture toughness after adding MWCNTs into PVDF will be discussed. These thin films are to be used to fabricate highly rugged acoustic sensors that will have the ability to withstand very high impulsive acceleration and have enough flexibility to be applied to irregular surfaces such as a helmet or clothing. The piezoelectric response of these films will also be discussed. [Work supported by the U.S. Army, contract # W15QKN-09-C-9999.]

10:00

4aEA7. Variations in standard couplers used in measurement of earphones and hearing aids. Charles B. King (Knowles Electronics, 1151 Maplewood Dr., Itasca IL 60143, charles.king@knowles.com)

In comparison to balanced armature transducers designs for ear phones and hearing aids, it is common to be have specifications that are tighter than +/−3 dB and be interested in differences between devices of less than 1 dB. Very often, measurement facilities will be scattered across the globe and thus it becomes very important to understand variations in the "standard" measurement couplers used. While couplers are standardized in IEC and ANSI and work seamlessly when looking for 5-dB variations; higher precision measurements require greater care. The user must be very aware of numerous variations possible and allowed within the standard and influenced by aspect ratios of the couplers, differences in the tuning at manufacture of the coupler, and barometric pressure at different locations. Measured data as well simulated data will be shown to help understand the range of potential errors and the frequency bands in which they occur.

10:15

4aEA8. Transduction in acoustically actuated carbon nanotube Structures. Barbara Deschamp, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts, Lowell, Elect. & Comp. Eng., 1 University Ave., Lowell, MA 01854)

A model of the process of mechanical to electrical energy transduction in a carbon nanotube (CNT) clamped between two metallic junctions and suspended over a substrate-backed gate electrode is presented. The CNT that is driven into resonant mechanical vibrations by an external acoustic field exhibits charge transport characteristics that are dependent on the transverse length scale of the CNT relative to the mean free path of the charge carriers. A circuit model that captures the ballistic transport features and Coulomb interactions as the CNT dimension decreases to the order of atomic scales is presented. The current resulting from the acoustically induced mechanical vibrations of the CNT depend not only on the time-varying capacitance between the CNT and gate electrode, but also on the probability and tunneling rate of a carrier through the junctions and into the CNT. The coupled effect of the resistance and capacitances of the tunnel junctions on the mechanical vibrations of the CNT and resulting charge transport are analyzed. [B.D. acknowledges support from GK-12 fellowship through National Science Foundation Grant No. 0841392.]

This poster paper will be on display and the author will be at the poster from 10:30 a.m. to 11:30 a.m.

4aEA9. Modeling of frequency responses for arbitrary earphone designs. Yu Du (School of Automotive Eng., Dalian Univ. of Technol., 2 Linggonglu, Ganjinzi District, Dalian, Liaoning Province 116024, China, yudu@dlut.edu.cn)

To meet a certain frequency response, the earphone design process often involves selecting receivers, style and dimensions of sound tubes, and damping used in the sound tubes. Currently, this is largely a trial-and-error procedure, especially when multiple drivers are used. Since there is lack of a reliable modeling method, the response characteristics of current earphone designs rely heavily on the experience of the designers' experiences. This study is concerned with the development of a modeling approach to aid the earphone design process. During the model development, it is assumed that the earphone response is measured inside an earcoupler (e.g., IEC 60711). Furthermore, it is also assumed that the dynamic characteristics of the selected receiver(s) and the impedance of the terminal earcoupler are known. The model predicts the response based on the selected structural features including the sound tubes and damping values between the receiver(s) and the earcoupler. Both lumped-parameter modeling and finite-element modeling methods are discussed. The predictions are then compared with experimental results. It is shown that the model results reasonably match the measurement data. It is concluded that this modeling approach can give insights before building physical prototypes, thus significantly increasing the efficiency of the earphone design process.

Session 4aED

Education in Acoustics and Diversity in Acoustics: Teaching Acoustics with Low Cost Materials and Homemade Instruments

Andrew C. H. Morrison, Cochair
DePaul Univ., Physics Dept., 2219 Kenmore Dr., Chicago, IL 60614

James M. Sabatier, Cochair
Univ. of Mississippi, National Center for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Invited Papers

8:20

4aED1. Demonstrating the effect of air temperature on wind instrument tuning. Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416-1031)

Wind instruments are known to play out of tune when the air temperature is well above or below room temperature, as may occur during outdoor performances. Because the speed of sound in air increases with rising temperature, brass and woodwind instruments tend to play sharp when the air is hot and flat when the air is cold. A simple demonstration of this effect is shown using PVC pipes. Temperature differences sufficient to make the effect audible are easily obtained using ice water and hot tap water. The demonstration is safe, inexpensive, and easy for students to perform. Extensions to the basic procedure are described, including the use of greater temperature extremes and comparison of measurements with theory. The demonstration is suitable for students of many different levels, using either a qualitative or quantitative approach.

8:40

4aED2. The hose-o-phone: A quasi-musical instrument. Thomas Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789, tmoore@rollins.edu)

A musical instrument made from rubber tubing and a plastic funnel can be used to demonstrate several important acoustic phenomena. This presentation will focus on the important physics that can be demonstrated using this simple and inexpensive “musical” instrument.

Contributed Papers

9:00

4aED3. A research-based approach to teaching the basic concepts of musical instruments with low cost instruments. Wendy K. Adams (Dept. of Phys., Univ. of Northern Colorado, CB 127, Greeley, CO 80639, wendy .adams@colorado.edu)

A week long unit covering the basic concepts of musical instruments /voice (source of vibration, resonance, and a way to change pitch) will be presented. Students often do not pick up on these basic ideas of musical instruments from a straight forward clear lecture. Physicists typically are trying to teach them why instruments work, which means focusing on what is common about all instruments. While typically their previous instruction (music classes, for example) tried to teach them how to categorize instruments focusing on what is different. It appears that this prior instruction, and the fact that sound waves are invisible make this a difficult topic for students. The unit includes the use of a homemade straw instrument, cup instrument, and reed instrument. Students compare and contrast different types of instruments in an attempt to generalize the necessary characteristics of musical instruments. They also investigate their own voices, an interactive simulation on resonance, and an interactive lecture demonstration on resonance (pasta and raisins).

9:15

4aED4. Low-cost, novel teaching tools for liberal-arts acoustics courses. Philip S. Spoor (CFIC-Q, Dr. 302 10th St., Troy, NY 12180)

Several teaching tools and methods are demonstrated, which have proven especially useful for teaching acoustics to students with a liberal-arts background. The author developed these tools over 5 years of teaching the acoustics of music and speech at the Pennsylvania State University. The class had no prerequisites, and would often include some students with a strong physics or math background, some with a strong musical background, some with both, and some with neither. The methods discussed in this presentation have been found to keep the concepts accessible yet interesting to students of widely varying backrounds. These methods include teaching the students to do Fourier analysis by ear on common sounds, therefore teaching themselves how a complex tone is comprised of simpler pieces. Another method involves making a Helmholtz resonator the centerpiece of an auditory puzzle, using ordinary (and inexpensive) objects. The puzzle is simple enough that anyone trained in acoustics gets it right away; but even many trained scientists without an acoustics background are stumped.

9:30

4aED5. A simple high school Lloyd’s mirror experiment: The difference between theory and experiment. Cleon E. Dean and Joshua A. Smith (Dept. of Phys., Georgia Southern Univ., P.O. Box 8031, Statesboro, GA 30460-8031, cdean@georgiasouthern.edu)

A baffled circular loudspeaker is driven at 10 000 Hz to provide a sound source for a high school level Lloyd’s mirror experiment. The experiment was to be performed in an ordinary building hallway. The experiment was also designed to produce interference using reflected sound from a hard floor while avoiding reflected sound from other surfaces. Modeling the sound as a Fresnel volume proved useful, but certain assumptions about the effectiveness of sound absorbing tile and flat foam rubber mats were less so. Substitution of more effective sound absorbing materials gave improved results when reflected sound was to be eliminated or the interference effect annulled.

9:45

4aED6. A homemade Edison tinfoil phonograph. Andrew R. McNeese, Richard D. Lenhart (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029), Jason D. Sagers, and Preston S. Wilson (The Univ. of Texas at Austin, Austin, TX 78712-0292)

In 1877 Thomas Edison invented the phonograph, a device capable of recording and reproducing sound. The original design used the sound induced vibrations of a stylus to etch a time-locked copy of the acoustic wave onto a rotating, foil-covered cylinder. Playback was made possible by reading the etched signal with a second stylus attached to a transmitting diaphragm. The transparency of Edison's original design makes the phonograph a useful tool to demonstrate and discuss the concepts of acoustic waves and sound-structure interaction. A short history of the invention is given, a homemade version of Edison's original phonograph is presented, and the essence of the sound-structure interaction is explained.

THURSDAY MORNING, 26 MAY 2011 — ASPEN, 8:30 TO 11:45 A.M.

Session 4aMU

Musical Acoustics: Singing and Perception in Musical Acoustics

Uwe J. Hansen, Chair
Indiana State Univ., Dept. of Physics, Terre Haute, IN 47809

Invited Papers

8:30

4aMU1. Formant strategies in professional male singers. Johan Sundberg (Dept. of Speech Music Hearing, KTH, Lindstedtsvxe4; gen 24, SE-10044 Stockholm, Sweden, pjohan@speech.kth.se), Filipa L (Univ. of Aveiro, 3810-193 Aveiro, Portugal), and Brian Gill (Steinhardt School of Culture, Education, and Human Development, New York, NY 10003)

Singers use formant tuning to avoid discontinuities in voice quality between registers. Around the passaggio, a set of spectrum characteristics has been identified as important to achieve this register equalization. Previous studies suggest different formant tuning strategies in male voices. The present investigation attempts to corroborate these suggestions experimentally. Eight professional tenor or baritone singers sang ascending scales on the vowels [a, æ, i, u] ending on pitches F4, F4, or G4 (350, 370, and 392 Hz, respectively) and including their passaggio notes. These scales were sung with two different formant strategies: (a) as in their normal classical singing technique and (b) as in a non-classical, musical theatre style. Formant frequencies were measured using the custom made DECAP inverse-filtering software (GRANQVIST). Results revealed formant tuning differences between the two styles. For the classical singing technique, systematic changes of formant frequencies with pitch were observed. For the back vowels F1 was adjusted to a frequency below the second partial and F2 to the vicinity of the third, confirming previous observations. Formant frequencies were often quite close to a partial. The major spectrum characteristics could be accurately replicated by synthesis applying the source filter theory of voice production.

9:00

4aMU2. A tessitura-gram for singers. Ingo R. Titze (Natl. Ctr. for Voice and Speech, 136 South Main St., Ste. 712, Salt Lake City, UT 84101-1623, ingo.titze@ncvs2.org)

In singing, tessitura refers to the pitch range and median pitch level of a song. Physiologically, a singer has his own tessitura. It is shown here that a voice range profile, which is a plot of intensity range or subglottal pressure range versus fundamental frequency, can be used to quantify the tessitura of a singer. Finding the best match between the singer's tessitura and the vocalist's tessitura can help performers predict their success on certain compositions that cannot be transposed or to find the ideal transposition. Concepts of vocal endurance and dynamic range are also discussed in light of the tessitura-gram.

Contributed Papers

9:30

4aMU3. The realization of rising tone contours in sung Cantonese. Murray H. Schellenberg (Dept. of Lingusitics, UBC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mhschellenberg@gmail.com)

Singers in tone languages are often thought to rely entirely on song melody to realize the tonal speech melody of the language. There is, however, some limited evidence that singers directly modify vowel $F0$ corresponding to lexical tone while singing. This paper analyzes the acoustic output of ten Cantonese singers singing a specially composed song to test whether singers adjust their performances to reflect mismatches between speech melody and song melody. It is found that singers include the dynamic portion of rising contour tones (falling tones were not included in this study) but that they do not adjust their performance to mark register components of tone.

9:45

4aMU4. Perception of high-frequency energy in singing and speech. Brian B. Monson, Andrew J. Lotto, and Brad H. Story (Dept. of Speech, Lang, and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ, bbmonson@email.arizona.edu)

Traditionally, acoustical energy above 5000 Hz in voice and speech has received little attention from the scientific community. Some recent studies have explored the role of this high-frequency energy (HFE) in qualitative percepts of speech and music. To determine what other perceptual information is extractable from this frequency range, listeners were presented with tokens containing only HFE from samples of singing and speech (i.e., singing and speech high-pass filtered at approximately 5.7 kHz). Results showed that listeners could successfully discriminate gender and mode of production (speech versus singing) given only acoustical energy above 5 kHz. Additionally, listeners successfully identified the song being sung and the words being spoken. The implication is that HFE potentially plays a more significant role in perception of singing and speech than that suggested by the current understanding. [Work supported by the NIH-NIDCD.]

10:00—10:15 Break

10:15

4aMU5. Absolute pitch is correlated with high performance on relative pitch tasks. Kevin Dooley and Diana Deutsch (Dept. of Psych., Mail Code 0109, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92037)

Absolute pitch (AP)—the ability to name a musical note without a reference note—has been shown to be associated with an advantage in pitch identification tasks; however, its usefulness in contexts involving relative pitch has so far been unclear. To explore this issue, 36 trained musicians—18 AP possessors and 18 non-possessors with equivalent age of onset of musical training and duration of musical training—were tested on several different interval naming tasks requiring only relative pitch. AP possession was highly positively correlated with performance on these tasks; $r=0.72$, p less than 0.001. Furthermore, the advantage of AP possession in performing these tasks was not dependent on musical key, interval size, or musical context. These findings support the hypothesis that AP may be beneficial in performing musical tasks including those that primarily require the processing of relative pitch.

10:30

4aMU6. Relative harmonic magnitude extraction for musical instrument detection. Sejin Oh and Jeung-Yoon Choi (Dept. of Elec. and Elecron. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, Korea)

Harmonic structure is widely used for assessing the timbre of pitched musical instruments. However, fundamental frequency and overtones are not easy to extract due to spurious partials and missing fundamentals. This study attempts to extract underlying harmonic structure based on source-filter modeling. The spectral envelope is first extracted by linear predictive analysis. To discard spurious partials, the source is divided out using noise level estimation, yielding harmonic structure characteristic to particular instruments. The resulting relative harmonic magnitudes, as well as the spectral envelope, are used to classify musical instruments. Experiments were conducted on the RWC musical instrument sound database. Results show that features are preserved for each musical instrument, over varying pitches.

10:45

4aMU7. Discrimination of instrument tones resynthesized with piecewise-linear approximated harmonic amplitude envelopes. Chung Lee, Andrew Horner (Dept. of Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong), and James W. Beauchamp (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

Harmonic amplitude envelopes of instrument tones were approximated using piecewise-linear segments to investigate how the discrimination of approximated sounds varies with the number of breakpoints and instruments. Listening tests were done to determine the discrimination of the approximated sounds from the originals. Apart from basic statistical analysis, various error measurements were evaluated based on the results of the listening tests. Relative-amplitude critical band error was the best error metric, accounting for about 82% of the discrimination variance. Strong correlations were found between the discrimination scores and spectral incoherence, with breath noise in the flute and bow noise in the violin particularly discriminable in piecewise-linear approximation.

11:00

4aMU8. Harmonic-temporal model with multiple function for sound synthesis. Toru Nakashika, Tetsuya Takiguchi, and Yasuo Ariki (Dept. of Comput. and Systems Eng., Kobe Univ., 1-1 Rokkodai-cho, Nadaku, Kobe 657-8501, Japan, nakashika@me.cs.scitec.kobe-u.ac.jp)

The most popular method for synthesizing musical timbre sound is additive synthesis, where the timbre sound is obtained by adding multiple harmonic partials with different time envelopes. Additive synthesis has many advantages such as high flexibility to control each partial individually and easy way to reconstruct musical tones from the spectrum. However, it is difficult to control the harmonic partials as a whole because time envelopes of each harmonic are independent of one another. Furthermore, the model has many parameters which are difficult to control. For example, many instruments require dozens of harmonics, where each envelope consists of about 1000 time points, and the parameters are hard to handle. In this paper, additive synthesis is extended to easily control a whole sound with only a few parameters. In this approach, all the time envelopes of each harmonic are parametrically modeled by a 2-D function, called multiple function. Two types of multiple functions are introduced with the update rules of their parameters based on an EM algorithm. The harmonic-temporal shape of an instrument sound can be directly controlled by changing the parameters of the proposed function slightly.

11:15

4aMU9. Multipitch analysis with specmurt based on the sparseness of the common harmonic structure pattern. Daiki Nishimura, Toru Nakashika, Tetsuya Takiguchi, and Yasuo Ariki (Dept. of Information Sci., Univ. of Kobe, 1-1 Rokkodai-cho, Nada-ku, Kobe 657-8501, Japan)

A novel multipitch analysis based on specmurt with the sparseness of the common harmonic structure pattern is presented. The conventional specmurt analysis is based on the idea that a fundamental frequency distribution is expressed as a deconvolution of the observed spectrum by the common harmonic structure pattern. However, it is considered impossible to obtain the strictly correct model of the common harmonic structure since harmonic structures can slightly vary depending on the pitch. Therefore a new method to analyze the fundamental frequency distribution without modeling the common harmonic structure is proposed. In this method, many possible fundamental frequencies are prepared using observed spectrum peaks, and harmonic and nonharmonic structures corresponding to them are obtained by the deconvolution of observed spectrum. Then, based on the sparseness, L1 or L2 norms are applied to only the appropriate harmonic structures in order to obtain a correct harmonic structure. The experimental result shows the effectiveness of our proposed method.

11:30

4aMU10. A vibrotactile music system based on sensory substitution. Deborah Egloff, Jonas Braasch, Philip Robinson, Doug Van Nort, and Ted Krueger (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, eglofd@rpi.edu)

The idea of the project reported here was to design a system that builds on touch to enable people with severe hearing impairments to "listen" to music through a process called sensory substitution. The goal was to transform the auditory parameter space into one that is adequate for haptic perception. The approach reported here builds on (i) the design of a haptic display, a tabletop device with 8–24 actuators that can be driven individually, (ii) machine learning algorithms, and (iii) a psychophysical study to determine which music cues can be perceived through touch. The latter was necessary because vibrotactile perception is not yet well understood in the context of music perception. The double-blind study analyzes how vibrotactile stimuli contribute to the perception, cognition, and distinction of sounds in human participants who have been trained versus those who have not. In order to ensure that normal-hearing participants could not hear sounds radiated from the haptic display, sound isolating headphones were used to playback pink noise during the experiment.

Session 4aNS

Noise: Analysis and Control of Information Technology Noise

Scott D. Sommerfeldt, Cochair
Brigham Young Univ., Dept of Physics and Astronomy, Provo, UT 84602

Kent L. Gee, Cochair
Brigham Young Univ., Dept of Physics and Astronomy, Provo, UT 84602

Invited Papers

8:00

4aNS1. Ecma International Acoustic Standard Activities: An enabler for controlling information technology noise emissions. D. Hellweg Robert, Jr. (Hellweg Acoust., 13 Pine Tree Rd., Wellesley, MA 02482), Dunens Egons (Hewlett-Packard Co., 11445 Compaq Ctr. Dr. W., Houston, TX 77070), and Eric Baugh (Intel Corp., 15400 NW Greenbrier Pkwy., Beaverton, OR 97006)

Ecma International is an organization that facilitates the timely creation of a wide range of standards for Information Technology (IT) products, including acoustics standards. Ecma standards have a history of short development time and revision. Each Ecma acoustics standard will be discussed demonstrating how it promotes the analysis and control of IT product noise. ECMA-74 is the test code for sound emissions from IT equipment. Annex C includes uniform operation and installation conditions for most types of IT equipment and is now referenced by the latest versions of ISO 7779:2010 and ANSI S12.10. This reference by other standards ensures that there is a world-wide consistency on mounting and operating conditions and that these procedures are up to date. The latest edition of ECMA-109 on noise declarations for customers presents revised guidelines to manufacturers on making declarations when testing a small sample size. ECMA-108 presents methods for determining high frequency sound power and has been submitted to ISO for revision to ISO 9295. ECMA TR/275 on measuring fan structure-borne noise was the basis of ANSI S12.11/Part2 (ISO 10302/Part 2). ECMA TR/99 covers constant sound power fan curves to assist product developers to select low-noise operation of fans at realistic operating points.

8:20

4aNS2. Measuring printer impulses using high frequency spectral content. Katy Ferguson (Hewlett-Packard Co., 11311 Chinden Blvd., Boise, ID 83714, katy.ferguson@hp.com)

Printer impulsive noise significantly contributes to the overall product sound quality. Printer manufacturers are seeking accurate impulsive metrics to understand and quantify customer perceptions and acceptability criteria. Numerous research papers [Baird, Inter-Noise 2005; Ali and Bray, NoiseCon 2004; etc.] have correlated both IT specific and general impulsive metrics to listening test results. These studies show that impulsive metrics that factor in frequency content have high correlation to user perception. This paper discusses the development and verification of a high frequency spectral content printer impulsive metric. This calculation excludes impulsive amplitude, using only the frequency content as a measure of user acceptability.

8:40

4aNS3. Overview of notebook fan noise reduction efforts. Eric Baugh (Intel Corp., MS CO5-166, 15400 NW Greenbrier Pkwy., Beaverton, OR 97006, eric.baugh@intel.com)

Notebook computers face particular challenges with regard to cooling the interior components and the surface of the chassis that is in contact with users, in addition to being operated in a wide range of thermal and acoustic ambient conditions. Fan airflow is responsible for about 85% of the cooling capability in typical designs, and is in turn limited by acoustic considerations. This paper will review fan-level and system-level efforts by Intel Corporation to promote low noise fan and system designs. Test fixturing to emulate the restricted space inside a notebook computer, combined with an ISO 10302 plenum, can be used to generate constant sound power fan curves, which capture multiple aspects of fan performance. It can be shown that the optimum design is different for the installed condition than for the free space condition where noise is typically reported today. At the system level, the chassis impedance is intimately connected to the airflow at a chosen noise limit. The location of the fan exhaust relative to the LCD panel can influence the operator position noise. Active control of fan noise has also been demonstrated in a notebook computer.

9:00

4aNS4. Multichannel algorithm for zone to zone active noise control (ANC) application. Ofira Rubin (Silentium Ltd., 2 Bergman St., Tamar Sci. Park, Rehovot 76703 Israel), Dani Cherkassky, and Ido Maron (Silentium Ltd., 2 Bergman St., Tamar Sci. Park, Rehovot 76703, Israel, ofira@silentium.com)

Multichannel acoustic active noise control (ANC) systems are increasingly being developed in order to overcome the limitations of the standard loudspeaker and to increase the spatial extents of the quiet zone. The method is to synthesize the physical characteristics of the unwanted noise field in an extended listening area. Multichannel algorithm allows controlling the sound field produced by loud-

speakers array. This paper presents a multichannel algorithm technique based on filtered-x LMS algorithm. The technique selected for this optimization problem is taking into account the effect of secondary sources coupling used for control purposes. The effectiveness of multichannel ANC algorithm using loudspeakers array and one sensor in a real system was verified. A quiet zone noise reduction of 6 dBA was demonstrated in 50-cm diameter for broadband signal.

9:20

4aNS5. Active control of centrifugal fan noise: Modeling design guidelines. J. James Esplin (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, jamesesplin@hotmail.com), John K. Boyle, Scott D. Sommerfeldt, and Kent L. Gee (Brigham Young Univ., Provo, UT 84602)

Information technology (IT) noise is very prevalent in today's society. Active noise control (ANC) has shown promise in minimizing the effect of fan-induced IT noise on users. Much of the previous research has concentrated on axial cooling fans, such as those found in desktops and servers. This approach was based on the concept of minimizing radiated acoustic power in a model of the fan radiation, and using those results to determine appropriate nearfield locations for the error sensor(s). This paper describes modifications to this previous method to develop a modeling approach to implement active noise control with a centrifugal blower, such as those found in fan trays and laptop computers. This model has been used to predict tonal noise inside and outside the duct, as well as how to best develop an ANC system for such an idealized setup. Differences between the axial fan model and the centrifugal blower model are discussed, as well as some limiting assumptions for each model.

Contributed Papers

9:40

4aNS6. Active control of centrifugal fan noise: Experimental results. John K. Boyle, J. James Esplin, Scott D. Sommerfeldt, and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, jkb321@gmail.com)

Previous work by these authors in active control of axial fans suggests an approach that can be successful in applying active control to small centrifugal fans used in fan trays and laptop computers. The modeling and analysis strategies developed for axial fans were modified for use with centrifugal fans mounted in a rectangular exhaust duct. Experimental verification allowed for proper inclusion of damping in the model. By minimizing the sound power radiated from the duct, optimal error sensor placement was predicted. Experimental results verified the effectiveness of placing the error sensor at these locations. Using predicted control source and error sensor locations, the rectangular duct was replaced by a centrifugal fan and duct attached to a heat sink, with the total dimensions being the same as the previous rectangular duct. The experimental results indicate that significant global reduction of the radiated tonal fan noise can be achieved.

9:55

4aNS7. Spatial active noise reduction with multiple error microphones. Ofira Rubin (SILENTIUM Ltd., 2 Bergman St., Tamar Sci. Park, Rehovot 76703, Israel, ofira@silentium.com) and Dani Cherkassky

Recently, research in the field of active noise control (ANC) has increased considerably. Currently there are three different approaches to extend the upper limiting frequency and attenuation zone of ANC systems. The first approach is to locate a single error microphone in close proximity to the noise source. This approach extends the upper limiting frequency but achieves local noise reduction. The second approach is to place a single error microphone at a remote point relative to the noise source. The problem with this approach is that due to coherence problems poor results are achieved. The third approach is .to place several error microphones around the noise source in various deployments. This approach, named as multiple error sensor technology (MEST), appears to be the most promising one. In this paper, three major phenomena were empirically investigated: the first is the contribution of MEST compared with a single error sensor, the second is the optimal deployment of the error microphones, and the third is the influence of the number of error microphones on the ANC performance. A significant improvement to ANC compared with classical single error microphone methods was achieved by applying MEST. Spatial broadband noise reduction of 6.4 dB(A) was demonstrated.

10:10—10:25 Break

10:25

4aNS8. Aeroacoustic beamforming for airfoil trailing edge tonal noise sources. Elias J. G. Arcondoulis, Con J. Doolan, Laura A. Brooks, and Anthony C. Zander (School of Mech. Eng., The Univ. of Adelaide, South Australia 5005, Australia, elias.arcondoulis@adelaide.edu.au)

Airfoils produce tonal and broadband noise at low to moderate Reynolds number flow conditions. The effect of variation in flow Reynolds number and airfoil angle of attack on the quantification and localization of the acoustic sources responsible for the observed tonal noise component is still uncertain. An investigation of the effect of Reynolds number, angle of attack, and acoustic frequency on the tonal noise sources will be presented. This study was performed using aeroacoustic beamforming in a small anechoic wind tunnel. Accurately locating the tonal noise source locations of an airfoil will improve knowledge of how airfoils produce tonal noise in this flow regime which will help the implementation of noise reduction techniques. In addition, some of the problems associated with beamforming low frequency noise sources and close proximity coherent tonal noise sources using conventional beamforming techniques and deconvolution algorithms will be addressed. [The authors would like to thank the School of Mechanical Engineering at the University of Adelaide for access and use of the facilities and equipment detailed in this presentation.]

10:40

4aNS9. A computational method for evaluating the installation effects of axial flow fan system noise. John S. Bibb, III (Caterpillar Inc., 5000 Womack Rd., Sanford, NC 27330)

Fan noise has become a significant source of noise in many environments. In product design the desire for low fan noise is often hindered by insufficient data in the design phase. In this work, a noise prediction scheme for low speed fans with significant inflow distortion was developed. Basic design parameters such as thrust, torque, blade shape, and blade thickness are used to predict a baseline noise signature. Flow disturbances from simple common shapes in the flow path are approximated and used to determine the noise level produced for a fan that is installed in a particular geometry. The inputs to the program are restricted to commonly available information that can be determined early in the design cycle. The process has been verified with an experimental setup in an anechoic chamber. The results of the experiment and the prediction from the program agree within 3 dB over a wide range of angles for the first two blade passing frequencies and a reasonable range of broadband noise.

10:55

4aNS10. Investigation of user complaints of sound masking delivered from underfloor air distribution grilles. Mark Rogers (Greenbusch Group, 1900 W Nickerson St., 201, Seattle, WA 98119)

A LEED-Gold certified government building includes six floors of open-plan office space that are equipped with electronic sound masking delivered from a raised floor having under-floor air distribution (UFAD). Results of user polling show mixed results for acoustic satisfaction, with significant complaints about speech privacy, and noise from the systems interfering with users' ability to work. This research focuses on this dissatisfaction, the general conflict between speech privacy ease of verbal communication, and acceptance of noise-generating treatments. Little research has been done on masking delivered via UFAD, so it was also desired to further the understanding of this design as part of a larger research program by the Federal government on Green Buildings. Specific objectives were to develop and test possible remedies for this system, and improve planning of future facilities that may have masking via UFAD. On-site measurements of masking and HVAC spectra, as well as noise reduction and observations for various conditions, were made. Post-measurement data analysis was conducted, and experimental remedies were developed and tested. The results and recommendations will be presented.

11:10

4aNS11. Bayesian parameter estimation for porous material characterization. Cameron J. Fackler and Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

This work proposes a method to estimate the material parameters of rigid-frame porous materials through measurement of the acoustic impedance of such materials. Porous materials are characterized by their physical parameters: porosity, tortuosity, and flow resistivity. For many materials, direct measurement of these parameters requires time-consuming or highly sensitive procedures. Based on some existing models for the characteristic impedance of a porous material in terms of the physical material parameters, Bayesian parameter estimation is used to estimate the physical parameters of a material from a measurement of its complex acoustic impedance. In addition to estimation of the values of the physical parameters, Bayesian analysis generates information on the uncertainties and interdependence of the parameters. The results obtained are compared to published data for several rigid-frame porous materials.

THURSDAY MORNING, 26 MAY 2011 WILLOW A, 8:00 A.M. TO 12:00 NOON

Session 4aPA

Physical Acoustics and Engineering Acoustics: Violent Cavitation Activity in Water and Other Liquids I

Lawrence A. Crum, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105

Thomas J. Matula, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105

Chair's Introduction—8:00

Invited Papers

8:05

4aPA1. Evidence for a plasma core during acoustic cavitation in sulfuric acid. Kenneth S. Suslick, Hangxun Xu (Univ. of Illinois at Urbana-Champaign, ksuslick@uiuc.edu), and David J. Flannigan (California Inst. of Technol.)

Extreme temperatures and pressures are produced through, acoustic cavitation: the formation, growth and collapse of bubbles in a liquid irradiated with high intensity ultrasound. Single bubbles have generally been assumed to give higher temperature conditions than bubble clouds, but confirmation from the single bubble sonoluminescence (SBSL) emission spectra has been problematic because SBSL typically produces featureless emission spectra that reveal little about the intra-cavity physical conditions or chemical processes. Here definitive evidence of the existence of a hot, highly energetic plasma core during SBSL is presented. From a luminescing bubble in sulfuric acid, excited state to excited state emission lines are observed both from noble gas ions ($Ar+$, $Kr+$, and $Xe+$) and from neutral atoms (Ne, Ar, Kr, and Xe). The excited states responsible for these emission lines range from 8.3 eV (for Xe) to 37.1 eV (for $Ar+$) above the ground state. Observation of emission lines allows for identification of intra-cavity species responsible for light emission; the energy levels and bandshapes of the emitters indicate that the plasma generated during cavitation is comprised of highly energetic atomic and ionic species with Ne as large as $10\times D^{21}$/cc. Ionization and plasma can also be created during multi-bubble sonoluminescence in sulfuric acid.

8:25

4aPA2. Time resolved spectra of the transition to a sonoluminescing blackbody. Seth Putterman, Brian Kappus, and Shaz Khalid (Dept. of Phys., Univ. of California, Los Angeles, CA 90095)

The passage of ultrasonic sound through a fluid leads to pulsations of an entrained bubble that are so nonlinear that acoustic energy density is concentrated by 12 orders of magnitude to generate picosecond flashes of ultraviolet light. The spectrum, for xenon bubbles, matches an 8000 K ideal Planck blackbody even though the micron radius of the bubble is small compared to the photon mean free path. To probe the formation of a blackbody we have weakened sonoluminescence by generating it with a water hammer. The implosions are

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now subsonic and the collapse densities are only 10^{21}/cc. Nevertheless, time resolved spectroscopy of these 1-s flashes indicates the formation of ideal blackbodies.

8:45

4aPA3. Acoustic parameters of sonoluminescence in alcohol aqueous solutions. Weizhong Chen, Weicheng Cui, and Suibao Qi (Key Lab. of Modern Acoust., Ministry of Education, and Inst. of Acoust., Nanjing Univ., Nanjing 210093, China, wzchen@nju.edu.cn)

The single bubble sonoluminescence (SBSL) has been investigated in alcohol aqueous solutions with different concentrations. The amplitude and frequency of the driving ultrasound for SBSL change with the alcohol concentration. That is, the best driving amplitude and frequency decrease as the concentration increases. A drop of alcohol can weaken, even quench the SBSL in pure water if we drive the bubble under the best condition of the pure water. On the other hand, if the driving ultrasound is optimized for the SBSL in alcohol aqueous solution, the decrease of the alcohol concentration will weaken the SBSL. In other words, the SBSL in pure water driven by the best ultrasound of aclohol aqueous solution will brighten as the alcohol is dissolved in. The parametric region for SBSL in pure water and alcohol aqueous solution is experimentally measured. Their difference can be explained in the theoretical framework of Rayleigh bubble dynamics.

9:05

4aPA4. Probing multibubble cavitation fields under different experimental conditions. Muthupandian Ashokkumar (School of Chemistry, Univ. of Melbourne, Victoria 3010, Australia)

The understanding and control of the spatial distribution and the type of cavitation bubbles are crucial for the efficient use of cavitation bubbles in these applications. A semi-quantitative comparison of the spatial distribution of the cavitation activity has been carried out using the images of sonoluminescence, sonophotoluminescence, and sonochemiluminescence at various ultrasound frequencies and at various acoustic amplitudes. The sonochemically active cavitation zones have been found to be much larger than the sonoluminescence cavitation zones. Also, the sonochemically active bubbles have been observed at relatively lower acoustic amplitudes than that required for sonoluminescence bubbles to appear. The acoustic power required for the observation of the initial cavitation bubbles increases with an increase in the ultrasound frequency. These observations highlight the complexities involved in acoustic cavitation. Multibubble sonoluminescence-based experimental technique has also been used for the detection and the control of type of cavitation at low and high ultrasound frequencies.

9:25

4aPA5. Where are the hot bubbles? Thomas Kurz, Philipp Koch, Daniel Schanz, and Werner Lauterborn (Third Phys. Inst., Univ. of Göttingen, Friedrich-Hund-Pl. 1, 37077 Göttingen, Germany, tkurz@dpi.physik.uni-goettingen.de)

To contribute significantly, e.g., to sonoluminescence light emission, bubbles oscillating in an ultrasonic field have to persist for some time that is large compared to the acoustic driving period. This is the case if they are stable against perturbations of shape or position, and do not dissolve or grow too quickly by rectified diffusion. For noble-gas filled bubbles in water with parameters in the corresponding parameter region called bubble habitat, the peak temperatures at collapse are calculated using a standard ODE model of bubble dynamics, assuming homogeneous adiabatic compression of the gas-vapor mixture. Results are presented in terms of phase diagrams with the bubble's rest radius, driving frequency, acoustic and static pressure, and gas concentration as control parameters. Hot bubbles are encountered in the giant response at high pressures and small bubble size. For such microbubbles, the assumption of homogeneity is tested by molecular dynamics simulations.

Contributed Paper

9:45

4aPA6. Spatially and temporally resolved sonoluminescence from compact bubble clouds. Jonathan R. Sukovich, A. Sampathkumar, R. G. Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jsukes@bu.edu), and D. Felipe Gaitan (Impulse Devices Inc., Grass Valley, CA)

Previous investigations of flash duration for single bubble sonoluminescence (SBSL) events in water have shown emission pulse widths on the order of 30–300 ps. The spatial extent of the light-emitting region in SBSL has not been successfully resolved, but it is less than 1 μm. Here we report temporal and spatial observations of light emission from laser-nucleated, compact bubble clouds at high static pressure. Employing high-speed imaging and PMT monitoring, we observe events with durations on the order of 50 ns, whose spatial extent can reach 1 mm in radius. The evolution of event size, spatial uniformity, and intensity will be monitored and compared with parametric data (maximum radius of cloud, outgoing shock velocity, static pressure, and post nucleation time) to discover correlations between light emission and hydrodynamics. [Work supported by the Impulse Devices, Inc.]

10:00—10:20 Break

Invited Papers

10:20

4aPA7. Cavitation inception and damage effects in liquid mercury exposed to a high-energy pulsed proton beam. Ronald A. Roy, Robin O. Cleveland, R. Glynn Holt, Parag V. Chitnis, Nicholas J. Manzi, Christopher E. Ormonde, Qi Wang (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215), Mark Wendel, and Bernie Riemer (Spallation Neutron Source, Bldg. 8600, MS 6473, Oak Ridge, TN 37831)

The Oak Ridge National Laboratory Spallation Neutron Source (SNS) employs a high-energy proton beam incident on a liquid mercury target to generate short bursts of neutrons. Concomitant with neutron production is the rapid heating of the mercury, resulting in the formation of acoustic shock waves and the nucleation of cavitation bubbles. The subsequent collapse of these bubbles may result in the premature erosion of the target's steel walls. We report on a multi-pronged effort to both understand and address the problem by (1) inhibiting cavitation inception using bubble injection, (2) analyze cavitation and cavitation cloud dynamics using passive noise diagnostics, and (3) mitigating vessel wall damage using surface modification techniques. [Work supported by the ORNL Spallation Neutron Source, which is managed by UT-Battelle, LLC, under Contract No. DE-AC05-00OR22725 for the U.S. Department of Energy.]

10:40

4aPA8. Single and multiple bubble shock-induced collapse: A computational study. Nicholas Hawker, Matthew Betney, and Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, yiannis.ventikos@eng.ox.ac.uk)

We present computational simulations of shock-bubble interaction, leading to violent bubble collapse. Two-dimensional, axisymmetric and 3-D simulations of single and multiple bubbles, at various arrangements, are discussed. A front-tracking methodology (FronTier) is used to model accurately the liquid-gas surface, together with shock-capturing compressible methodologies for computing the shock propagation. A wide range of shock intensities is simulated (from 1 GPa, up to 100 GPa), and pressure, density, and temperature profiles are analyzed. The results obtained reveal a particularly complex behavior, even in the single bubble case, highly non-spherical, with a multitude of shock and contact structures emerging and interacting. Well-known phenomena, such as jet formation, are reproduced and parameterized. The subnanosecond and near-nanometer resolutions of the simulations reveal spatially and temporally discrete secondary collapse events and allow for the evaluation of conditions relevant to numerous processes and practical applications. The relative importance of conduction, viscosity, and surface tension are compared and discussed. Results obtained are juxtaposed to relevant available experimental data and features observed are discussed in the light of the finer resolution the computations offer.

Contributed Papers

11:00

4aPA9. The effects of fluid compressibility on multibubble cavitation for high-intensity focused ultrasound. David Sinden, Eleanor Stride, and Nader Saffari (Dept. Mech. Eng., Univ. College London, London WC1E 7JE, United Kigdom, d.sinden@ucl.ac.uk)

In this talk the effects of liquid compressibility and viscosity on cavitation occurring during high-intensity focused ultrasound treatments are discussed. Using techniques from dynamical systems analysis, the criterion for inertial cavitation is formally generalized for multi-bubble models. It is shown that the threshold for inertial cavitation remains independent of the number of bubbles present. It is conjectured that this is because the reradiated pressure field from any one bubble only has a significant influence on the oscillations of the surrounding bubbles if it collapses inertially. For interacting bubbles the effects of compressibility are analyzed through the inclusion of the appropriate time delays in the bubble interactions. This may result in instability, but it is shown that for values of viscosity relevant to tissue, the system is stable. Instabilities may develop in the full system limiting the ability of a bubble to undergo repetitive bounded oscillations. There are four main mechanisms by which a bubble can become unstable: radial, translational, diffusive, and non-spherical instabilities. In each case it is shown that on including delayed interactions the bubbles are more likely to undergo stable oscillations. The physical implications of these findings are discussed.

11:15

4aPA10. The effect of static pressure on the inertial cavitation threshold and collapse strength. Kenneth B. Bader, Jason L. Raymong, Joel Mobley, Charles C. Church (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS, 38677, kbader@olemiss.edu), and D. Felipe Gaitan (Impulse Devices, Inc. Grass Valley, CA 95945)

The conditions within an inertially collapsing bubble are known to produce high temperatures and pressures. Previous investigators have proposed various mechanisms for driving the collapse in order to maximize the energy density of the bubble's contents. Working in fluids at high static pressures (up to 30 MPa) has shown some promising results. Preliminary data indicate that the strength of the collapse scales with the static pressure in the fluid. We report here the results of a series of rigorous experiments designed to better define this relation at the threshold of inertial cavitation. The hydrostatic pressure dependence of the inertial cavitation threshold in ultrapure water was measured in a radially symmetric standing wave field in a spherical resonator driven at 26 kHz. At this threshold, the collapse strength will be reported in terms of the resulting photon radiance below 400 nm and the relative shock wave amplitude. These experimental results will be compared to predictions of collapse strength. [Work supported by Impulse Devices, Inc.]

11:30

4aPA11. Dynamics of tandem bubble interaction in a microfluidic channel. Fang Yuan, Georgy Sankin, and Pei Zhong (Dept. of Mech. Eng. and Mat. Sci., Duke Univ., 101 Sci. Dr. Durham, NC 27708)

The dynamics of tandem bubble interaction in a microfluidic channel (800×25 μm^2, W $\times$ H) are investigated using high-speed photography with resultant fluid motion characterized by particle imaging velocimetry. Single or tandem bubble is produced reliably via laser absorption by micron-sized gold dots (6 μm in diameter with 40 μm in separation distance) coated on a glass surface of the microfluidic channel. Using two pulsed Nd:YAG lasers at $\lambda = 1064$ nm and about 10 μJ/pulse, the dynamics of tandem bubble interaction (individual maximum bubble diameter of 50 μm with a corresponding collapse time of 5.7 μs) are examined at different phase delays. In close proximity ($\gamma = 0.8$), the tandem bubble interacts strongly with each other leading to asymmetric deformation of the bubble walls and jet formation, as well as the production of two pairs of vortices in the surrounding fluid rotating in opposite directions. The direction and speed of the jet (up to 95 m/s), as well as the orientation and strength of the vortices can be varied by adjusting the phase delay.

11:45

4aPA12. Deformation and interaction of cylindrical bubbles. Yurii A. Ilinskii, Todd A. Hay, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Experiments performed by Sankin *et al.* [Phys. Rev. Lett. **105**, 078101 (2010)] demonstrate shape deformation of two coupled laser-generated bubbles confined between parallel surfaces. Due to their confinement, their motion is predominantly two-dimensional. It is known that any change in the volume of a cylindrical bubble in an unbounded incompressible fluid is prohibited because the effective mass loading is infinite. The natural frequency of a cylindrical bubble in its pure radial pulsation mode is correspondingly zero, and it becomes finite only if the surrounding fluid is either compressible or of limited extent. Because the higher-order modes of an oscillating cylindrical bubble involve no change in volume, their natural frequencies remain finite in an infinite incompressible fluid. Using the approach developed to model shape deformation of two coupled spherical bubbles [Kurihara *et al.*, J. Acoust. Soc. Am. **127**, 1760 (2010)] we obtained a set of coupled second-order ordinary differential equations in two dimensions describing the pulsation, translation, and deformation of coupled cylindrical bubbles. The presentation will focus on the deformation of a single cylindrical bubble, and the interaction and deformation of two cylindrical bubbles. [Work supported by NIH Grant Nos. DK070618 and EB011603.]

THURSDAY MORNING, 26 MAY 2011 METROPOLITAN B, 8:00 A.M. TO 12:00 NOON

Session 4aPP

Psychological and Physiological Acoustics: Masking and Spectral Processing (Poster Session)

Lori J. Leibold, Chair
Univ. of North Carolina, Div. of Speech Hearing Sciences, Chapel Hill, NC 27599-7190

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

4aPP1. Evaluating models of spectral density discrimination. Christophe N. J. Stoelinga and Robert A. Lutfi (Dept. of Communicative Disord., Univ. of Wisconsin-Madison, 1975 Willow Dr., Madison, WI 53706, stoelinga @wisc.edu)

Spectral density, defined as the number of partials comprising a sound divided by its bandwidth, has been suggested as a cue for the identification of the size and shape of sound sources [Lutfi, *Auditory Perception of Sound Sources* (Springer-Verlag, New York, 2008), p. 23]. Two models have been proposed for the discrimination of spectral density. One assumes that the cue for discrimination is a change in the number of resolved partials, NRP [Stoelinga and Lutfi, Assoc. Res. Otolaryngology **31**, 310 (2008)]. The other assumes that the cue is a change in the rate of power fluctuations in the sound, RPF [Hartmann *et al.*, J. Acoust. Soc. Am. **79**, 1915–1925 (1986)]. The two models were tested in separate experiments using a two-interval, forced-choice procedure for measuring spectral density discrimination. In the first experiment, NRP was varied from trial-to-trial, independently of spectral density, by clustering partials into unresolved groups. In the second experiment, RPF was varied from trial-to-trial by manipulating the relative phases of partials. The models were evaluated by comparing their analytic predictions for trial-by-trial responses in both experiments to obtained responses from human listeners. [Work supported by the NIDCD.]

4aPP2. Discrimination of stimulus variance. Neal F. Viemeister, Mark A. Stellmack, and Andrew J. Byrne (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN 55455)

In real-world settings, auditory signals such as speech are highly variable and differ across occurrences and sources. The question addressed is our sensitivity to such differences in global stimulus variability. In a variance discrimination task, the stimuli were sequences of five 100-ms tone pulses separated by 30 ms. The frequency of each tone was sampled independently from a probability distribution that was Gaussian on logarithmic frequency. In the non-signal interval of the 2IFC task, the sampled distribution had a mean frequency M_f and variance V_f. In the signal interval, the variance of the sequence was $V_f + \Delta V_f$. The task was to choose the interval with the larger variance; feedback was provided. To restrict the use of decision strategies, M_f was randomly chosen for each presentation ($\bar{M}_f$=2 kHz). Psychometric functions [$P(C)$ vs ΔV_f] were obtained for various values of V_f, and overall performance was poorer than that of an ideal observer. However, like the ideal, Weber's law behavior was observed: Constant ΔV_f / V_f yielded the same performance. Degrading the Ideal with a fixed frequency resolution noise and a multiplicative computational noise provides an excellent account of the real data. [Work supported by NIDCD DC00683.]

4aPP3. Relative loudness of speech feedback during speech production. Dragana Barac-Cikoja, Stephanie Karch, Melissa Kokx, and Lucas Lancaster (Gallaudet Univ., 800 Florida Ave. NE, Washington, DC 20002, dragana.barac-cikoja@gallaudet.edu)

Relative loudness of speech feedback during speech production was investigated using an adaptive, two-track, two interval forced choice procedure. The participants indicated which of the two signals, the speech feedback (listening while speaking interval) or its audio recording (listening only interval), sounded louder. Based on the subject's choice, the gain [indeabel sound pressure level (SPL) on the recording was raised or lowered on the subsequent trial. Relative loudness of the speech feedback was defined as the difference in the sound pressure level of each signal (feedback and replay), when the two were experienced as equally loud. The subject's speech was recorded with a microphone placed above his/her right ear, and both speech signals were presented via insert earphones. In the canal speech recordings were collected for later analysis. Fifteen normally hearing adults were tested. At the point of subjective equality, the SPL of the recorded and the live signals were equal for some of the subjects, while for others, the SPL of the live speech exceeded the SPL of the recorded speech by as much as 4 dB. The subjects' preferred speaking level and their acoustic reflex thresholds did not account for the observed individual differences. [Work supported by RERC-HE to Gallaudet University.]

4aPP4. Temporal effects with multiple-burst multitone masking. Eric R. Thompson, Christine R. Mason, and Gerald Kidd (Psychoacoustics Lab., Sargent College, Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, thomer@bu.edu)

Previous work has shown that the detection threshold for a tone-burst target presented simultaneously with a randomized multitone masker that is not affected by increasing the number of identical target-plus-masker bursts.

However, when the masker burst is repeated, but the target tone is presented only in alternate bursts, masked detection thresholds may be reduced significantly [Kidd *et al.*, (1994). J. Acoust. Soc. Am. **95**(6), 3475–3480]. In the present study, similar conditions were tested, but the masker-alone bursts were presented either ipsilateral or contralateral to the target-plus-masker bursts. In addition, the detectability of a single target burst was measured when the target-plus-masker burst was either preceded or followed by masker-alone bursts. As is typical with random multitone maskers, there were large threshold differences between listeners, but ipsilateral presentation of the alternate masker-alone bursts provided greater benefit than contralateral presentation. Furthermore, some listeners showed no improvement of single-burst detection thresholds with either pre- or postcursors, while others showed a sizable improvement that was similar for both pre- and postcursors. Discussion will focus on possible mechanisms underlying these temporal informational masking effects.

4aPP5. Illusory continuity and masking: Evidence for an illusory tone percept through a notched noise. Valter Ciocca and Nicholas Haywood (School of Audiol. and Speech Sci., Univ. of British Columbia, 2177 Wesbrook Mall, Vancouver, BC V6T 1Z3, Canada)

Two experiments measured the illusory continuity of a frequency glide through a noise burst. Expt. 1 used a 2I-2AFC procedure to measure detection of the (target) portion of the frequency glide that overlapped in time with the noise, as a function of noise level. The noise had a frequency notch around the frequency range of the target. The portions of the glide preceding and following the noise (flankers) could be present or absent. Performance at low- and intermediate-noise levels was poorer with present than with absent flankers (at high noise levels, performance was at chance for both conditions). This suggests that listeners either perceptually restored the missing target or that the presence of the flankers resulted in some informational masking of the target. In Exp. 2 listeners rated directly their perception of continuity of the frequency glide for the same flanker conditions of Exp. 1. When the target was absent, continuity ratings increased as noise level increased, and did not differ between intermediate- and high-noise levels. These results suggest that illusory continuity occurred when the noise level was not high enough to mask the target entirely.

4aPP6. Spatial influences on the spectral restoration of narrowband speech. Miguel Cepeda, Virginia Best, and Barbara Shinn-Cunningham (Dept. of Biomedical Eng. and Hearing Res. Ctr., Boston Univ., Boston, MA 02215)

In phonemic restoration, both perceived continuity and intelligibility of temporally interrupted speech improve with the addition of noise in the speech gaps. Perceived continuity decreases slightly when interrupted speech and "filler" noise are presented from different directions; however, a recent study found that intelligibility is not strongly influenced by spatial cues. A few studies have demonstrated spectral restoration: improvements in the intelligibility of speech that has spectral notches when band-passed filler noise is added to the spectral gaps. Here, we explored whether the spatial location speech and filler noise influence intelligibility in spectral restoration conditions. We tested intelligibility of two 3/4-wide speech bands in a variety of configurations. Control conditions included diotic presentation of a low-frequency speech band alone, a high-frequency band alone, and the two bands together. Intelligibility was poor for either band alone, but significantly better with the two bands together. When we added diotic noise to the spectral gap between the low- and high-frequency bands, intelligibility improved. Finally, when we presented the noise with an ITD so that it was perceived from off center, intelligibility was lower than when both filler noise and speech bands were diotic. Results demonstrate a spatial influence on spectral restoration.

4aPP7. Event-related potentials and perception of signals in several types of background noise. Curtis J. Billings, Keri O. Bennett, Michelle R. Molis, and Marjorie R. Leek (Natl. Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW Veterans Hospital Rd., Portland, OR 97239, curtis.billings2@va.gov)

Many factors contribute to successful perception of auditory stimuli in noise, including neural encoding in the central auditory system. Physiological measures such as event-related potentials (ERPs) can provide a view of neural encoding that may inform our understanding of listeners' abilities to perceive signals in the presence of background noise. We set out to determine the effect of stimulus type on neural encoding and perception of signals in noise. Nine young, normal-hearing adults participated. ERPs were recorded (the N1-P2 and P3), and a sentence-in-noise identification task was completed in quiet and in three noise conditions: (1) steady-state speech spectrum noise, (2) interrupted noise, and (3) four-talker babble. Stimuli were presented at a signal-to-noise ratio of -3 dB. Results demonstrated significant effects of noise type on both physiological and behavioral measures with the four-talker babble condition showing the largest decrements in ERP and behavioral measures relative to testing in quiet. In addition, significant correlations were found between ERPs and sentence-in-noise measures. Differential effects of noise type on N1 and P3 ERPs will be discussed and compared with behavioral results. Results suggest that ERPs may be a useful tool for measuring neurophysiological correlates of energetic and informational masking.

4aPP8. Speech perception in steady and amplitude modulated band-pass noise. Su-Hyun Jin (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712), Yingjiu Nie, and Peggy Nelson (Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455)

The purpose of the current study was to examine the effects of temporal and spectral interferences on sentence recognition for cochlear (CI) implant listeners. Nie and Nelson (2010) investigated vocoded speech perception in amplitude-modulated band-pass noises in order to assess young normal-hearing (NH) listeners' speech understanding through cochlear implant simulations. They reported that while the spectra of the noise bands affected vocoder speech perception, there was no significant effect of the noise AM rate on performance, indicating spectral but not temporal interference. As a follow-up, young CI listeners with various devices and processing strategies participated in the current study. IEEE sentences and white noise were divided into 16 bands, and band-pass noise maskers were formed from varying combinations of 4–6 adjacent bands. The spectra of the maskers relative to the spectra of speech were set to be one of the following: completely overlapping, partially overlapping or completely separate from the speech. Noise was sinusoidally modulated at three rates: 4, 16, and 32 Hz. It is hypothesized that sentence recognition scores for CI listeners are affected by both spectral and temporal characteristics of the noise, which is different from the performance for NH listeners reported previously.

4aPP9. Relationship between basilar membrane compression estimates and masking release. Melanie J. Gregan (Dept. of SLHS, Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, grega005@umn.edu), Andrew J. Oxenham, and Peggy B. Nelson (Univ. of Minnesota, Minneapolis, MN 55455)

Listeners with cochlear hearing loss often obtain less benefit, or masking release (MR), than do normal-hearing listeners when a steady-state masker is replaced by a temporally fluctuating one, even when overall audibility differences between the two listener groups are taken into account. One possible explanation for the reduced MR seen in these listeners is loss of the active cochlear process that results in less compression and hence less amplification for speech segments that coincide with temporal gaps in the masker. This study tests whether varying degrees of cochlear compression can predict the degree of loss of MR. Behavioral estimates of compression, using temporal masking curves, were compared with the magnitude of MR for speech and for pure tones in a group of hearing-impaired listeners and a control group of age-matched noise-masked normal-hearing listeners. Compression estimates were made at 500, 1500, and 4000 Hz. Masking period patterns in a 10-Hz square-wave-gated speech-shaped noise were measured, as was MR for band-limited (500–4000-Hz) IEEE sentences, using the same temporally gated noise. The results will be used to test for a relationship between masking release and residual cochlear compression. [Work supported by NIH grants R01DC03909 and R01DC008306.]

4aPP10. Relative importance of temporal fine structure and spectral resolution on speech masking release. Jong Ho Won, Minhyun Park, and Jay T. Rubinstein (V. M. Bloedel Hearing Res. Ctr., Univ. of Washington, Box 357923, Seattle, WA 98195-7923)

This study aimed to evaluate the relative importance of temporal fine structure (TFS) and spectral resolution on speech masking release. Speech perception abilities were measured with amplitude modulated noise and steady noise maskers in normal-hearing subjects. A vocoder processing technique was developed to systematically vary the amount of TFS delivered in stimuli: 0%, 50%, 75%, and 100% of TFS were presented. Spectral resolution was controlled by varying the number of vocoder channels: 8- and 32-channels were tested. Both TFS and spectral resolution were found to significantly contribute to speech perception in the two maskers. The addition of TFS information increased the speech masking release for 8- and 32-channel conditions, but the improvement was significantly greater for 8-channel condition. The results support the hypothesis that the effect of TFS on speech masking release will be greater when spectral resolution is limited. [Work supported by NIH Grants No. R01-DC007525 and T32-DC000033.]

4aPP11. The relationship between diotic and dichotic word recognition performance in young and older adults. Christina Roup and Katie Lamoreau (Dept. of Speech Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210)

A left-ear disadvantage for dichotic listening and a deficit in recognition of speech in a competing message have been described for older adults; however, few studies have reported performance on both measures in the same group of listeners. The purpose of the present study was to assess binaural word recognition abilities in competitive listening situations among young and older adults. Young adults (18–30 years) with normal hearing and older adults (60–90 years) with bilateral mild to severe sensorineural hearing loss participated. Word recognition performance was evaluated using NU-6 monosyllabic words presented in multitalker babble and dichotically. Psychometric functions across five signal-to-noise ratios were measured in three conditions: monaural right, monaural left, and diotic. Dichotic word recognition (DWR) was also measured in three conditions: free recall, directed-right, and directed-left. Results for the psychometric functions revealed better recognition performance in the diotic condition than for either monaural condition. Results for DWR indicated a mean right ear advantage for both the free recall and directed-right conditions, and a mean left ear advantage for the directed-left condition. Group data are consistent with a binaural advantage for speech recognition in competing listening conditions for both young and older adults. Individual differences will be discussed.

4aPP12. Interactions between hearing loss and the spectral overlap of competing sounds in masked speech intelligibility. Virginia Best, Eric R. Thompson, Christine R. Mason, and Gerald Kidd, Jr. (Dept. Speech, Lang. and Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu)

This experiment explored the ideas that energetic masking (EM) limits the benefit achievable from spatial separation in a competing speech task and that differences in EM between normal-hearing (NH) and hearing-impaired (HI) listeners can explain differences in spatial release from masking (SRM). Target sentences and similar masker sentences (or in another condition, masker noises) were filtered into four narrow spectral bands and presented simultaneously. The spectral overlap of the target and masker bands was varied from minimal (interleaved bands) to maximal (identical bands). Both NH and HI listeners showed spectral tuning, such that masking by noise or speech was reduced as spectral overlap was reduced. In fact, this signal processing often improved intelligibility relative to that seen with unfiltered speech. Moreover, thresholds in the noise and speech maskers were closely related, suggesting that EM was the main factor limiting performance across tasks. More masking was found in HI as compared to NH listeners for all spectral overlaps and both maskers, and the HI group generally had less SRM. However, variations in spectral overlap did not lead to dramatic changes in SRM or in the difference in SRM observed between groups.

4aPP13. Spectrotemporal integration in listeners with normal hearing and those with noise-induced hearing loss. Evelyn M. Hoglund, YongHee Oh, and Lawrence L. Feth (Speech and Hearing Sci. The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, hoglund.1@osu.edu)

The multiple looks model provides a reasonable explanation for temporal integration [Viemeister and Wakefield (1991)], and has been shown to hold for spectral integration as well [Grose and Hall (1997); Bacon *et al.*, (2002)]. Prior results show that spectrotemporal integration of 1–8, 10-ms tone bursts in quiet are comparable for signals that change in frequency across time [Hoglund and Feth (2009)]. In this study, signals were extended (up to 12 tones) in both time and frequency. Thresholds were measured first in quiet and then with a pink masking noise. Consistent with other studies, results for normal hearing listeners in quiet or in noise exhibit integration functions with a shallower slope than predicted by an ideal listener. Listeners with noise-induced hearing loss (NIHL) were tested using the same conditions. Differences in integration performance are discussed for each set of conditions. Early results suggest that spectrotemporal integration may be more vulnerable to NIHL than either spectral- or temporal integration alone. [Research supported by a grant from the Office of Naval Research N000140911017.]

4aPP14. Discrimination of stochastic frequency modulation as a predictor of electro-acoustic stimulation benefit. Marine Ardoint (Psychoacoustics Lab., Dept. of Speech Hearing Sci., Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287-0102), Stanley Sheft (Rush Univ. Medical Ctr., Chicago, IL 60612), Kate Helms-Tillery, Christopher A. Brown, and Sid P. Bacon (Arizona State Univ., Tempe, AZ 85287-0102)

The amount of benefit in speech intelligibility in noise can vary widely for cochlear-implant (CI) users when low-frequency acoustic stimulation is combined with electric stimulation. This variability may, in part, be explained by the listener's ability to follow fundamental frequency (f0) changes with electric stimulation only: those who benefit the least may be most adept at processing f0 in the electric region. The present study measured the ability of CI users to discriminate stochastic FM patterns when presented acoustically or electrically only. The stimuli were 200-Hz carriers randomly modulated in frequency by 5-Hz lowpass noise. A cued 2IFC procedure was used. In one of the two observation intervals, the cue pattern was repeated; in the other, the modulator was inverted so that the pattern of instantaneous frequency fluctuation mirrored that of the cue pattern at the center frequency. Discrimination thresholds were measured in terms of the FM depth needed to discriminate the deviant from the cue pattern. Additional conditions disrupted FM-to-AM conversion by adding extraneous AM to either the carrier or a noise masker. The discrimination results will be compared to the benefit in speech intelligibility obtained by CI users when presented with low-frequency acoustic cues. [Work supported by NIDCD.]

4aPP15. Perceptual mechanism for spectral-ripple discrimination in cochlear implant listeners: Behavioral and model data. Jong Ho Won, Gary L. Jones, Ward R. Drennan, Elyse M. Jameyson, and Jay T. Rubinstein (V.M. Bloedel Hearing Res. Ctr., Univ. of Washington, Box 357923, Seattle, WA 98195-7923)

The present study investigated the perceptual mechanism for spectral-ripple discrimination in cochlear implant listeners. The main goal of this study was to determine whether cochlear implant listeners use a local intensity cue or global spectral shape for spectral-ripple discrimination. The effect of channel interaction on spectral-ripple discrimination was also evaluated. Results showed that it is unlikely that cochlear implant listeners depend on a local intensity cue for robust spectral-ripple discrimination. A phenomenological model of spectral-ripple discrimination, as an ideal observer, showed that a perceptual mechanism based on discrimination of a single intensity difference cannot account for performance of cochlear implant listeners. Instead, there was a significant dependence of thresholds on spectral modulation depth and channel interaction. The evidence supports the hypothesis that spectral-ripple discrimination involves integrating information from multiple channels. [Work supported by NIH Grant Nos. R01-DC007525, P30-DC04661, F31-DC009755, F32-DC011431, and T32-DC000033 and an educational fellowship from Advanced Bionics Corporation.]

4aPP16. Level-dependent intelligibility of flat-spectrum speech. Hisaaki Tabuchi and Donal G. Sinex (Dept. of Psych., Utah State Univ., Logan, UT 84322-2810, hisaaki.tabuchi@usu.edu)

This study compared the intelligibility of speech after manipulations of amplitude- and phase-spectra similar to those reported by Kazama *et al.* [J. Acoust. Soc. Am. **127**, 1432–1439 (2010)]. Two types of stimuli were used. (1) Randomized-phase speech (RPS) had the same amplitude-spectra as the unprocessed speech, but the original phases were replaced with randomized values. (2) Flat-spectrum speech (FSS) had the same phase-spectra as the

unprocessed speech, but the original amplitudes were replaced with a constant amplitude. Each manipulation was carried out for time windows that varied from 0.25 to 1024 ms, and intelligibility was measured at 40, 55, and 70 dBA. For long window lengths (> 64–128 ms), the intelligibility for FSS was nearly perfect, but the intelligibility for RPS was lower than 90 % correct at 256–512 ms and decreased to chance performance at 1024 ms. For short window lengths (< 4–8 ms), the intelligibility for both FSS and RPS decreased. For intermediate window lengths (4–64 ms), the intelligibility for FSS was poor at 40 dBA and increased with level. Intelligibility for RPS did not vary with level. Nearly-perfect intelligibility was obtained at 4–8 and 64–128-ms time windows for both RPS and FSS. Non-linear properties are discussed. [Work supported by DC010615 to D.G.S.]

4aPP17. Native language experience influences the perception of envelope and temporal fine structure cues. Christina DeFrancisci, Jong Ho Won, and Kelly L. Tremblay (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 N.E. 42nd St., Box 354875, Seattle, WA 98105-6246)

This study examined speakers of American English and Korean languages with normal hearing to assess their ability to understand intact as well as vocoded speech. Speech perception abilities were assessed using vowels, consonants, and consonant-nucleus vowel-consonant (CNC) words belonging to the English language. Stimuli were processed using three different methods: (1) intact speech, (2) temporal envelope speech (E), or (3) temporal fine structure (TFS) speech. It was hypothesized that native speakers would perform better than non-native speakers when acoustic information was limited to envelope and TFS cues. Overall, both groups showed a significant effect of stimulus condition with performance being best for the intact condition, followed by the E- and TFS-conditions, respectively. Between groups, American English speakers outperformed the Korean speakers particularly for E, and TFS-conditions, suggesting that a person's native language experience provides an advantage when acoustic information is limited to E and TFS cues. Moreover, compared to non-native speakers, we speculate that prior language experience enables native speakers to compare E and TFS cues to an existing memory base so they can better use limited acoustic information. [Work supported by NIH NIDCD Grant No. R01-DC007705 and T32-DC000033.]

4aPP18. Remote frequency masking of a 4000 Hertz signal by a 500–1000 Hertz noise band. Andrea Hillock-Dunn, Lori Leibold, and Emily Buss (Dept. of Allied Health Sci., Univ. of North Carolina, Campus Box 7190, Chapel Hill, NC 27599, ahdunn@med.unc.edu)

Werner and Bargones (1991) showed that infants are susceptible to remote-frequency masking of a 1 kHz tonal signal by a 4–10 kHz noise band, conditions for which adults do not experience masking. Differences in frequency selectivity with age cannot account for these findings, implicating informational masking. To further substantiate this interpretation and to assay whether such an effect is specific to high-frequency maskers, the current experiment compared the effects of a 500–501 kHz noise band on detection of a 4 kHz tone in 8–11 month-olds ($n = 10$) and adults ($n = 9$). A two-stage approach was employed whereby an adaptive procedure established the signal level needed to produce detection scores of 70.7% in quiet. That level was used in fixed conditions in quiet and in noise. These data were used to compute *d*-prime and estimate thresholds. Overall, infants' thresholds were higher in noise than in quiet, whereas adults' thresholds did not change across the two conditions. Results suggest that remote-frequency masking in infants occurs for maskers above or below the signal frequency.

4aPP19. Effect of signal-temporal uncertainty during childhood: Detection of a tonal signal in a random-frequency, two-tone masker. Angela Y. Bonino, Lori J. Leibold (Dept. of Allied Health Sci., Univ. of North Carolina, Campus Box 7190, Chapel Hill, NC 27599, abonino@med.unc.edu), and Emily Buss (Univ. of North Carolina, Chapel Hill, NC 27599)

The purpose of this study was to determine if listeners benefit from a cue indicating when in time a signal occurs. In order to examine age-related differences in performance, three groups of children (5–7, 8–10, and 11–13 year-olds) and adults were tested. Thresholds were measured for a 120-ms, 1000-Hz signal presented in a continuous random-frequency, two-tone masker presented at 50 dB SPL. Each 120-ms masker burst was composed of a pair of tones, one selected from a uniform distribution 300–920 Hz and the other from 1080–3000 Hz. This masker was selected because it is expected to produce primarily informational masking. Listeners completed two temporal conditions: (1) temporally-defined, with the signal embedded in a 600-ms light cue and (2) temporally-uncertain, with no light cue. Two phases of testing were completed for each temporal condition. For phase 1, a single-interval, adaptive track procedure determined the midpoint of the psychometric function. In phase 2, signal levels around the midpoint were tested using a single-interval method of constant stimuli procedure. For both temporal conditions, listeners completed four runs of 40 trials, with an equal signal-nonsignal probability. Results indicate a substantial release from masking with reduced signal-temporal uncertainty for both children and adults.

4aPP20. Infants' responses to spectral and temporal degradations of speech signals. Laurianne Cabrera, Josiane Bertoncini (CNRS, Univ. Paris Descartes, 45 rue des Saints Peres, Paris 75006, France, laurianne.cabrera@etu.parisdescartes.fr), and Christian Lorenzi (ENS-Univ. Paris Descartes, 75005 Paris, France)

The present study assessed the ability of 6-month-old infants with normal hearing (NH) to discriminate between voiced and unvoiced stop consonants (/aba/versus/apa/) using vocoded disyllables and the head-turn preference procedure. The disyllables were processed by 4- or 32-band noise-excited vocoders in order to (i) degrade temporal fine structure (TFS) cues while preserving spectral- and temporal-envelope cues, (ii) degrade TFS and temporal-envelope cues (temporal envelopes being lowpass filtered at 16 Hz in each frequency band), and (iii) degrade TFS and spectral-envelope cues (temporal-envelope cues being preserved in four broad frequency bands). Overall, the results showed that infants are able to discriminate voicing in each experimental condition. These findings suggest that as adults, 6-month-old infants require minimal spectral information to achieve robust speech discrimination as long as slow (<16 Hz) temporal-envelope cues are available in a few spectral regions. These findings also suggest that the impoverished spectral and temporal cues delivered by current cochlear implants processors can be used by 6-month-old infants to discriminate speech sounds.

4aPP21. Responses of neurons in the inferior colliculus of the awake rabbit to iterated rippled noise and harmonic tone complexes. Brian C Flynn and Laurel H. Carney (Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, brian.flynn@rochester.edu)

The perception of pitch involves complex processing of acoustical signals, requiring the grouping of widely separated frequency components into a coherent sound object. Where and how pitch processing occurs in the auditory system is currently unknown. Fully characterizing the response of neurons in the nuclei of the ascending auditory pathway is an important first step in understanding the mechanism of pitch perception. While the responses of neurons in the auditory nerve and cochlear nucleus to pitch stimuli have previously been explored, the responses of neurons in the inferior colliculus have not been characterized. The inferior colliculus exhibits properties that make it a candidate for a pitch processing center, including converging input from nearly all lower auditory nuclei and bandpass tuning to amplitude modulation. Responses of inferior colliculus neurons to pitch stimuli are heterogeneous; however, interval histograms from some units in the inferior colliculus reveal locking to the dominant periods of complex stimuli such as iterated rippled noise and harmonic tone complexes. Characteristic responses of inferior colliculus neurons will be presented and implications for pitch processing in the inferior colliculus will be discussed. [Work supported by NIH-NIDCD RO1DC01641 and T32DC009974.]

4aPP22. Pitch perception and frequency-following response in inharmonic signals. Steven J. Aiken, Kevin LeClair, and Michael Kiefte (Human Commun. Disord., Dalhousie Univ., 5599 Fenwick St., Halifax, NS B3H 1R2, Canada)

It has been suggested that auditory-evoked frequency-following response (FFR) is related to pitch [Greenberg *et al.*, Hear. Res. **25**, 91–114 (1987)], which is usually related to fundamental frequency of complex sounds. We examined FFR to dual-tone multi-frequency (DTMF) signals used in telephone communication. As these tones have been used to produce melodies, it was expected that they would have a measurable pitch and this was also determined in a behavioral task. Twelve subjects matched perceived pitch of DTMF signals to a pure tone via method of adjustment. In addition, the FFR to the same signals was recorded from the same subjects to establish whether significant FFR energy was present at the frequency of the perceived pitch. Correlation analysis of data from the behavioral task showed wide variation among subjects, with many identifying the perceived pitch as matching the lower of the two partials. Significant FFR energy was detected at both partials, which does not clarify the relationship between perceived pitch and the evoked response. In addition, we show that perceived pitch is highly context dependent and present data from FFR to stimuli in melodic contexts.

4aPP23. Pitch perception for sequences of pulses alternating different resonance scales. Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Kyoto 610-1197, Japan, minoru.tsuzaki@kcua.ac.jp), Toshie Matui (Kwansei Gakuin Univ., Hyogo 669-1337, Japan), Chihiro Takeshima (Oberlin Univ., Tokyo 194-0294, Japan), and Toshio Irino (Wakayama Univ., Wakayama-city 640-8510, Japan)

Vowels are produced as a sequence of vocal tract impulse responses which are periodically excited by glottal pulses. Each impulse response reflects the shape and the size of the vocal tract. The size, i.e., the resonance scale, is kept almost constant in the normal speech sounds. It has been found that we can sensitively "hear" such size variations in speech and that the size can be a cue to identify the sound source, i.e., the talker. What do we perceive if impulse responses with two different resonance scales are alternatively repeated. Do we perceive an intermediate sized talker speaking the same pitch as the original? Do we perceive two speech streams of the different sized talkers with the half pitch? By answering these questions, we could clarify the auditory mechanisms for detecting size, periodicity, and resonance. The test stimuli were synthesized whose consecutive glottal pulses were alternated between two resonance scales with several degrees. Pitch matching experiments were performed where the listeners could adjust the fundamental frequency of the comparison stimuli whose each pulse had an identical scale factor. To a certain degree of the resonance scale deviation, the subjective pitch appeared to match the original pitch. If the scale deviation exceeded that limit, the pitch went down by an octave.

4aPP24. Spectral processing does not give rise to behaviorally relevant cues for "pitch" perception in mammals. William P. Shofner and Megan Chaney (Dept. of Speech & Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405)

Mechanisms giving rise to pitch reflect spectral or temporal processing is still equivocal, because sounds having strong harmonic structures also have strong temporal structures. When a harmonic tone complex is passed through a noise vocoder, the resulting sound can have a harmonic structure with large peak-to-valley ratios, but little or no temporal structure. To test the role of harmonic structure in mammals, we measured behavioral responses to vocoded tone complexes in chinchillas using a stimulus generalization paradigm. Animals discriminated a harmonic tone complex from a one-channel vocoded version of the complex. When tested with vocoded versions generated with 8–128 channels, animals generalized to the one-channel version and showed no gradient in their behavioral responses. This suggests that spectral structure was not the cue for the behavioral response. To further test this, chinchillas discriminated an iterated rippled noise from the one-channel vocoded tone complex. When tested using vocoded tone complexes having harmonic peak-to-valley ratios that were larger than or similar to the rippled noise, animals again generalized to the 1-channel version rather than the to rippled noise. The results suggest that mammalian "pitch" attributes arise through temporal, not spectral, processing mechanisms. [Work supported by NIDCD R01 DC005596.]

4aPP25. The frequency following response for dichotic pitch stimuli: No evidence for pitch encoding. Hedwig E. Gockel, Robert P. Carlyon, Anahita Mehta (MRC-Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, hedwig.gockel@mrc-cbu.cam.ac.uk), and Christopher J. Plack (Uni. of Manchester, Manchester M13 9PL, United Kingdom)

The frequency following response (FFR) was measured for five subjects during the last 350 ms of a 450-ms three-tone complex with a 244-Hz F0. All components were ramped on together in both ears; some were turned off gradually so that in the last 400 ms the harmonics presented were (i) 2+3+4 to the left ear ("mono"), (ii) 2+4 to the left and 3 to the right ("dichotic"), (iii) 2+4 to the left ("2+4"), (iv) 3 right ("3"). Stimuli mono and dichotic had the same pitch. A "vertical" montage (+=Fz, −=C7, ground=mid-forehead) was used. FFR waveforms to alternating-polarity stimuli were averaged for each polarity and added, to enhance envelope, or subtracted, to enhance temporal fine structure information. In both cases, FFR magnitude spectra for the dichotic condition were similar to the sum of the spectra for conditions 2+4 and 3, indicating that information from both ears was present, but lacked peaks at 244 Hz and other distortion products visible for condition mono. Furthermore, in both cases, autocorrelation functions for the dichotic condition were similar to those for 2+4 and dissimilar to those for mono. Thus, the FFR does not reflect similarity of pitch for dichotic and monaural presentation [Work supported by Wellcome Trust Grant No. 088263.]

4aPP26. Effects of contralateral acoustic stimulation on hearing threshold fine structure and spontaneous otoacoustic emissions. James B. Dewey, Jungmee Lee, and Sumitrajit Dhar (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, jbdewey@u.northwestern.edu)

When plotted with fine frequency resolution, pure-tone thresholds often exhibit fine structure, a regular pattern of alternating minima and maxima. Spontaneous otoacoustic emissions (SOAEs) commonly occur at frequencies near those of threshold minima, and both phenomena are suggested to depend on active and vulnerable cochlear processes. The present study utilized contralateral acoustic stimulation (CAS), which modulates these processes via the medial olivocochlear efferent system, to further explore the relationship among SOAEs, threshold fine structure, and cochlear status. In normal-hearing individuals, thresholds and SOAEs were measured in quiet and with a 60 dB SPL broadband (0.1–10 kHz) noise presented to the contralateral ear. Thresholds were obtained in 1/500th–1/100th-octave steps at frequencies both near and far from those of SOAEs. Consistent with previous reports, CAS elevated SOAE frequency and reduced SOAE level. Threshold fine structure patterns were also shifted upward in frequency, though little reduction in the presence or depth of fine structure was observed. For frequency regions far from SOAEs, CAS elevated thresholds. However, at frequencies near SOAEs, where fine structure was more pronounced, thresholds often improved with CAS. Analysis of the relationships between concurrent changes in SOAE and thresholds will be presented.

4aPP27. Modeling the antimasking effects of the olivocochlear reflex in auditory-nerve responses to tones in noise. Ananthakrishna Chintanpalli (Biomedical Eng., Purdue Univ., W. Lafayette, IN 47907), Skyler G. Jennings, Michael G. Heinz, and Elizabeth A. Strickland (Purdue Univ., West Lafayette, IN 47907, mheinz@purdue.edu)

The medial olivocochlear reflex (MOCR) has been hypothesized to provide benefit for listening in noise. The present study modeled antimasking effects in single auditory-nerve (AN) fibers [Kawase *et al.*, J. Neurophysiol. **70**, 2533–2549 (1993)]. A well-established computational model for normal-hearing and hearing-impaired AN responses [Zilany and Bruce, J. Acoust. Soc. Am. **120**, 1446–1466 (2006)] was extended by using reductions in outer-hair-cell (OHC) gain to mimic the MOCR. Tone responses in noise were examined as a function of tone and noise level and OHC gain reduction. Signal detection theory was used to predict detection and discrimination for different spontaneous-rate fibers. Model results were consistent with physiological data. Decreasing OHC gain decreased the noise response and increased maximum firing rate to the tone, thus modeling the MOCR ability to decompress AN-fiber dynamic range (particularly high-spontaneous-rate fibers). For each masker level, an optimal OHC gain reduction (i.e., maximum discrimination without increased detection threshold) was found to be physiologically realistic. Thus, OHC gain reduction improved tone-in-noise discrimination even though it produced a "hearing loss" in quiet. Combining MOCR effects with the sensorineural-hearing-loss effects already captured by this AN model will be beneficial for exploring implications of their interaction for listening in noisy situations. [Work supported by NIH Grant No. R01-DC008327.]

4aPP28. Cochlear measures of selective auditory attention. Kyle P. Walsh, Edward G. Pasanen, and Dennis McFadden (Dept. of Psych. & Ctr. for Perceptual Systems, The Univ. of Texas at Austin, 1 University Station A8000, Austin, TX 78712, kylewalsh@mail.utexas.edu)

The possibility that selective auditory attention can affect the responses of the cochlea via the medial olivocochlear (MOC) pathway was investigated in human listeners using a nonlinear version of the stimulus-frequency otoacoustic emission (SFOAE), called the nSFOAE [Walsh *et al.* (2010)]. During nSFOAE recording, listeners attended to one of two competing speech streams, each composed of seven randomly-selected spoken digits that were interleaved with the nSFOAE stimuli. The talker was female in one ear and male in the other (randomized across trials). The task of the listener was to match a subset of the digits spoken by the female talker to one of two choices presented visually on a computer screen. The nSFOAE stimulus was a tone (4.0 kHz, 200 ms) presented simultaneously to the two ears, either in quiet (tone-alone) or in noise (tone-plus-noise). When nSFOAEs were measured during periods of selective attention, the tone-plus-noise responses exhibited larger maximum magnitudes, and larger changes from the tone-alone magnitudes, compared to the responses measured during a no-attention condition, in which the competing speech streams were presented, but with no digit-matching required. The differences in magnitudes across conditions were as much as 4–5 dB. [Work supported by the NIDCD.]

4aPP29. A comparison between contralateral suppression in cochlear microphonics and distortion product otoacoustic emissions. Fadi Najem (Dept. of Commun. Sci. and Disord., Missouri State Univ., 901 South Natl. Ave., Springfield, MO 65897, fadi2000@live.missouristate.edu), Wafaa Kaf, Neil DiSarno (Missouri State Univ., Springfield, MO 65897), and John Ferraro (Univ. of Kansas, Kansas City, KS 66160)

This study compares the contralateral suppressive effect of the auditory efferent system on cochlear microphonics versus distortion-product OAEs (DPOAEs) in order to identify the potential versus the mechanical changes of the outer hair cells. This was achieved by recording the CM and DPOAEs from the right ear of 16 normal-hearing young female adults with and without broad band noise (BBN) in the left ears. DPOAEs were recorded at 0.5, 2, and 4 kHz F2 frequencies, and at 0.5 and 2 kHz tone bursts and clicks for CM using condensation and rarefaction polarity. Results revealed that both DPOAEs and CM recordings showed suppression and enhancement effects of BBN, but there were no statistically significant differences within each recording (with-without BBN) and between recordings (CM versus DPOAEs) ($P = 0.923$ at 0.5 kHz, $P = 0.858$ at 2 kHz, and $P = 0.332$ at 4 kHz or clicks). This could be related to the small effect and the lack of sensitive enough equipment. In general, larger effect was observed at 0.5 kHz in DPOAEs, but at 2 kHz in CM. Interestingly, when the stimulus polarity of the CM was changed from condensation to rarefaction at 2 kHz in the presence of BBN, six participants showed CM recordings that changed from suppression to enhancement or vice versa, suggesting a possible relationship.

4aPP30. The potential role of the medial olivocochlear reflex in the estimation of cochlear input-output functions. Madison K. Schumann and Elizabeth A. Strickland (Dept. SLHS, Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, mschuma@purdue.edu)

Research in our laboratory suggests that forward masking may partially be a result of reduced cochlear gain via the medial olivocochlear reflex (MOCR). The MOCR is not activated for approximately 20 ms following elicitor onset (in this case the masker). Growth of masking functions (GOM) and temporal masking curves (TMCs) are forward masking techniques that have been used to estimate the cochlear input-output (I-O) function. Both techniques assume that there is a linear response to a masker approximately an octave below the signal frequency. In the GOM technique, masker threshold is measured as a function of signal level, while in the TMC technique, masker threshold is measured as a function of signal delay from the masker. A short (20-ms) masker was used to estimate I-O functions using the two techniques. Gain should not be reduced in the GOM condition, but could be reduced with delay in the TMC condition. I-O functions estimated using the GOM and TMC techniques diverge at high masker levels. These points correspond to long delays in the TMC technique, where the MOCR may be playing a role. The gain reduction hypothesis and other hypotheses will be explored. [Research Supported by a Grant from NIH(NIDCD) R01 DC008327.]

4aPP31. Evaluating the effects of efferent feedback, temporal integration, and off-frequency listening on perceptual estimates of frequency selectivity. Skyler G. Jennings and Elizabeth A. Strickland (Dept. of Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47906)

A computational model was used to predict detection thresholds in two forward masking tasks used to estimate frequency selectivity. Tasks simulated involved obtaining psychophysical tuning curves and notch-noise tuning characteristics. These simulations were obtained for a series of signal levels ranging from 20–90 dB SPL and for three masking conditions. Short (20 ms) and long (120 ms) maskers were used in the first and second masking conditions, respectively. The third masking condition was identical to the first, except an additional sinusoid (called a "precursor") preceded the masker. The precursor and long masker were assumed to reduce cochlear gain during the time window in which the signal was detected. Conversely, the short masker was assumed to have no effect on cochlear gain during the presentation of the signal. Several variables related to frequency selectivity were evaluated to determine their influence on estimated filter sharpness, filter high-side/low-side slopes, and filter best frequency. These variables included (1) gain reduction via the medial olivocochlear reflex, (2) temporal integration, and (3) off-frequency listening. Model predictions are compared to psychophysical data, and the influence of the aforementioned variables is discussed.

4aPP32. Does reduction in cochlear gain explain the overshoot effect? Mark Fletcher, Jessica de Boer, and Katrin Krumbholz (MRC Inst. of Hearing Res., University Park, Nottingham NG7 2RD, United Kingdom)

Under certain conditions, detection of a masked tone is improved by a preceding sound (a precursor); this is the overshoot effect. Despite over half a century of research, its underlying mechanisms remain unknown. A popular hypothesis links overshoot to reduction in cochlear gain by the medial olivocochlear reflex. This is thought to reduce excitatory masking when the masker is at the signal frequency (within-channel effect) and suppressive masking when the masker is remote from the signal in frequency (across-channel effect). This hypothesis was examined in the first of the two experiments presented in this study. The results found no within-channel overshoot, indicating that the effect must be due to factors other than gain reduction at the signal frequency. While there was substantial across-frequency overshoot, the pattern of results was inconsistent with reduction in suppressive masking. Interpretation of results from overshoot experiments is often complicated by the possibility that the precursor itself might have a masking effect on the signal. The second experiment presented in this study was designed to overcome this problem.

4aPP33. Masking patterns of stimuli exhibiting enhanced detection for spectrally notched precursors. Yi Shen and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 184 Social Sci. Lab., Irvine, CA 92697, shen.yi@uci.edu)

A narrow-band signal is subjective to less masking from a simultaneous notched masker if it is preceded by a precursor that occupies the same spectral region as the masker. The present study investigates the effect of the precursor on the internal representations of the signal and masker by measuring the masking pattern for short-duration probes as a function of the spectrotemporal placement. In the enhancement measurements, the signal was a 0.6-octave noise centered around 1 kHz, and the masker and precursor were broadband noises with spectral notches at the signal frequency. The amount of enhancement was measured as a function of the temporal separation between the precursor and masker. For corresponding conditions, the detection thresholds of a 6-ms tonal probe were measured at several temporal locations and at frequency regions occupied by either the signal or masker. These measured masking patterns were compared against the amount of enhancement, which showed that the presence of the precursor altered the internal representations of both the signal and masker. These masking patterns provide an opportunity to generate hypotheses on the plausible mechanisms underlying the auditory enhancement phenomenon.

4aPP34. Loudness matching for enhanced sinusoids. Virginia M. Richards, Eva Maria Carreira, and Yi Shen (Dept. Cognit. Sci., Univ. of California, Irvine, 184 Social Sci. Lab, Irvine, CA 92697, v.m.richards@uci.edu)

When a target sound is masked by a notched masker, thresholds can be lower when a preview of the masker precedes the detection interval. Two experiments were run to determine whether signals "enhanced" by such precursors are also louder. Trials included three intervals. On each trial a random, multi-tone masker was chosen, and presented in each interval. For the loudness matching task, a 1000-Hz target tone was added to the second and third intervals; the level of the test tone in the second interval was adjusted to match the loudness of the standard tone in the third interval. For a standard tone 3–8 dB above threshold, the test tone was adjusted to be approximately 5 dB less intense than the standard, indicating a difference in loudness. For more intense standard levels, this difference decreased. In the second experiment, thresholds for the detection of a tone added to the second or third interval were estimated. Thresholds were approximately 10 dB lower for the second than the third interval. These results indicate that stimuli that induce enhancement (i.e., threshold difference between intervals 2 and 3) also induce increases in loudness, suggesting a role of subjective amplification of the signal for enhancement.

4aPP35. Enhancing the intelligibility of high-intensity speech: Evidence of inhibition in the lower auditory pathway. James A. Bashford, Richard M. Warren, and Peter W. Lenz (Psych. Dept., Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201)

Intelligibility of narrowband speech declines considerably at high intensities, but substantial recovery from this "rollover" occurs when flanking noise bands are added. The present study employed two types of added noise: narrowband noise matching the spectral limits of the rectangular speech band (producing within-band masking) versus broadband noise (producing within-band masking plus simultaneous enhancement by out-of-band noise components). When noise added to diotic speech in experiment 1 was interaurally uncorrelated rather than diotic, intelligibility increased 5%, regardless of noise bandwidth. Interestingly, regardless of interaural correlation, intelligibility was about 13% higher with broadband rather than narrowband noise, indicating that noise-induced recovery from rollover precedes binaural processing. In experiment 2, diotic noise was presented either continuously or gated on and off with individual sentences. Intelligibility was 6% higher with continuous noise, showing adaptation of masking, which occurred regardless of noise bandwidth. Moreover, intelligibility was about 11% higher with broadband rather than narrowband noise, regardless of gating, ruling out peripheral adaptation as a source of recovery from rollover. These and other findings to be discussed are consistent with previous suggestions that intelligibility at high intensities is preserved by inhibition of rate-saturated auditory-nerve input to secondary neurons of the cochlear nucleus. [Work supported by NIH.]

4aPP36. Accurate estimation of the temporal dynamics of bouncing events. Bruno L. Giordano (CIRMMT-Dept. of Music Res., McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A1E3, Canada, bruno.giordano@music.mcgill.ca), Valeriy Shafiro (Rush Univ. Medical Ctr., 600 S. Paulina Str., 1015 AAC, Chicago, IL 60612), and Anatoliy Kharkhurin (American Univ. of Sharjah, P.O. Box 26666, Sharjah, United Arab Emirates)

The sound of a bouncing object is rich in dynamic acoustical information: subsequent bounces are more tightly spaced in time, have lower energy, and tend to excite less strongly the high-frequency resonant modes of the bounced-upon object. Previous studies on bouncing events show that dynamic information is not used to perceive the properties of the bouncing object. We tested whether this is the case when listeners are asked to predict the dynamic behavior of a bouncing event. Stimuli were recorded by dropping one of four different balls from various heights onto a hard linoleum surface. After hearing two, three, four, or five bounces, participants pressed a button to estimate the temporal location of the next bounce. No performance feedback was given. Participants never heard the bounce whose temporal location they were estimating. Bounce-time estimates were very accurate ($r = 0.96$). Acoustic analyzes revealed a strong focus on timing information and a secondary reliance on energetic information. Spectral information appeared to have negligible effects. These findings demonstrate the extreme versatility of human listeners in using variable acoustic cues to determine the dynamic behavior of real-world objects.

4aPP37. Automated detection of alarm sounds. Robert A. Lutfi and Inseok Heo (Dept. of Communicative Disord., Auditory Behavioral Res. Lab., Univ. of Wisconsin, Madison, WI 53706)

Alarm sounds are an important class of sounds with distinctive, easily recognized features that serve as a call to action. These properties make the automated detection of alarm sounds a model problem for computational auditory scene analysis [Ellis, D. (2001). CRAC Workshop, Aalborg, DK, pp. 59–62]. We compare two approaches to alarm detection, one based on a change in overall sound level (rms) and the other a change in the power-normalized autocorrelation function (PNA). ROC curves in each case were obtained for different exemplars of four classes of alarm sounds (bells /chimes, buzzers/beepers, horns/whistles, and sirens) embedded in four noise backgrounds (cafeteria, park, traffic, and music). Detection was based on 1 s time-segments drawn at random from longer recordings of these sounds. The recordings were equated in average power. Both detection algorithms produced asymmetric ROCs with similar hit rates of 80% or more for false-alarm rates above 25%. PNA gave significantly higher hit rates than rms at false alarms rates below 25%, but did not perform as well for sirens or in the background of music. The results are considered in terms of the role periodicity may play in human detection of natural targets in natural noise backgrounds.

Session 4aSC

Speech Communication: Acoustic Analysis of Children's Speech

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Chair's Introduction—8:25

Invited Papers

8:30

4aSC1. Stop-like articulation of voiced dental fricative \ð\ in 2-year-old children: An acoustic analysis. Sherry Y. Zhao (Speech Commun. Group, Res. Lab. of Electron., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, syzhao@alum.mit.edu)

The voiced dental fricative \ð\ is frequently modified from its canonical form in American English, often becoming nasalized, lateralized, or stop-like in casual speech [Manuel (1995); Manuel and Wyrick (1999); Zhao (2010)]. Despite modifications in their production, \ð\ variants in adult speech were found to exhibit acoustic evidence suggesting a dental place of articulation, consistent with Stevens' (2002) feature-cue-based model of speech processing. However, it is unclear whether children produce similar modifications for \ð\. This study compares the stop-like variant of \ð\ produced by 2 to 3-year-old children in the Imbrie Corpus [Imbrie (2005)] with adult productions. The data indicate that stop-like \ð\ frequently occurred in utterance-initial position, sometimes occurred when preceded by obstruent consonants, but rarely occurred when preceded by sonorant consonants or vowels in the children's speech; these trends are similar to those found in adult data. Furthermore, stop-like \ð\ tokens had higher burst spectrum peak, lower normalized burst amplitude, and lower F2 at following vowel onset than \d\ tokens in the same vowel context; these acoustic differences are consistent with a dental place of articulation. The data suggest that children, like adults, may be preserving the dental articulation instead of substituting \d\ for \ð\, as is sometimes assumed.

8:55

4aSC2. Acoustic characterization of /s/ spectra of adolescents: Moving beyond moments. Christine H. Shadle, Laura L. Koenig, and Jonathan L. Preston (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu)

The fricative /s/ is generally acquired later than other speech sounds by children. Spectral moments have been used to quantify comparisons of both children's and adults' fricatives, but it is difficult to link moment values to a particular production difference: e.g., the first moment may be relatively low if place is more posterior, the noise source is weaker, or voicing persists into the /s/. This study adapts parameters originally developed for analysis of adults' fricatives to children's fricatives. Twelve children (6 boys, 6 girls), 10–15 years old with typical speech, were recorded saying 55–60 words with initial /s/, in singletons or clusters. Multitaper spectra were computed at the beginning, middle and end of each /s/, and parameters were derived from them to capture the main mid-range resonance, relative amplitudes for mid-to-low and mid-to-high ranges, and their change during /s/. Results showed that the mid-to-low-range relative amplitudes increased mid-/s/; the mid-to-high-range relative amplitude decreased; the main peak's frequency decreased in labialized contexts, all of which occur in adult /s/. One parameter indicated low-frequency spectral variability. These parameters thus capture known acoustic effects of source and filter changes occurring during /s/ and are promising both for measuring developmental changes and assessing misarticulations.

9:20

4aSC3. Developing acoustic measures to evaluate the emergence of phonological contrast. Mary E. Beckman (Dept. of Linguist., Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210-1298), Jan Edwards (Univ. of Wisconsin, Madison, WI 53706), and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN 55455)

Acoustic analysis has been used in studying phonological development since at least the 1960s when Preston and colleagues measured voice onset time (VOT) in English- and Arabic-learning toddlers. Studies applying measures such as VOT, for stop voicing, and spectral peak frequency, for fricative place, show that the emergence of robust phonemic contrast is more gradual than it seems when phonetic transcription is used alone. For example, Japanese-speaking children's short-lag productions of target /d/ labeled as "[t]" might show somewhat shorter VOT than their productions of target /t/. Studies that supplement acoustic measures with continuous labeling tasks, such as a visual analog scale with "d" and "t" as target endpoints, support the idea that not only is this developmental phenomenon of "covert contrast" ubiquitous, the associated subphonemic variation can be perceived by adult listeners given the appropriate task. Moreover, listeners with different kinds of experience with young children have different sensitivities to this acoustic variation.

This finding suggests that appropriate acoustic measures can be developed to provide psychometrically valid measures over the full course of development from initial stages of producing undifferentiated or inappropriate cues to producing the adult community pattern for the contrast. [Work supported by NIDCD grant 02932 and NSF grants BSC-0729306, BSC-0729140, and BCS-0729277.]

9:45

4aSC4. Syllable organization in children's early American English: Acoustic evidence. Jill C. Thorson (Dept. of Cognit., Linguistic Psychol. Sci., Brown Univ., Box 1821, 229 Waterman St., Providence, RI, 02912, Jill_Thorson@brown.edu) and Katherine Demuth (Macquarie Univ., Sydney, NSW 2109, Australia)

Children's early CVC target words are often produced with epenthesis or heavily aspirated release (e.g., cat [*kæt*^{h}]). This has led some to suggest that these are actually produced as two-syllables ($CV.C^{(V)}$) [Goad and Brannen (2003)]. To evaluate this claim, this study conducted an acoustic investigation of syllable timing in American English 2-year-olds' production of CVC words. These were compared with productions of disyllabic CV.CV words. For adults, V_1 duration and C_2-closure duration were expected to be longer in CVC than in CV.CV words [Lehiste (1972); Lisker (1972)]. However, if durations were found to be similar across conditions for children, this might support the claim that children syllabify CVC words as two syllables (CV.C(V)). Participants were three 2-year-olds (mean=2;4) and three adults (mean=23). The stimuli were 4 prerecorded nonce words (/bɑk/, /bɑg/, /bɑkə/, /bɑgə/). Speech productions were sampled at 44 kHz and four repetitions of each word were acoustically analyzed. As predicted, adults showed significantly longer V_1 and C_2-closure durations in CVC compared to the CV.CV targets ($p < 0.01$). Children showed more variable, non-significant trends ($p = 0.06$). This suggests that 2-year-olds are not treating CVC forms like disyllables, nor are they adult-like in their timing relations. [Work supported by NIH R01HD057606.]

10:10—10:20 Break

10:20

4aSC5. The utility of content-free acoustic analyzes for assessing prosodic disorder in children. Melissa A. Redford (Dept. of Linguist., Univ. of Oregon, Eugene, OR 97403, redford@uoregon.edu) and Jolynn Cornell-Fabiano (César E. Chávez Elementary School, Eugene, OR 97402)

Global rhythm metrics have advanced our understanding of perceptually salient language rhythms and have proved useful for understanding perceptions of foreign accent. The metrics have also helped further the development of automated acoustic analyzes of speech, which are being applied to the clinical problems of early and differential diagnoses of speech and language disorder. This talk will consider the utility of such metrics for the assessment and treatment of children with disordered prosody. Results from an investigation of acoustic factors that contribute to the perception of disordered prosody in 24 school-age children suggest that global, content-free metrics of rhythm and intonation may be relatively poor predictors of experienced listeners' perceptions of disorder in individual children though the metrics correlate with listeners' differentiation of children with typical and atypical language development and of children with different eligibilities for speech and language services. Content-dependent acoustic analyzes more accurately account for the variability in judgments across children within particular groups. These results will be used to advocate for research that is aimed at acoustically characterizing prosodic disorder in individual children and at discovering the perceptual weightings and social significance of the acoustic correlates of disordered prosody. [Work supported by NICHD/NIH.]

Contributed Papers

10:45

4aSC6. Automatic acoustic-phonetic analyzes of thousands of hours of conversational speech in hard of hearing children. Mark VanDam (Ctr. for Childhood Deafness, Boys Town Natl. Res. Hosp., 555 N. 30th St., Omaha, NE 68131, mark.vandam@boystown.org)

The current longitudinal project examines speech and language development in 40 preschool children with mild- to severe-hearing loss (and a cohort of typically developing peers) as part of large, multicenter project on outcomes of children with hearing loss. Whole-day recordings are collected once monthly for 1 year and subjected to detailed acoustic-phonetic analyzes. To date, we have collected several hundred whole-day recordings using a commercially available, body-worn recorder and software (the LENA Foundation). The software performs unsupervised, offline analysis of the audio and produces (1) a time-aligned, XML-coded output including the talker labels adult-female, adult-male, target-child, and other-child and (2) the PCM wave file of the entire day. Audio segments of specific talkers are then extracted with custom software designed for detailed acoustic-phonetic analyzes of speech. For example, fundamental frequency was extracted for target-child utterances and compared in varied conversational exchanges with different interlocutors. This technology is suited to examine large-scale natural speech and langauge corpora. Developmental changes can be examined in domains ranging from prosody to conversational use of speech. Details of this research project, including advantages and challenges, will be discussed as well as theoretical implications for phonology and phonetics of child speech development. [Work supported by: NIH/NIDCD 5R01-DC009560-01S1.]

11:00

4aSC7. Emerging lexical representations in early infancy. Francisco Lacerda (Dept. of Linguist., BabyLab, Phonet. Lab, Stockholm Univ., SE-106 91 Stockholm, Sweden, frasse@ling.su.se)

This presentation proposes the view that typical early infant-directed speech (IDS) in the infant's ecological setting provides enough multisensory information to trigger the emergence of lexical representations, a theoretical view inspired by analyzes of the care-givers' speech toward infants at different developmental stages. It will be argued that the adult's speaking style in early IDS within the context of the infants' immediate ecological setting provides enough correlated sensory information to derive potentially relevant lexical items from the stream of speech sounds and objects to which the infant is exposed. The proposed theoretical model of early language development [ecological theory of language acquisition (ETLA)] is supported both by experimental results indicating that 40 Swedish infants, in the age ranges 8–10 and 14–16 months, could infer object names from looking at 1-min video materials where objects were shown and described by natural utterances produced in languages unknown to the infants, and by mathematical simulations of how early lexical items can be derived from typical infant-directed speech handled by the infant's general-purpose multisensory (audio-visual) representation capabilities. [Work supported by grant from The Bank of Sweden Tercentenary Foundation (Grant No. K2003:0867, MILLE), EU-NEST (Project No. 5010, CONTACT), Knut and Alice Wallenberg Foundation (Grant No. KAW2005.0115), and Stockholm University.

11:15

4aSC8. Alveolar and velar stop releases during speech development. Richard S. McGowan, Margaret Denny, Michel T-T. Jackson (CReSS LLC, 1 Seaborn Pl., Lexington, MA 02420), and Susan Nittrouer (The Ohio State Univ., Columbus, OH)

It has been shown that children learning English tend to use tongue-to-palate contact patterns while producing velar and alveolar stop consonants that are not as well differentiated as for adults [Cheng *et al.*, J. Speech, Language, Hearing Res. **50**, 375–392 (2007)]. The electropalatographic experiments suggest that young children have not developed the fine motor control necessary for mature articulation of lingual consonants. It is possible that this may persist to become a speech disorder as they mature. We offer a complementary perspective that focuses on the mismatch between the acoustic and aerodynamic scaling from children's vocal tracts to adult vocal tracts. In learning the motor control necessary to make a good distinction between alveolar and velar stops may be hindered by scaling mismatches. In this paper we will highlight acoustic measures of four children recorded every 6 months from the ages of 12–48 months playing with a care giver. We will consider both the formant frequencies at release and the shape of the burst spectrum to both help characterize the development acoustically and to infer what is causing the acoustics in the articulation. [Work supported by Grant No. NIDCD-RO1-001247.]

THURSDAY MORNING, 26 MAY 2011 METROPOLITAN A, 7:40 A.M. TO 12:25 P.M.

Session 4aSP

Signal Processing in Acoustics, Animal Bioacoustics, and Underwater Acoustics: Tracking Using Least-Mean Squares (LMS) Recursive Estimators, Autoregressive (AR) Models, or Related Methods

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Invited Papers

7:40

4aSP1. Model-based tracking. Edmund J. Sullivan (Prometheus Inc., 21 Arnold Ave., Newport, 02840)

Tracking of an acoustic source in the ocean with a towed array is usually done with a Kalman filter. Although the Kalman filter is a model-based processor, in the case of tracking in the ocean the model is usually the kinematic state of the target itself, i.e., its position and speed. However, this approach is burdened by the necessity of maneuvering the towship in order to obtain two independent bearing measurements. In the method presented here, the signal itself is modeled, including the wavefront curvature and the speed of the towed array. By this means, an increase in the gain and the apparent aperture of the array is obtained, leading to an improvement in the bearing resolution and the ability to obtain an estimate of the range of the source. Experimental results for bearing estimation and wavefront curvature ranging are shown. A framework for the joint estimation of range and bearing using an unscented Kalman filter is then presented in which no towship maneuver is required.

8:00

4aSP2. Modal tracking in a shallow ocean: An adaptive particle filtering approach. James V Candy (Lawrence Livermore Natl. Lab., P.O. Box 808, L-151, Livermore, CA 94551)

It is well-known that the shallow ocean is an ever changing, uncertain environment dominated by temperature fluctuations and ambient noise. The need for a processor that can adapt to these environmental changes while simultaneously tracking the evolution of modal functions is necessary for localization, inversion, and enhancement. An approach to this problem is made possible by developing a sequential Bayesian processor capable of providing a joint solution to both the modal function tracking (estimation) and environmental adaptivity problem. The posterior distribution required will contain multiple peaks and space-varying statistics requiring a sequential (nonstationary) Bayesian approach. The final design that evolves is a so-called particle filter. The particle filter is a sequential Markov chain Monte Carlo processor capable of providing reasonable performance for a multi-mode (probability distribution), space-varying problem. The tracking results are applied to synthesized pressure-field data from the actual experimental parameters. The tracking results are compared to that of the modern unscented Kalman tracker solution.

8:20

4aSP3. Tracking noise in noise. Leon Cohen (Dept. of Phys., City Univ. of New York, 695 Park. Ave., New York, NY 1056) and Lorenzo Galleani (Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy)

Very often one wants to track one type of noise embedded in another noise. In addition, the noises may be nonstationary. For example, suppose one wants to track in time and or space a reverberation type noise that is embedded in an background ambient noise. Many of the techniques that have been developed for deterministic signals do not work for such situations. We discuss various methods that have been developed to track noises and to separate noises and to estimate the parameters of non-stationary noises. We will show that a particular effect method to understand nonstationary noises is to study them in phase space. We give a number of examples.

4a THU. AM

8:40

4aSP4. Two-stage diver tracking in a high-clutter harbor environment. Geoffrey S. Edelson, Dianne E. Egnor, Philip J. Haney (BAE Systems, P.O. Box 868, Nashua, NH 03061-0868, geoffrey.s.edelson@baesystems.com), Patrick Edson, and Peter J. Stein (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063)

One approach to detecting swimmers with active sonar is to deploy a larger number of relatively simple nodes and network them together. Instead of using a complex multi-element phased array with electronic beamforming, this system uses air-backed parabolic reflectors, each with an omni-directional transducer. To avoid performance degrading acoustic interactions, the available operating frequency band is managed at the channel level. The physical configuration of this sonar system presents challenges for tracking through and across beams, and between nodes. Swimmers can exhibit low target strength and the acoustic clutter fields can be dense and highly dynamic. To detect and track swimmers in this environment, we employ a windowed Hough-transform (HT) tracker at the beam level. The HT has received wide use for track initiation. However, because of the processing gain required to continually track a weak target in such a significant clutter field, the HT is used in this case to maintain as well as to initiate tracks. The single- and multi-node tracker is a Kalman-based, multi-object tracker capable of tracking any number of stationary, constant-velocity and/or maneuvering targets. The multi-node tracker receives fused observations as inputs and creates accurate estimates based on the target position, velocity, and acceleration.

9:00

4aSP5. Machine learning approach to passive sonar localization. Brett E. Bissinger and R. Lee Culver (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

A machine learning-based depth classifier for passive sonar is under development. The classifier is based on a support vector machine (SVM) paired with backward feature selection through margin-based feature elimination (MFE). The overcomplete feature space consists of features that have been shown to work in other passive sonar classifiers, features that have potential based on efforts in similar fields, and features based on consideration of the physics of the problem. Examples include central moments, autocorrelation coefficients, and modal amplitudes. The MFE algorithm can be used to reduce the dimensionality of the feature space, ranking the features according to their utility in the classification task. The classifier is applied to a horizontal array with a source at endfire. The most powerful features are obtained using an approach similar to matched-mode processing, except that the horizontal structure of the modes is utilized. [Work supported by Office of Naval Research Grant No. 321US.]

9:20

4aSP6. Tracking in ocean acoustics for geoacoustic inversion. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ 07102)

Tracking acoustic features, namely, multipath arrivals and modal frequencies arriving at a specific time, is performed with the goal of improving geoacoustic inversion in ocean environments. Sequential Bayesian filtering results obtained with particle filters are compared to conventional maximum likelihood estimates in terms of the accuracy of the environmental and geometry parameter estimates that they produce. Previous sequential tracking methods are improved to provide higher resolution in estimation. Probability density functions are calculated for uncertainty characterization of both multipath arrival times or modal frequencies (depending on the circumstances of the measurements) and environmental parameter estimates. [Work supported by ONR.]

9:40

4aSP7. Bayesian tracking of multiple sources in an uncertain ocean environment. Stan E. Dosso and Michael J. Wilmut (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

This paper describes a Bayesian approach to the problem of simultaneous tracking of multiple acoustic sources in a shallow-water environment in which water-column and seabed properties are not well known. The Bayesian formulation is based on treating the environmental parameters, noise statistics, and locations and complex strengths (amplitudes and phases) of multiple sources as unknown random variables constrained by acoustic data and prior information. Markov-chain Monte Carlo methods are applied to numerically sample the posterior probability density to integrate over unknown environmental parameters in a principal-component space. Closed-form maximum-likelihood expressions for source strengths and noise variance at each frequency allow these parameters to be sampled implicitly, substantially reducing the dimensionality of the inversion. The result is a set of time-ordered joint marginal probability distributions for the range and depth of each source, from which optimal track estimates and uncertainties are obtained.

10:00—10:15 Break

10:15

4aSP8. Geoacoustic inversion in the Shallow Water 2006 shelfbreak region using sequential Bayesian filters. Caglar Yardim, Peter Gerstoft, and William S. Hodgkiss (Marine Physical Lab., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238)

Environmental parameters can have a large spatial variability in shelfbreak regions. In this paper, sequential Bayesian filtering, or particle filtering (PF), is used for tracking this variation. The PFs can operate in nonstationary dynamic ocean environments that affect acoustic data via complex, nonlinear equations and non-Gaussian probability density functions. PFs are used to process the data obtained during the Shallow Water 2006 (SW06) Experiment conducted on the New Jersey Continental Shelf. The PFs are first used to track the source and the stable environment with little spatial variation just northwest of the shelfbreak. The PFs track the environment using a smaller number of particles relative to conventional geoacoustic inversion methods that use successive independent inversions. Then the range-dependent shelfbreak region is analyzed and the PF methods are compared to conventional geoacoustic inversion techniques. [Work supported by the ONR.]

10:35

4aSP9. A "tracker within a tracker": Tracking multiple marine mammals. Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawaii, 2540 Dole St., Honolulu, HI 96816)

In some multiple marine mammal datasets, it is possible to separate animals by tracking slowly varying call features such as peak frequency and amplitude. Separating animals in this way can be a useful step toward tracking the spatial position of the animals. This talk discusses this "tracker within a tracker" approach to multiple-animal datasets. Details of both trackers and examples from application to real datasets will be presented. [Work supported by ONR and NSF].

Contributed Papers

10:55

4aSP10. Underwater navigation/tracking using a single transponder. T. C. Yang, Jeff Schindall (Naval Res. Lab., Washington, DC 20375), and W.-B. Yang (Nat. Inst. Standards Tech., Gaithersburg, MD 20899)

Underwater vehicle navigation and/or tracking are commonly done using several transponders forming either a long or a short base line. Transponders are deployed in advance and vehicle position is then determined by triangulation using acoustic travel time between the vehicle and transponders. Multiple transponders are expensive and inconvenient due to time required for deployment and surveying their positions. This paper presents methods to navigate/track a vehicle using only one transponder which can be dropped with its position known from GPS. One method to navigate is to estimate the range and bearing using travel time and beamforming if the vehicle is equipped with several receivers. The other method uses Doppler sensitive waveforms, such as *m*-sequence signals. Bearing is estimated using Doppler shift assuming that the vehicle knows its speed relative to the ground, and range is estimated by travel time as usual. Experimental data are presented to illustrate both methods. Errors are minimized using an extended Kalman filter. [Work supported by the Office of Naval Research.]

11:10

4aSP11. Tracking tomographic arrivals. Matthew Dzieciuch (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0225)

Ocean acoustic tomography records an acoustic pattern of peaks that is interpreted in terms of ray arrivals. Before peaks can be tracked in the time-evolving environment, they must be associated with the correct ray path. Scattering by range-dependent ocean structure is a complicating factor. Fine scale structure limits the time and bandwidth coherence of an arrival and thus a single peak per ray path becomes multiple fading peaks. Associating the peak data with the ray model is often done by hand, but this has two drawbacks. First, a human is required to look at the data precluding processing the data *in situ*. Second, there is no guarantee that the analysis is repeatable. To address these concerns, three popular methods from radar data processing, the nearest neighbor standard filter, the probabalistic data association filter, and Viterbi data association, are proposed to solve the automatic data association problem. The peak arrival time, angle, and amplitude are clues used to construct a scoring function based on the likelihood ratio. A Kalman filter is used to restrict the expected arrival pattern to shifts and stretching between transmissions. A comparison of these methods with actual tomographic data will be presented.

11:25

4aSP12. Sequential trans-dimensional Monte Carlo methods for range-dependent geoacoustic inversion. Jan Dettmer, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada), and Charles W. Holland (The Penn State Univ., State College, PA)

This paper develops a sequential trans-dimensional Monte Carlo algorithm for geoacoustic inversion in a strongly range-dependent environment. The particle-filter algorithm applies advanced Markov chain Monte Carlo methods in combination with importance sampling techniques to carry out geoacoustic inversions for consecutive data sets acquired along a track. Environmental changes along the track, such as changes to the number of sediment layers, are accounted for with trans-dimensional partition modeling which intrinsically determines the amount of structure supported by the data information content. Abrupt environmental change between consecutive data sets of high information content (peaked likelihood) are addressed by bridging distributions implemented using annealed importance sampling. Such distributions provide an efficient method to locate high-likelihood regions for new data which are distant and/or disjoint from previous high-likelihood regions. The algorithm is applied to reflection-coefficient data along a track, as can be collected using towed arrays. A simulated environment varies rapidly along the track, with changes in the number of layers, layer thicknesses, and geoacoustic parameters within layers. In addition, the seabed contains a geologic fault where all layers are offset abruptly and an erosional channel. Abrupt changes in noise level between consecutive data sets are also considered. [Work supported by ONR.]

11:40

4aSP13. Dependence of Bayesian passive sonar localization performance on signal-to-ratio. Andrew T. Pyzdek, Alex W. Sell, Brett E. Bissinger, and R. Lee Culver (Penn State Univ., P.O. Box 30, State College, PA 16804, atp5120@psu.edu)

Low frequency acoustic signals propagating in shallow water are strongly affected by interference between multiple paths resulting from boundary interactions. These interactions cause an interference pattern in the transmission loss (TL), which Jemmott [(2010)] successfully used to localize a moving source in range and depth. Jemmott's Bayesian localization algorithm employs Monte Carlo simulations to build a probability density function (pdf) model for TL based on uncertainty in environmental parameters such as water column depth, sound speed profile, and bathymetry. The TL pdf models are incorporated into the recursive histogram filter as prior pdfs and used to process received signal amplitudes and generate a posterior pdf representing the likelihood of the source location. The localization algorithm has been shown to be robust to known uncertainty in environmental parameters, but other sources of uncertainty such as ambient noise have not been included in the work to date. This paper examines how performance of the algorithm depends on signal-to-noise ratio when the noise is not included in the prior distributions.

11:55

4aSP14. Bayesian channel estimation for *K*-distributed fading models. Alison B. Laferriere (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, alaferriere@whoi.edu) and James C. Preisig (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Current underwater acoustic channel estimation techniques generally apply linear MMSE estimation. This approach is optimal in a mean square error sense under the assumption that the impulse response fluctuations are well characterized by Gaussian statistics, leading to a Rayleigh distributed envelope. However, field data suggest that the envelope statistics of the channel are better modeled by the *K*-distribution. MAP and MMSE estimators for a scalar channel tap have been derived assuming *K*-distributed fading models. The MAP estimator for a vector of *K*-distributed channel taps has also been derived. The implementation of these estimators on simulated data demonstrates an improvement in performance over linear MMSE estimation. [Work supported by the ONR Grant No. N00014-05-10085 and NSF Grant No. OCE-0519903.]

12:10

4aSP15. A multiscale sparse processing approach to tracking the delay-Doppler spread in shallow water acoustic communications. Ananya Sen Gupta and James Preisig (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., 98 Water St., Woods Hole, MA 02543)

Sparse sensing approaches have been applied to track the delay-Doppler spread in shallow water acoustic communications with varying degrees of success. Tracking the complex-valued delay-Doppler spread function coefficients is challenging due to the ill-conditioned, nonanalytic, and time-varying nature of the channel estimation problem. Since the coefficients typically follow a sparse distribution, fast adaptive least squared approaches fall short of the performance goals. We recently proposed an adaptive sparseoptimization technique, which efficiently tracks the time-varying coefficients by imposing a smooth nonconvex cost function over the $L1$ norm of the sparse coefficients and the $L2$ norm of the estimation error. However, a key limitation of this approach is the assumption that the sparsity level of the distribution stays fixed and is uniform across the different Doppler frequencies. Due to uncertainty principles governing time and frequency resolution of nonstationary processes, using uniform averaging window length across the entire Doppler spread also leads to nonuniform resolution of the estimates across the different frequency ranges. We present a multiscale algorithm where we employ variable averaging window length as well as variable sparsity factor to track the delay-Doppler spread across different ranges of frequencies over experimental field data.

THURSDAY MORNING, 26 MAY 2011 — GRAND BALLROOM D, 8:40 TO 11:45 A.M.

Session 4aUW

Underwater Acoustics and Acoustical Oceanography: Propagation, Modeling, and Inversion III

Brian T. Hefner, Chair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Chair's Introduction—8:40

Contributed Papers

8:45

4aUW1. A random matrix approach to long range acoustic propagation in the ocean. Katherine C. Hegewisch and Steven Tomsovic (Dept. of Phys. and Astronomy, Wash. State Univ., Pullman, WA 99164-2814, khegewisch @wsu.edu)

Low frequency sound propagates in the ocean within a wave guide, formed by the confining effects of temperature, salinity, and pressure on the sound speed. Within the wave guide, sound scatters due to range dependent sound speed oscillations from internal waves. These weak perturbations in the sound speed serve to randomize the acoustic signal so that the structure formed by the time series of arrivals at the receivers (i.e., the timefront) contains only minimal average information about the propagation through the action of a generalized central limit theorem. This study characterizes this information by the parameters of a statistical ensemble model for the propagation. The propagation is described by the evolution of a unitary propagation matrix, whose elements are the complex probability amplitudes for modal transitions during the propagation. The ensemble model is constructed from a product of unitary random matrices utilizing complex Gaussian random variables with minimal information about the propagation to 50 km stored in a matrix of variances and a vector of mean phases. A comparison of the properties of the average intensity timefront resulting from this ensemble model is made with those from the simulated propagations for several ranges. [This work was supported by ONR and NSF.]

9:00

4aUW2. A method for numerical representation of arbitrary boundaries in acoustic wavefront propagation. Sheri Martinelli (Dept. Torpedo Systems, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841)

An enhancement to the application of a wavefront propagation algorithm for underwater acoustics based on the level set method is presented. The presence of discontinuities in the phase space at reflecting boundaries requires the use of specialized differencing techniques to prevent oscillations in the computed solutions of the acoustic phase. A weighted essentially non-oscillatory method is applied for this purpose, necessitating the use of uniformly spaced grids over the computational domain. In order to represent arbitrary boundaries in this implementation, one has to either modify the grid to include the boundary (reduces the convergence rate) or develop an appropriate boundary condition to apply within the region of interest. Earlier versions of this work approximated the actual boundary location by the location of the nearest grid point. This resulted in a stair-step effect in the solutions, which resolves with finer grid resolution. However, the phase space is high-dimensional; a highly resolved grid can be impractical. In this work, a new method is proposed and validated which applies a model based on ray interactions with the boundary to improve results on sparser grids. [Work supported by ONR 333 and by the Science, Mathematics, & Research for Transformation (SMART) Program.]

9:15

4aUW3. Stochastic basis expansions applied to acoustic propagation in an uncertain, range, and depth-dependent, multi-layered waveguide. Jaison Novick and Steven Finette (Acoust. Div. Naval Res. Lab., 4555 Overlook Ave. S.W., Washington, DC 20375, jaison.novick.ctr@nrl.navy.mil)

We study the utility of stochastic basis expansions in acoustic propagation through a multi-layered ocean waveguide in the presence of environmental uncertainty. Environmental uncertainty means that the parameters that describe the waveguide are treated probabilistically. Specifically, in the differential equation governing propagation, the uncertainty appears in the sound speed profile as an explicit dependence on a set of random variables. This implies that the acoustic field itself is a random field. Stochastic basis expansions are attractive because of their often exponential convergence. We use a complete set of multivariate orthogonal polynomials to compute the acoustic field's statistics. The field propagates by a wide-angle parabolic equation through a rectangular waveguide comprised of three layers separated by two horizontal interfaces. A pressure release surface and hard bottom bound the waveguide. The water's sound speed is a constant perturbed by a small, random range, and depth-dependent term that models a frontal zone, the middle sedimentary layer is stratified and modeled to represent uncertainty in sound speed measurements, and the bottom layer is a deterministic attenuating layer. We compare the field's statistics generated by the stochastic basis to those generated by Monte Carlo simulations.

9:30

4aUW4. Vertical intensity structure in a shallow water waveguide. David R. Dall'Osto, Peter H. Dahl (Dept. of Mech. Eng. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), and Jee Woong Choi (Dept. of Environ. Marine Sci. Hanyang Univ., Ansan, Korea)

Acoustic intensity in an ocean waveguide is described by the local pressure and particle velocity, both of which can be described as a sum of modes. Analysis of the interaction between these modal components gives insight into the formation of characteristic intensity structures, such as interference patterns. Observations of the modal structure of the pressure and vertical velocity in a shallow water waveguide are presented using experimental data from an experiment off Korean coastal waters, the transverse acoustic variability experiment (TAVEX) that took place in August of 2008 17 km northeast of the Ieodo weather station, in waters 62 m deep. Mode filtering is performed on a 16 element vertical array that spans the water column (3 m spacing) for broadband (imploding light bulb) sources detonated at ranges from 200 to 1000 m at 40 m depth. The vertical velocity field, determined through the finite-difference approximation, and the pressure field at 230 Hz are represented by six propagating modes and their corresponding modal amplitudes. The interaction of these modal components is analyzed and PE simulated data are presented for comparison. Nondimensional indices are formulated relating the modal components of vector intensity and their utility as field indicators will be discussed. [Research supported by the U.S. Office of Naval Research and Agency for Defense Development, Korea.]

9:45

4aUW5. Striation-based beamforming using the waveguide invariant for passive sonar. Daniel Rouseff (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105) and Lisa M. Zurk (Portland State Univ., Portland, OR 97201)

Consider a horizontal array measuring the coherent field produced by a distant acoustic source. If the resulting acoustic intensity were mapped as a function of position along the array and frequency, the image would exhibit striations, bands of high intensity. It is well known that the slope of the observed striations can be related to bearing of the source and the ratio between the so-called waveguide invariant and the range to the source. The canonical value of the waveguide invariant in shallow water is 1, but deviations by 30% or more are not uncommon when there is a sharp thermocline. By beamforming the coherent field conventionally at a single frequency, one can determine the bearing of the source and hence the invariant-to-range ratio. In the present work, striation-based beamforming is developed. Rather than pick a single frequency and then beamform, each point on the array is first evaluated at a slightly different frequency with the frequency offset determined by the slope of the observed intensity striations. It is shown that the striation-based beamformer can be used to produce an estimate for the waveguide invariant that is independent of the source range. Simulation results are presented. [Work supported by the ONR.]

10:00—10:15 Break

10:15

4aUW6. A dedispersion transform: Applications development. Ning Wang, Yuling Yao, Dazhi Gao, and Haozhong Wang (Dept. of Phys., Ocean Univ. of China, 238 Song Ling RD., Qingdao, China)

Shallow water sound propagation is of typically (mode) dispersive, in particular, significant for low frequency and long-range propagation. Recently, a novel transform we called to dedispersion transform (DDT) was introduced recently by authors [Ning Wang, 9th Western Pacific Acoustic Conference Beijing (2009)]. It is a bi-parameter Fourier-like unitary transform, which can remove multimode dispersion of acoustic signals in shallow water waveguide, the only parameter required known a prior is an approximate beta-value. Some preliminary results on real-data processing has shown that the DDT works very well. In the present presentation, we would like to present some applications and relevant developments on the DDT. (1) A review of dedispersion transform: motivations, some formula and two real-data applications (source depth estimation and optimization of beamforming processing for transient signals). (2) An improved Wigner–Vills distribution suitable for dispersive multimode signals. It may be useful for dealing with various signal correlations in the time-frequency domain. (3) A theoretical exploration on the possibility for applications of DDT to mode-coupling and reverberation problems. (4) The final topic is a method using single mode dispersion to make low-frequency source ranging without referring any prior environmental information of sea bottom, real-data processing is also shown.

10:30

4aUW7. Convergence of polynomial chaos expansion based estimates of acoustic field and array beam response probability density functions. Thomas J. Hayward and Roger M. Oba (Naval Res. Lab., Washington, DC 20375)

The polynomial chaos expansion (PCE) method has been applied recently to the computation of acoustic field uncertainties arising from uncertainties of the acoustic propagation environment [Finette, J. Acoust. Soc. Am. **120**(5)]. The present work investigates, through computational studies, the properties of acoustic field probability density functions (PDFs) and the rate and manner of convergence of PCE based approximations to those PDFs. The studies assume a stratified ocean waveguide, with uncertainties of sound speed and attenuation represented by joint PDFs. Numerically accurate computations of the field and beam response PDFs are first computed from the parameter PDFs using the normal-mode expansion and the change in variables' theorem of probability theory. The resulting field and beam power PDFs exhibit irregular functional behavior and singularities associated with features of the mapping from the parameter space to the field or beam power space. The singularities have implications for the choice of suitable metrics for assessing the approximation of the field PDFs by PDFs derived from truncated polynomial chaos expansions. Function-space metrics based on existing theory of convergence of orthogonal polynomial expansions are investigated. Computational examples illustrate some qualitative aspects of the PDF approximations. [Work supported by the Office of Naval Research.]

10:45

4aUW8. Observations of mode scattering from axial and off-axial sources in the North Eastern Pacific, Long Range Ocean Acoustic Propagation Experiment. Tarun K. Chandrayadula, John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943), Peter F. Worcester, Matthew A. Dzieciuch (UCSD, La Jolla, CA 92093), James A. Mercer (Univ. of Washington, Seattle, WA 98195), Bruce M. Howe (Univ. of Hawaii at Manoa, Honolulu, HI 96822), and Rex K. Andrew (Univ. of Washington, Seattle, WA 98195)

Experimental observations of the axial modes across range are crucial to verify and complement existing theories for internal-wave induced mode scattering. The 2004 Long Range Ocean Acoustic Propagation Experiment (LOAPEX) was conducted with the main objective of analyzing internal-wave induced mode fluctuations at ranges of 50–3200 km. During LOAPEX, a 75 Hz broadband source made transmissions at an off-axial depth of 350 m and an axial-depth of 800 m. The transmissions were received across a 40 element vertical array capable of resolving the first ten acoustic modes. The mode arrivals from the two source depths significantly differ from each other. For the axial source, the observed mode arrivals are strongly influenced by the low modes directly excited at the source or by nearest neighbor coupling. The shallow source in contrast excites the low modes at a much lower energy, and thus the axial arrivals from the shallow source transmissions are due to scattering from the high modes. This talk discusses the LOAPEX mode statistics for the two source depths. The observed mode statistics are compared with predictions from theory and simulations. [Work sponsored by a National Research Council Research Associateship Award at Naval Postgraduate School.]

11:00

4aUW9. Analysis of the vertical directionality of ambient noise: SPICE04 experiment measurements and model simulations. Mehdi Farrokhrooz, Kathleen E. Wage, Matthew A. Dzieciuch, and Peter F. Worcester (Ocean Acoust. Signal Processing Group, George Mason Univ., 4400 University Dr. MSN 1G5, Fairfax, VA 22030-4444, mfarrokh@gmu.edu)

The SPICE04 experiment provided a unique opportunity to observe the ambient noise in the North Pacific [Worcester *et al.*, J. Acoust. Soc. Am. **120**(5), p. 3020]. Receptions on two 700-m segments of a deep vertical line array (DVLA) facilitate measurements of the vertical directionality of the noise above and below the surface conjugate depth. Several authors have proposed models for deep ocean noise, e.g., the spatial harmonic model developed by Cox [J. Acoust. Soc. Am. **54**(5), pp. 1289–1301] and the normal mode model developed by Kuperman and Ingenito [J. Acoust. Soc. Am. **67**(6), 1988–1996] and extended by Perkins *et al.* [J. Acoust. Soc. Am. **93**(2), pp. 739–752]. The simulated data provided by these models can be used to assess the effects of array tilt on estimates of vertical directionality. In this talk, we first analyze the impact of tilt for the DVLA using the array element positions measured during the experiment. Then we compare the SPICE04 noise estimates for the two 700-m array segments and the relevant model output. [Work sponsored by ONR.]

11:15

4aUW10. Experimental determination of the three-dimensional primary field of a seismic airgun array. Arslan M. Tashmukhambetov (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, atashmuk@uno.edu), George E. Ioup, Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA), Natalia A. Sidorovskaia (Univ. of Louisiana, Lafayette, LA), Joal J. Newcomb (Naval Oceanograph. Office, Stennis Space Ctr., MS), James M. Stephens, Grayson H. Rayborn (Univ. of Southern Mississippi, Hattiesburg, MS), and Phil Summerfield (ExxonMobil Corp., UIT/E&G DM/DS/Geodetics & Cartography, 233 Benmar, Houston, TX)

The Littoral Acoustic Demonstration Center (LADC) recorded acoustic and related data on three moored arrays and ship-deployed hydrophones, which together spanned the full water column to measure the 3-D acoustic field of a seismic airgun array, in September 2007. The seismic source vessel shot specified lines to give detailed solid angle and range information concerning the field of the primary arrival. The data were collected in the western Gulf of Mexico between the East Break and Alamos Canyon regions. Peak pressures, sound exposure levels, total shot energy spectra, one-third octave band analyzes, and source directivity studies are used to characterize the field. Three-dimensional maps of these quantities are generated to show dependence on emission and azimuthal angles and range. Longer lines were shot for propagation analyzes. Source signatures from NUCLEUS and GUNDALF are used in propagation modeling. [Research supported by the Joint Industry Programme through the International Association of Oil and Gas Producers.]

11:30

4aUW11. Target detection in a shallow water environment using a two-dimensional array. John Gebbie, Martin Siderius (Dept. of ECE, NEAR-Lab, Portland State Univ., P.O. Box 751, Portland, OR 97207, jgebbie@ece.pdx.edu), and John S. Allen (Univ. of Hawaii-Manoa, Honolulu, HI 96822)

The detection and tracking of targets in cluttered, shallow water locations have become increasingly important for both maritime security and coastal environmental monitoring. Ambient noise contributions from snapping shrimp, as well as recreational and commercial ship traffic, can limit target detection performance. Few studies have investigated the possibility of enhanced passive acoustic detection in these locations using multiple arrays. Two synchronized 24 element arrays with 300 Hz–50 kHz bandwidth were deployed at the Kilo Nalu Nearshore Reef Observatory (University of Hawaii, Honolulu, HI) in an "L" shape configuration with a vertical array within the 12 m water column and horizontal array orthogonally aligned on the seafloor. This subtropical site is located between Honolulu Harbor and a recreational boating and diving area. Broadband snapping shrimp noise, which dominates the background levels, has a diurnal variation. A series of experiments examined the feasibility of passive acoustic monitoring with array beamforming of a variety of targets including small craft, open circuit scuba divers, and a REMUS autonomous underwater vehicle. The relative performance of narrowband versus broadband beamforming techniques for detection and tracking was quantified for these targets. Moreover, the applicability of nearfield techniques, such as wavefront curvature, was also investigated. [Work supported by the DHS.]

Session 4pAAa

Architectural Acoustics, Noise, and Committee on Standards: Soundscape: Standardization and Implementation II

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square, Vernon, CT 06066

Gary Siebein, Cochair
Univ. of Florida, Dept. of Architecture, Gainesville, FL 32611-5702

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Soundscape walking tour. Dennis A. Paoletti (Shen Milsom & Wilke LLC, 33 New Montgomery St., San Francisco, CA 94105, dpaoletti@smwllc.com)

As part of the American Institute of Architects, San Francisco Chapter's Architecture Festival, Dennis Paoletti led a walking tour through downtown, experiencing the City's sights and sounds. Participants learned about environmental noise parameters and the importance of criteria for indoor and exterior spaces. They learned how sound is evaluated in the acoustical community. A brief introduction to the basics of acoustics was presented at the SMW office before the group toured downtown. Real life environmental noise conditions were experienced including where and how to achieve a level of calm and quiet within the hectic din of the City's soundscape environment. Stories about architectural acoustics were illustrated from the many local buildings that Dennis had consulted on. Additionally, the group occasionally stepped into a building where Dennis illustrated some practical examples of the acoustical parameters and criteria previously discussed. Paoletti was interviewed on public radio and asked to repeat the Soundscape Walking Tour for the City Guides, a group of volunteers who lead tours throughout San Francisco. One of the representative comments received: "Thank you for leading the Soundscape Walking Tour; all of us "heard" a version of the City we had never encountered before."

1:25

4pAAa2. The diversity and uncertainty of the sonic experience in the classroom. Sangbong Shin (School of Architecture, Univ. of Florida, P.O. B 115702, Gainesville, FL 32611, archisangbong@gmail.com)

It is critical to fully understand acoustical environments in classrooms because voice communication has been the dominant pedagogy for teachers in teaching students. Even though several standards have been applied to measure and understand the acoustical environment of core learning spaces, these are not adequate to examine multiple settings, identify the sources and receivers and their dynamic relationships, and to comprehend the nature of the specific acoustic events that occur in educational spaces since it is almost impossible to explain the diversity and uncertainty of the sonic experience in the classroom. In order to examine the variability in quantitative acoustical parameters in the spaces, an experiment was conducted with diverse configurations of measurement setups including different source types and diverse positions, different receiver types and diverse positions, diverse student conditions, and different measurement methods. To establish protocols for measurements in terms of soundscape of classrooms, analyzes were performed with impulse response signals in dynamic situations that yield specific results to reveal unique relationships between space, sound, and meaning. The results show that signal number average measurement used in the standards cannot measure the perception of sound for students as well as the physical characteristics of the soundscape in the classrooms.

1:45

4pAAa3. Soundscape of a worship space. Sang Bum Park (School of Architecture, Univ. of Florida, Gainesville, FL 32611, sbpark04@gmail.com)

Worship spaces have a variety of sonic experiences, which they must accommodate such as the service of word and the service of music. Depending on the size and architectural properties of the church, speech or music may need to be electronically reinforced or amplified by sound system. Soundscape methods can be used to evaluate each of the multiple sources present in the room at multiple locations and receiver locations. Natural acoustic and reinforced sounds from the choir, organ, musical accompaniments, pastor or priest, key readers, and congregation were considered. Three churches were chosen for this study with regard to the level of reverberance and of the use of a sound system. Analysis of data required soundscape methods showed that while reverberation time was not very different at each receiver location, early decay time (EDT), center time (Tc), and speech transmission index (STI) obtained from the natural sound propagation in the rooms were different from the measurements made using the sound system because of the location, directivity, frequency response, and coverage of the loudspeakers. In conclusion, worship spaces have to be evaluated for both natural acoustic projection and reinforced or amplified projection of sounds using multiple source and receiver locations.

2:05

4pAAa4. Soundscape of band room: Listening criteria. Lucky Tsaih (School of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611, akustx@ufl.edu)

Three band rehearsal videos have been analyzed to investigate the specific issues that conductors address or work on during a band rehearsal. These specific issues were also identified by the conductors and music students as the listening criteria during the rehearsal through interviews with conductors and questionnaires administered to music students. The analysis of video recordings of three band rehearsals shows that the conductor addressed to issues related to intonation (17%), rhythm (27%), dynamics (18%), articulation (19%), and other musical issues such as style, phrasing, tone quality, breath release time, and/or nonmusical issues such as student's discipline (21%) of the time. Questionnaires given to student musicians and faculty instructors yielded similar results with 73%–94% of students and 50%–83% of instructors naming these as primary issues dealt with in college and high school music rehearsals. This raises questions that these issues should form the basis for acoustical design criteria for this building type.

THURSDAY AFTERNOON, 26 MAY 2011 GRAND BALLROOM B, 2:40 TO 4:45 P.M.

Session 4pAAb

Architectural Acoustics and Noise: Developments in Plumbing Noise Control

David L. Adams, Chair
D.L. Adams Associates, Inc., 1701 Boulder St., Denver, CO 80211

Chair's Introduction—2:40

Invited Papers

2:45

4pAAb1. A case history of office building plumbing noise, its causes and reduction. James E. Robinson (Coffeen Fricke and Assoc., Inc., 14827 W. 95th St., Lenexa, KS 66215, jrobinson@cfaconsulting.com)

Short cuts in commercial construction often lead to acoustical problems after occupancy. In this example, a restroom was located immediately adjacent to the private office of a top executive in a hospital office building. Plumbing noise and vibration were being transmitted through the common partition's structure and through airborne paths. Short comings in the partition design and construction are discussed and several noise paths are identified. Recommended solutions are outlined and after implementation the client was pleased with the result.

3:05

4pAAb2. A subjective evaluation of the American Society of Heating, Refrigerating and Air-Conditioning Engineers plumbing noise guidelines. Michael R. Yantis (Sparling, 720 Olive Way, Ste. 1400, Seattle, WA 98101)

An evaluation of the ASHRAE plumbing noise guidelines was made by measuring plumbing noise and asking a small, trained jury to evaluate the noise subjectively. Measurements were made of supply and waste noise in a residential building where the plumbing system was isolated from direct contact with the structure and another residential building where no isolation was present. Background noise levels were also documented. In addition to listening to and evaluating the plumbing noise in the sample buildings, the plumbing noise was recorded during the measurements and played back to jury members in various room types and in the midst of various background noise levels. In addition to evaluating the acceptability of the noise versus the A-weighted noise level, the frequency characteristics of isolated and non-isolated plumbing noise were evaluated in an attempt to determine if A-weighted criteria are sufficient to identify acceptable plumbing noise levels.

3:25

4pAAb3. Plumbing noise control strategies in wood structures. John LoVerde and Wayland Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

There are a number of strategies that have been developed to isolate plumbing piping from structures. Generally effective methods of plumbing noise control include the installation of cast iron waste/vent/storm drain piping, and installing pre-manufactured or custom isolating clamps and brackets on all piping connection points [LoVerde and Dong, J. Acoust. Soc. Am. **118**, 1856 (2005)]. The level of isolation provided by these methods is relatively well understood and the methods are selected based on the level of plumbing noise reduction desired. However, there are many projects where the capability of the tradesman, constructability, and/or budget allow for little or no vibration isolation. For these projects, the authors have developed alternative strategies including space planning, plumbing routing, materials choices, and isolation at limited locations, which have been successful in controlling plumbing noise. Examples are given of successful acoustical designs with these limitations.

3:45

4pAAb4. An experience reducing toilet flushing noise reaching adjacent offices. Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, asaseattle@sacnc.com)

An office building was experiencing loud toilet flushing noise in offices adjacent to the toilets. The toilets were a 1.2 gallon per flush model. A study by others documented some maximum A-weighted slow sound levels greater than 60 dB with most greater than 50 dB and recommended a limit of 40 dB. The building owner made changes including an improved wall on one of the seven affected floors and asked the author to repeat the measurements. Results were mostly in the range of 40 to 50. Sound levels within the rest rooms were greater than 80 dB, and within the toilet stall of the loudest toilet—96 dB. The sound peaked in the 2000 Hz octave. A wall on one floor was modified by adding a layer of damped gypsum with successful results. The remaining walls were then modified similarly. The resulting average level of four flushes was less than 40 dB in all offices though some individual flushes were slightly greater than 40 dB. It was evident that structural flanking was controlling the remaining sound reaching the offices.

4:05

4pAAb5. Plumbing specifications for low noise, where are the codes and standards? Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Sq., Vernon, CT 06066, bbrooks@brooksacoustics.com)

Effective materials and installation techniques for low noise plumbing designs have been known for many years. However, with the advent of value engineering (VE), many of the traditional materials and installation methods have disappeared from all but a few projects. For example, what happened to the cast iron and copper that we all know and love? Also, separate, well insulated plumbing chases are now a rare luxury. These factors increase the challenges to develop acceptable plumbing designs from a noise perspective. Further, what do the applicable codes and standards say about plumbing noise specifications? The implications that these aspects of the design process have on plumbing noise performance are discussed.

4:25

4pAAb6. Pump noise control in high rise condominiums. Jerry G. Lilly (JGL Acoustics, Inc., 5266 NW Village Park Dr., Issaquah, WA 98027)

Noise generated by small pumps circulating water and refrigerants in multifamily residential buildings can be significant enough to generate complaints from residents if proper precautions are not taken. This presentation highlights several multifamily residential projects where pump and piping system noise was a significant source of noise complaints, and discusses what steps were taken to resolve the situation. General guidelines that can be used to prevent these problems in future projects will also be presented.

THURSDAY AFTERNOON, 26 MAY 2011 ISSAQUAH, 1:30 TO 5:40 P.M.

Session 4pAB

Animal Bioacoustics and Acoustical Oceanography: Killer Whale Acoustics

David K. Mellinger, Chair
Oregon State Univ., 2030 S.E. Marine Science Dr., Newport, OR 97365

Invited Papers

1:30

4pAB1. The role of acoustics in defining killer whale populations and societies in the Northeastern Pacific Ocean. John K. B. Ford (Pacific Biological Station, Fisheries and Oceans Canada, 3190 Hammond Bay Rd., Nanaimo, BC V9T6N7, Canada), Harald Yurk (Vancouver Aquarium, Vancouver, BC V6G3E2, Canada), and Volker B. Deecke (Univ. of St. Andrews, Fife KY16 8LB, United Kingdom)

Stable, culturally inherited repertoires of discrete pulsed calls are characteristic of the acoustic behavior of killer whales. Call repertoires may have important roles in the evolution of social segregation and reproductive isolation of sympatric killer whale populations. Here we present the results of analyzes of recordings collected from killer whale populations in coastal waters of the Northeastern Pacific from the Aleutian Islands to the Gulf of California over the past 30 years. At least three acoustically, genetically, and ecologically distinct lineages of killer whales, known as residents, transients, and offshores, inhabit these waters. Call repertoires within these lineages can further distinguish populations, communities, or smaller social groups, depending on social structure and patterns of dispersal. Salmon-feeding residents live permanently in their natal matrilines and have group-specific dialects that encode maternal genealogy. Mammal-feeding transient groups have less stable societies and tend not to have group-specific dialects, though there are regional call differences among subpopulations. Offshore killer whales, which range widely along the continental shelf and may specialize on sharks, have distinct call repertoires that appear to vary among groups. Killer whale calls can provide important insights into the structure of populations at a scale that cannot be resolved through genetic studies.

4p THU. PM

1:50

4pAB2. Studying killer whale predation in the field: A sound approach to detecting kills. Volker B. Deecke (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, Fife, KY16 8LB Scotland, United Kingdom, volker.deecke@st-andrews.ac.uk)

Killer whales are top predators in many marine ecosystems and play an important role in regulating marine mammal populations. Studies investigating the feeding ecology of mammal-eating killer whales have been compromised by our inability to consistently identify kills using visual observation. This study aimed to identify predation events using acoustic rather than visual cues. Recordings were made during focal follows of transient killer whales in British Columbia and Alaska using towable hydrophones. During ten follows, kills were confirmed using traditional methods and these recordings were scanned for echolocation clicks, whistles, pulsed calls, and characteristic sounds apparently generated during prey handling. All acoustic analyses were done blind to the behavior context. A discriminant function analysis showed that kills could be identified from the acoustic record with high certainty, primarily because rates of all sound types increased after a kill. Prey-handling sounds proved to be the best indicator that an attack was successful. An analysis of structural parameters of prey-handling sounds and estimate of their source level is also presented. These results show that acoustic cues can improve our ability to detect feeding events. Acoustic monitoring should therefore be incorporated into any field study aiming to quantify predation by mammal-eating killer whales.

2:10

4pAB3. Investigating acoustics, behavior and vessel noise exposure in endangered killer whales (*Orcinus orca*) using digital acoustic recording tags. Marla M. Holt, M. Bradley Hanson, Candice K. Emmons (Marine Mammal Ecology Team, NOAA Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. East, Seattle, WA 98112, marla.holt@noaa.gov), Robin W. Baird, Jeff Hogan, Jeff Foster (Cascadia Res. Collective, Olympia, WA 98501), Deborah Giles (Univ. of California, Davis, CA 95616), and Kenneth C. Balcomb (Ctr. for Whale Res., Friday Harbor, WA 98250)

Southern resident killer whales (SRKWs) are a fish-eating, endangered population that frequents the inland waters of Washington and British Columbia. Several risk factors have been identified that could hinder population recovery, including prey quantity and/or quality and disturbance by vessel presence and/or noise. There is a well-developed whale watching industry in the area, with an average of about 20 vessels viewing SRKWs during summer daylight hours. In some studies, killer whales decreased foraging behavior with vessel presence but details about noise exposure and prey capture events were lacking. Other research has characterized how vessel traffic increased background noise levels in SRKW habitat but data on how such exposure varies in a diving killer whale were also lacking. Our current research involves using suction cup attached digital acoustic recording tags (DTAGs) equipped with hydrophones and dive sensors on SRKWs to better characterize their use of sound, particularly during foraging, and to quantify vessel noise received at the whale. Data on vessel type and behavior and whale behavior, as well as prey samples, are also collected concurrently. This paper will describe the experimental approach taken and some preliminary results assessing risk factors potentially affecting the recovery of SRKWs.

2:30

4pAB4. Detecting killer whale whistles and squeals. David K. Mellinger (NOAA/PMEL, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, david.mellinger@oregonstate.edu)

An algorithm has been devised for detection of tonal sounds, including whistles of odontocetes and moans and other sounds of baleen whales [Mellinger *et al.*, J. Acoust. Soc. Am. (in press)]. Here we test for detection of whistles, squeals, burst pulses, and other tonal and tonal-like sounds of killer whales (*Orcinus orca*). The method is controlled using a large number of parameters, and these parameters must be chosen to as to detect killer whale sounds and not sounds of other cetaceans. The cosmopolitan range of killer whales means that they are sympatric with nearly every other species of cetacean, and thus detection must handle these other species. Here detectors are configured to detect orca tonal sounds and to avoid detecting sounds of sympatric cetaceans, such as pilot whales (*Globicephala* spp.), that produce whistles and other tonal vocalizations that overlap the frequency range of killer whales.

2:50

4pAB5. Masking of southern resident killer whale signals by commercial ship noise. Scott R. Veirs (Beam Reach Marine Sci. and Sustainability School, 7044 17th Ave. NE, Seattle, WA 98115, scott@beamreach.org) and Val R. Veirs (Colorado College, Friday Harbor, WA 98250)

The endangered southern resident killer whales (SRKWs) emit sound to communicate with each other and to hunt fish. Communication or fishing are possible only within a distance (R) at which a signal can be detected. We determine detection distance by comparing the power spectra of the ambient noise and the received signal, with attention to the auditory response curve of the receiver. In Haro Strait, the center of the SRKW critical habitat, about 21 commercial ships per day, increases the ambient noise level by about 20 dB. To assess how ship noise may affect the SRKW communication and hunting, we define the fractional reduction in the zone of audibility at any location and time as the ratio of the area where signal detection is expected to occur in the increased noise regime to the maximum detection area expected under ideal conditions $R^2/Rmax^2$. We map the decreased zones of audibility in Haro Strait during average and extreme ship noise by combining field measurements of spreading rates with source power spectra (1–100 kHz) of common SRKW signals and typical ships.

3:10

4pAB6. Shipping noise and vocal compensation by Southern Resident killer whales: Haro Strait as a study case. Jason D. Wood (SMRU Ltd., P.O. Box 764, Friday Harbor, WA 98250, jw@smru.co.uk), Peggy Foreman (Univ. of Washington, Seattle, WA, 98195), Val Veirs, and Scott Veirs (Beam Reach, Marine Sci. and Sustainability School, Seattle, WA 98115)

02
5/13/11
by: Scott Ve.

Southern resident killer whales (SRKWs) use acoustic signals to navigate, forage, and facilitate social dynamics. Researchers have published evidence that suggests SRKW compensate for increased background noise by increasing the source level and duration of their signals. Unpublished reports have also suggested that SRKW may compensate for background noise by repeating their signals and by preferentially using certain signal types. Most of this work has focused on noise from whale watching vessels or general background noise. Haro Strait is both the center of the summertime home range of the SRKW and an important shipping channel. From September 2009 to December 2010 almost 10 000 ships transited through Haro Strait with an average of 21 ships passing per day. Ship transits in Haro Strait can increase background noise by up to 20 dB and are detectable above background noise for up to 30 min. This may be impacting the ability of SRKW to detect and utilize their acoustic signals. A five hydrophone array and Automatic Identification System receiver located at the Lime Kiln Lighthouse were used to record passing ships and SRKW in Haro Strait. This project investigates signal compensation strategies in SRKW in correlation with increased noise from passing ships.

3:25

4pAB7. Orca hearing weighted decibels: Underwater sound measurements appropriate to studies of Orcinus (killer whales). David Bain (Friday Harbor Labs, Univ. of Washington, Friday Harbor, WA 98250, dbain@u.washington.edu), Scott Veirs, and Val Veirs (Beam Reach Marine Sci. and Sustainability School, Seattle, WA 98115)

In community noise studies, sound levels are usually measured under the dB-A weighting scheme, which was introduced 50 years ago in an effort to match noise measurements to the response of human listeners. Here we propose an underwater noise decibel weighting scheme matched to the hearing sensitivity of killer whales (dB-O). This scheme is based on a convolution of the spectral energy of sound with the frequency-specific hearing detection thresholds of killer whales. The biological significance of noise sources may be more readily discerned if underwater sounds are quantified dB-O weighted. Further, use of this measure would emphasize the importance of broad-band measurement of noise rather than characterizing noise sources by the frequency with the peak power-spectral density and the source level of low frequency components. We compare the measures of representative noise sources, which have been recorded within the range of Southern Resident Killer Whales, including small boats, ships, airguns, and midfrequency sonar, using both flat and db-O weighted levels. While dB-O provides a more relevant characterization of noise than flat measurements (e.g., for predicting noise-induced stress), more detailed measurements will be required to address masking of biological signals, whose frequency structure varies with type of phonation and direction.

3:40—3:55 Break

3:55

4pAB8. The relationship between anthropogenic noise and frequency contours of southern resident killer whale (Orcinus orca) stereotyped vocalizations. Jennifer B. Tennessen (Huck Inst. of the Life Sci., Penn State Univ., 165 Appl. Res. Lab., University Park, PA 16802, jbt148@psu.edu)

Marine mammals evolved alongside naturally occurring noise in underwater environments, but recent increases in ocean anthropogenic noise present challenges to effective acoustic communication. Several species of marine mammals have demonstrated compensation strategies to avoid masking, including modifying call duration, frequency, rate, or amplitude. This research explored the relationship between noise and the vocalization behavior of endangered southern resident killer whales threatened in part by vessel traffic. Using extracted frequency contours obtained from recordings made between 2001 and 2007 from bottom-mounted hydrophones, the null hypothesis of no difference in the frequency contours of stereotyped calls across noise classes was tested. The results will be discussed in the context of vocal compensation thresholds in species that communicate using stereotyped calls, and management implications will be suggested.

4:10

4pAB9. Are click rates in killer whales an indicator of group behavior and foraging hotspots? Erica L. Beneze (Beam Reach Marine Sci. and Sustainability School, Univ. of Arizona, 1013 E. West Circle Dr., Tucson, AZ 85719, ebeneze@email.arizona.edu), Jason Wood (Beam Reach Marine Sci. and Sustainability School, SMRU Ltd., Friday Harbor, WA 98250), Scott Veirs (Beam Reach Marine Sci. and Sustainability School, Seattle, WA 98115), and Val Veirs (Beam Reach Marine Sci. and Sustainability School, Friday Harbor, WA 98250)

Killer whales use sound to communicate, find food, and navigate through the ocean. Southern Resident killer whales are specialized hunters and predominantly target Chinook salmon. It is presumed that these whales use echolocation clicks to distinguish between different species of salmon and to navigate. If this is the case, then click rates should vary by group behavior as the need for locating prey and navigating change. It has also been suggested that certain areas are utilized heavily by this population for foraging (hotspots) and some of these areas have been included in NOAAs proposed “no-go” zone. If click rates during foraging are distinct, then hotspots should be identifiable by click rates. This study tested if click rates varied by behavior state and geographic area. Group behavior was categorized into five states: foraging, traveling, milling, resting, and socializing. Click rate varied significantly by behavior state and by area. Socializing had the highest click rate followed by foraging, traveling, milling, and then resting. The Southern Residents had higher click rates in foraging hotspots.

4:25

4pAB10. Killer whale vocal behavior during joining events. Dawn M. Grebner (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093, dgrebner@ucsd.edu) and David L. Bradley (Grad. Prog. in Acoust., Penn State Univ., State College, PA 16804)

Killer whale discrete pulsed calls (DPCs) have been hypothesized to maintain group cohesion, but the specific usage of these calls in relation to animal movements (trajectory, orientation, joinings, etc.) is not well understood. We examine here the use of DPCs by individuals and small groups involved in joining events. DPC rates were examined before and after joining events, while vocal behavior during call initiations and one- and two-way vocal exchanges were also explored. Acoustic data were obtained using a triangular hydrophone array in Johnstone Strait in the summers of 2006 and 2007, while killer whale behaviors and locations were observed from a cliff using a video camera and theodolite, respectively. The relative arrival times of the killer whale sounds at each array hydrophone was used to determine the location of the vocalizing whale. Sounds were then associated in time and space with the corresponding killer whale’s position and behavior. Both DPC rates and the number of two-way vocal exchanges were found to be greater prior to joinings. These results indicate that DPCs play an important role in the vocal exchanges between killer whales prior to joinings.

4:40

4pAB11. Acoustic monitoring of killer whale populations off the west coast of Vancouver Island. Amalis Riera (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada), John K. Ford (Fisheries and Oceans Canada, Nanaimo, BC V9T6N7, Canada), John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093), and N. Ross Chapman (Univ. of Victoria, Victoria, BC V8P5C2, Canada)

Killer whales inhabiting the waters of British Columbia and Washington are at risk. The occurrence of these animals has been extensively monitored in protected inshore waters off the east coast of Vancouver Island and Puget

Sound, but there has been very little visual effort off Southwestern Vancouver Island and Washington. The objective of this study is to use acoustic techniques to determine the seasonal use of that area by different lineages of killer whales. Other questions include whether the current critical habitat needs to be expanded, or if there is any habitat sharing between different populations. Long-term acoustic data were obtained from passive acoustic devices including high frequency acoustic recording packages at three different sites. The recordings were analyzed using long term spectral averages, a tool that allows efficient analysis of large data sets in shorter periods of time. California-type transient calls were heard throughout the year, Southern Residents were detected from February through June, and Northern Residents were found from July through September. These results show a considerable overlap in habitat range between Southern and Northern Residents, as the latter use the southern parts of their range more frequently than previously thought.

4:55

4pAB12. Monitoring southern resident killer whale behavior on the outer coast of Washington using passive acoustics. Candice K. Emmons, M. Bradley Hanson, Marla M. Holt, Dawn P. Noren (NOAA/NMFS/Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. E, Seattle, WA 98112, candice.emmons@noaa.gov), and Marc O. Lammers (Hawaii Inst. of Marine Biology, Kailua, HI 96734)

Killer whales produce population-specific pulsed calls, and remote acoustic monitoring has been utilized to better assess the seasonal distribution and movements of southern resident killer whales (SRKWs) in the coastal waters of Washington State due to the many factors that limit visual sightings. In 2008 and 2009, Ecological Acoustic Recorders (EARs) were deployed at four locations spanning the Washington coast. The EARs were duty cycled to record for 30 s at 5- and 7-min intervals providing additional information about length of detections and acoustic behavior. SRKW were detected at all four locations. 26 of the 48 SRKW detections had sufficient data for analysis of acoustic behavior, and detections ranged in duration from a few minutes to one 24-h period. For each recording, we classified the presence, absence, and rates of various killer whale sounds to determine how they differed spatially and temporally. These preliminary data will be compared to multiple studies whose findings have demonstrated how the acoustic behavior of piscivorous killer whales differs among activity states: resting, socializing, foraging and traveling. These differences can provide information necessary to help identify important foraging areas and times on the outer coast in the absence of visual sightings.

5:10

4pAB13. Acoustic presence of killer whales in Resurrection Bay, Alaska during May–June 2010. Sandra Love (Alaska Pacific Univ., 4101 University Dr., Anchorage, AK 99508, slove@alaskapacific.edu) and Ana Širović (Univ. of Cali. San Diego, La Jolla, CA 92093-0205)

Killer whales produce three different types of calls: whistles, pulsed calls, and clicks. Pulsed calls have been studied in detail to determine their relation to social organization and behavior as well as pod or individual identification. The functions of whistles is not well known, but are thought to be used for close-range communication. Clicks are mainly used for echolocating. The vocal repertoire of killer whales recorded in Resurrection Bay, AK during May and June 2010 was investigated to determine their call characteristics and diel calling patterns. Data were recorded using an Ecological Acoustic Recorder, which sampled at 20 kHz on a 75% duty cycle. The characteristics of 12 loud and clear calls were compared to calls of known resident and transient pods [Yurk *et al.* (2002); Saulitis *et al.* (2005), respectively] to determine which pods were present in Resurrection Bay during this time. The most commonly identified calls were pulsed calls. Four calls could have belonged to resident pods (AD, AJ, or AK) and one could have belonged to the AT1 transient pod. Calling occurred more often during the day, but there was no real night time during the survey. Daytime calling may have been used for socializing.

5:25

4pAB14. Vocalizations and site fidelity of transient killer whales of the Pribilof Islands, Alaska. Kelly A. Newman and Alan M. Springer (School of Fisheries and Ocean Sci., Univ. of Alaska Fairbanks, 905 N. Koyukuk Dr., 245 O'Neill Bldg. P.O. Box 757220, Fairbanks, At 99775-7220)

Acoustic recorders can be superior to visual techniques for detecting marine mammals. Monitoring the presence of predators that exploit depleted prey populations, such as killer whales and northern fur seals at the Pribilof Islands in the Bering Sea, At, facilitates a better understanding of the role predators play in prey population dynamics and ecosystem function. Killer whales frequent the Pribilofs and prey upon fur seals when they are abundant during spring-fall, but their broader geographic range, stock structure, and fidelity to this predation hot spot are unknown. Identifying killer whale call types can address these questions because killer whale groups share calls. This study documented nine killer whale call types at the Pribilofs during the summers of 2006–2008. Calls were characterized by spectral and temporal properties, component structure and contour shape. The two most common calls were detected at one island more than another 64 km away, which suggests site preference. Calls during feeding observations are described as well as the efficacy of using bottom-mounted recorders to study ecological interactions. Pribilof calls are structurally distinct from other published records of Alaskan killer whale calls, and with further study will foster a deeper understanding of predator impacts in biologically sensitive areas.

Session 4pBA

Biomedical Acoustics: Ultrasound Imaging and Numerical Methods

Francesco P. Curra, Cochair
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Jonathan Mamou, Cochair
Riverside Research Inst., F. L. Lizzi Center for Biomedical Engineering, 156 William St., New York, NY 10038

Contributed Papers

1:00

4pBA1. Methods of quantifying muscle properties in cervical pain via sonoelastography and tactile imaging. Jeffrey J. Ballyns (Dept. of Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, jballyns@gmu.edu), Vladimir Egorov, Armen Sarvazyan (Artann Labs., Trenton, NJ 08618), Jennifer Hammond, Jay P. Shah (Clinical Ctr., Bethesda, MD 20892), Lynn Gerber (George Mason Univ., Fairfax, VA 22030), and Siddhartha Sikdar (George Mason Univ., Fairfax, VA 22030)

Myofascial trigger points (MTrPs) are a very common, yet poorly understood and overlooked, cause of nonarticular musculoskeletal pain. MTrPs are localized, stiff, hyperirritable tender nodules, palpated in taut bands of skeletal muscle. We are investigating sonoelastography and tactile imaging as complementary objective methods to measure the physical and mechanical properties of MTrPs. Sonoelastography was performed with an external 92-Hz vibration in the upper trapezius muscles in patients with acute neck pain ($n=45$). The area of reduced vibration amplitude was measured as an estimate of the size of the stiff MTrPs. A subset of patients ($n=3$) was studied via tactile imaging, using an array of pressure sensors mimicking a physical examination. Young's modulus of MTrPs was estimated using a previously developed technique for breast lesion characterization. Using sonoelastography, active sites (spontaneously painful with palpable MTrPs) had larger MTrPs (0.57 ± 0.20 cm^2) compared to latent (MTrPs painful on palpation) sites (0.36 ± 0.16 cm^2), or palpably normal (0.17 ± 0.22 cm^2) sites [$P<0.01$, one-way analysis of variance (ANOVA)]. Using tactile imaging, preliminary trends show that active sites are likely to be stiffer (61 ± 25 kPa) than latent sites (38 ± 18 kPa). These results demonstrate that MTrPs may be objectively imaged and quantitatively characterized. Further studies are needed to confirm these preliminary results.

1:15

4pBA2. Detection of hemorrhage regions in the brain using ultrasound vibrometry. Peter Cameron, Peter Rogers, Michael Gray, and James Martin (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332)

Detection of hemorrhage regions in the brain using an ultrasound technique was investigated. Ultrasound techniques are desirable to complement magnetic resonance imaging and computed tomography x-ray techniques for several reasons. Ultrasound does not subject the body to harmful radiation, the equipment is more portable and relatively inexpensive, and ultrasound does not have restrictions on use around metal objects. The technique could thus have application to monitoring patients with a high probability of brain hemorrhage. Efforts have been focused on developing a finite element model and corresponding laboratory experiment. The approach was to apply a low-frequency translational vibration which generated shear waves in the tissue material that could be interrogated using a ranging ultrasonic vibrometer. The experiment consisted of a synthetic tissue phantom encapsulated in a solid shell. The material properties in a hematoma vary with time as coagulation occurs, beginning as a liquid and transitioning to a soft elastic solid. The relatively short wavelength in an inclusion containing coagulating blood compared to that in tissue provided contrast to the inclusion region. The effect was enhanced by the fact that wavelengths at practical excitation frequencies were comparable to typical inclusion dimensions causing high amplitudes due to resonance.

1:30

4pBA3. Three dimensional impedance map analysis of rabbit liver. Alexander J. Dapore, Alexander D. Pawlicki, Sandhya Sarwate, and William D. O'Brien, Jr. (Univ. of Illinois at Urbana Champaign, 405 N. Matthews, Urbana, IL 61801)

One challenge of quantitative ultrasound is the identification of the scattering sites in tissue. Three dimensional (3-D) impedance maps (3DZMs) created from a series of histological tissue slide images are a useful tool to identify the scattering structures. 3DZMs are virtual (computational) data sets of real tissue that can be used to study fundamental ultrasonic properties. In this work fatty rabbit liver was chemically fixed and thinly sliced (3 μm) to create a series of H&E histology images. These images were realigned to one another using a registration scheme and pixels were assigned an impedance value to create a 3-D map of acoustic impedance. Through a power spectral analysis of the reconstructed 3-D volume, the effective scatterer diameter was estimated for the tissue using the fluid filled sphere form factor model. The results showed that when weighting the estimation toward smaller scatterer sizes, the effective scatterer diameter was 7.04 ± 1.30 μm. In the actual tissue this diameter corresponds closely to the size of the liver cell nucleus. These results provide encouragement that the reconstructed 3-D volume is an accurate acoustic representation of the tissue, and suggest that the nucleus could be a primary source of scattering in fatty liver. [Work supported by NIH Grant No. CA111289.]

1:45

4pBA4. Time-reversal techniques in ultrasound-assisted deep vein thrombosis treatment: Technology development and *in vitro* evaluation. George K. Lewis, Jr., William L. Olbricht (Dept. of Biomedical Eng., Cornell Univ., Ithaca, NY 14853), and Armen Sarvazyan (Artann Labs., Inc., West Trenton, NJ)

We describe a thrombolysis method that combines time-reversal acoustic (TRA) focusing with an AngioJet thrombectomy catheter to improve the dissolution of thrombus in the treatment of deep venous thrombosis (DVT). Commercially available ultrasound-assisted DVT systems have been shown to accelerate thrombolysis by up to 40% and reduce the amount of thrombolytic drug required for treatment by 50%–70%. Current ultrasound-assisted DVT treatments utilize a specially designed intravascular catheter with therapeutic ultrasound transducer on the tip and side. Additionally, MRI guided high intensity focused ultrasound systems have shown to rapidly lyse thrombus. We describe a novel TRA focusing system that is used to infuse fluids into the thrombus while simultaneously exposing the clot to safe levels of 1-MHz ultrasound energy. The system includes a combined infusion catheter-hydrophone, a ten-channel ultralow-output impedance amplifier, a broad-band ultrasound resonator, and MATLAB-based, TRA control and user-interface. TRA allows easy focusing of ultrasound therapy to the catheter tip from an external source, without complex phase-correction and array design. The system has been tested *in vitro* in a DVT phantom model, and results show that it provides 1-mm spatial resolution focusing. We present results in using TRA focusing to lyse bovine blood clots in the physiological size DVT phantom.

2:00

4pBA5. Three-dimensional detection of metastases in freshly excised human lymph nodes using quantitative ultrasound backscatter and envelope parameters. Jonathan Mamou (Lizzi Ctr., Riverside Res. Inst., New York, NY, jmamou@rri-usa.org), Alain Coron (CNRS and Univ. of Paris, Paris, France), Emi Saegusa-Beecroft, Masaki Hata (Univ. of Hawaii, Honolulu, HI), Michael L. Oelze (Univ. of Illinois, Urbana-Champaign, IL), Eugene Yanagihara (Univ. of Hawaii, Honolulu, HI), Tadashi Yamaguchi (Chiba Univ., Chiba, Japan), Pascal Laugier (CNRS and Univ. of Paris, Paris, France), Junji Machi (Univ. of Hawaii, Honolulu, HI), and Ernest J. Feleppa (Riverside Res. Inst., New York, NY)

High-frequency quantitative ultrasound (QUS) may offer a reliable means of identifying tumor foci in dissected lymph nodes. Detection of metastases is essential for staging and treatment planning. Conventional histopathology methods do not allow nodes to be examined over their entire volume. Therefore, our objective is to develop QUS methods to improve detection of clinically significant lymph-node metastases. A single-element 26-MHz ultrasound transducer was used to scan and digitally acquire rf echo-signal data in three-dimensional (3D) from more than 200 lymph nodes. Thirteen QUS estimates based on backscatter spectra and envelope statistics were computed in 3D. Serial-sectioning histology was performed at 50-μm intervals to depict cancer foci in 3D. Classification based on QUS estimates was performed using linear-discriminant analyzes; areas under ROC curves (AUCs) were computed. The most-significant QUS estimates for metastases detection were identified. Comparison of the 3D QUS results and 3D histology showed promising classification results. The AUC for the linear combination of four QUS estimates was 0.91 for a dataset of 73 breast-cancer nodes. Similarly, using only two QUS estimates, an AUC of 0.97 was obtained for a dataset of 143 colorectal-cancer nodes. These results suggest that QUS may be effective in distinguishing metastatic nodes from normal nodes.

2:15

4pBA6. Steering acoustic intensity patterns generated by time-reversal, to move therapeutic particles in tissue. Raghu Raghavan, Tim Poston (Therataxis, 1101 East 33rd St., Baltimore, MD 21218, raghu@therataxis.com), George K. Lewis, Jr. (Cornell Univ., Ithaca, NY 14853), Gaurav Gandhi, and Armen Sarvazyan (Artann Labs., Lambertville, NJ 08530, armen@artannlabs.com)

Convection-enhanced delivery (CED) is used in clinical trials to enhance the distribution of therapeutic molecules directly injected into tissue. Recent time-reversal acoustics (TRA) experiments [Lewis *et al.*, J. Acous. Soc. Am. **128** 2335 (2010)] show potential to improve CED with the use of delivery of acoustic forces within the tissue. TRA was used to focus ultrasound at the location of a transducer placed within rodent brain. Methods have been proposed to modify the TRA signals to steer the high intensity spots to new sites within a refractive inhomogeneous region such as the brain. We compare theory and measurement for the non-linear interpolation scheme proposed by Raghavan *et al.* [J. Acous. Soc. Am. **128** 2336 (2010)], first in inhomogeneous gel phantoms. We assess the accuracy as well as stability of control for intensity and placement of "hot spots." Time permitting, we shall present results for hot spot control in brain tissue and the resulting ability to improve distribution of small molecules delivered into brain tissue by CED. The ability to reliably steer high intensity spots in inhomogeneous media will allow applications beyond enhancing CED and beyond medicine.

2:30

4pBA7. A coded excitation technique for the functional imaging of coronary atherosclerosis using ultrasound contrast agents. Himanshu Shekhar and M.M. Doyley (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627)

Acute coronary syndromes may occur when life-threatening atherosclerotic plaques rupture in the advanced stages of cardiovascular disease. There is increasing evidence that plaque neovascularization accelerates the progression and disruption of atherosclerotic plaque. Plaque neovessels may be detected by subharmonic intravascular ultrasound (IVUS) imaging with ultrasound contrast agents (UCAs). The assessment of plaque neovascularity and perfusion at high spatial and contrast resolution may help identify those most at risk of acute coronary syndromes. While theoretical considerations dictate the use of high peak pressures and long excitation pulses for obtaining high contrast subharmonic IVUS images, microbubble disruption and the risk of hemorrhage limit the peak pressures practicable. Moreover, the use of long pulses degrades the axial resolution of achievable. In this paper, we report a novel excitation strategy using preemphasized chirps for microbubble insonation, which significantly enhances subharmonic signal from UCA. Therefore, low peak pressures can be employed to obtain high contrast resolution and the axial resolution can be restored by pulse compression. This technique was validated by numerical simulations and flow studies at high transmit frequencies (20 MHz). High spatial and contrast resolution achievable by this technique may significantly enhance the clinical potential for functional imaging of coronary atherosclerosis.

2:45—3:00 Break

3:00

4pBA8. Complex wavelet analysis of high frequency ultrasound backscatter from low echogenic biospecimen. Sushma Srinivas and Aaron Fleischman (Biomedical Eng., BioMEMS Lab., 9500 Euclid Ave., ND20, Cleveland, OH 44195)

Conventional microscopic methods of imaging biological specimen are subjected to various limitations. These include a tradeoff between high spatial resolution and field-of-view, limited depth-of-field, and the need to fix and stain the samples among many other limitations. Although ultrasound biomicroscopy allows the visualization of tissues at the microscopic level, improvement in the device resolution alone may not suffice to image a thin layer of biospecimen like a layer of cells due to their low echogenicity. A new method of image generation of the biospecimen using high frequency ultrasound without some of the aforementioned limitations is reported. Madin-Darby Canine Kidney epithelial cells in culture were imaged using a 40 MHz focused polyvinylidene fluoride polymer transducer. Modulation of the backscattered signal from this low echogenic biological specimen was detected. The backscattered signal better represented as a nonstationary signal when analyzed using complex wavelet analysis resulted in the detection of inflection points in the signal and the change in waveform due to propagation through the biological interface. Results demonstrate that the proposed technique better delineates a thin layer of low echogenic biospecimen layer than conventional or discrete wavelet methods. [Work supported by Cleveland State University Research Council's Doctoral Dissertation Research Expense Award Program and American Heart Association predoctoral fellowship].

3:15

4pBA9. A frequency compounding technique for improving the point spread function of superharmonic imaging systems. Koen W. A. van Dongen (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands), Mikhail G. Danilouchkine (Erasmus Medical Ctr., Dr. Molewaterplein 50, 3015 GR Rotterdam, The Netherlands), Libertario Demi (Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands), Paul L. M. J. van Neer, Nico de Jong (Erasmus Medical Ctr., Dr. Molewaterplein 50, 3015 GR Rotterdam, The Netherlands), and Martin D. Verweij (Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands)

Compared to fundamental imaging, harmonic imaging improves the axial and lateral resolution, while reducing the reflections from nearby artifacts and suppressing the effect of grating lobes. Superharmonic imaging combines reflections from the third, fourth, and fifth harmonic of the transmitted ultrasound pulse to improve the image quality of medical echography. A drawback of adding harmonic reflections is the possible degradation of the point spread function by ripple artifacts. The dual pulse technique reduces these ripples by transmitting two consecutive pulses with a slightly shifted frequency, and imaging the sum of both superharmonic reflections. Because two pulse-echo cycles are required, the frame rate is

half that of a single pulse technique. In this presentation, a frequency compounding technique is described for a superharmonic imaging system with an interleaved phased array. With this technique, both frequency-shifted pulses are transmitted simultaneously by the odd and even transmit elements, respectively. Numerical results obtained with the iterative nonlinear contrast Source method are shown. Moreover, these are validated against experimental results. The results show that frequency compounding provides the benefits of the dual pulse technique while avoiding reduction of the frame rate. [Work supported by STW and NCF.]

3:30

4pBA10. A dual pulse technique for improving the point spread function of superharmonic imaging systems. Martin D. Verweij, Libertario Demi (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci, Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands), Paul L. M. J. van Neer, Mikhail G. Danilouchkine, Nico de Jong (Erasmus Medical Ctr., 3015 GR Rotterdam, The Netherlands), and Koen W. A. van Dongen (Delft Univ. of Technol., 2628 CJ Delft, The Netherlands)

Nonlinear propagation causes the generation of higher harmonics of the emitted fundamental spectrum. Nowadays, medical echography employs reflections of the second harmonic, because this yields improved axial and lateral resolutions, and less reflections from nearby artifacts and grating lobes, as compared to fundamental imaging. To further exploit the benefits of higher harmonic imaging while keeping sufficient signal strength for detection, superharmonic imaging combines reflections of the third, fourth, and fifth harmonics. A drawback of adding harmonic reflections is the possible occurrence of ripples in the point spread function (PSF). Recently, a dual pulse technique was proposed for avoiding these ripples. This technique uses the emission of two consecutive pulses with a slightly different frequency, and performs imaging after summation of both superharmonic reflections. In this presentation, it is theoretically explained why this approach yields a better PSF than single pulse superharmonic imaging. For a superharmonic imaging system with an interleaved phased array, numerical results obtained with the iterative nonlinear contrast source method are shown. Moreover, these are validated against experimental results. The results confirm that the proposed technique significantly reduces ripple artifacts and gives a more compact PSF than the third harmonic alone. [Work supported by the STW and the NCF.]

3:45

4pBA11. Two-dimensional simulations of ultrasound propagation in random anisotropic media: Application to trabecular bone assessment. Marie Muller (Institut Langevin, ESPCI ParisTech, Univ. Paris Diderot, CNRS UMR 7587, 10 rue Vauquelin, 75231 Paris Cedex 05, France, marie.muller@espci.fr), Blandine Dobigny, Emmanuel Bossy (ESPCI ParisTech, 75231 Paris Cedex 05, France), and Arnaud Derode (Univ. Paris Diderot, 75231 Paris Cedex 05, France)

A better understanding of the mechanisms of ultrasound propagation in trabecular bone could considerably help improving the quantitative ultrasonic techniques commonly used to assess bone quality. Some phenomena experimentally observed in trabecular bone remain poorly understood, such as the possible propagation of two compressional waves with different velocities. In this study, elastic wave propagation has been simulated using a finite-difference time-domain method in two-dimensions. Trabecular bone was modeled by a binary random medium with fully controlled elasticity and anisotropy. To do so, elliptic-shaped patterns were randomly distributed on two-dimensional (2-D) maps with an orientation ensuring global anisotropy. The coherent wave was obtained by averaging over a large number of random maps. Several conditions for the observation of the two waves have been identified: (i) The propagation has to occur in a direction parallel to the main orientation of the medium. (ii) some of the elliptic pattern elements had to be connected, which suggests the importance of a percolation threshold. (iii) It is necessary to take into account shear waves in the solid phase. This suggests that bone microarchitecture parameters (anisotropy and connectivity) could be retrieved from ultrasonic measurements, improving the evaluation of fracture risk.

4:00

4pBA12. Numerical simulation of focused nonlinear acoustic beams: A Fourier continuation direct solver and comparisons to a paraxial approximation. Robin O. Cleveland, Theresa Y. Cheung (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215), Nathan Albin, and Oscar P. Bruno (Caltech, Pasadena, CA 91125)

A recently developed Fourier-continuation (FC) method is used to develop a numerical algorithm, which can accurately solve the fully nonlinear acoustic vector wave equation for the type of hundred-wavelength domains arising in the field of focused medical ultrasound. The FC nonlinear-ultrasound solver is used to outline the domain of applicability of the widely used Khokhlov–Zabolotskaya–Kuznetsov (KZK) approximation in focused ultrasound particularly in configurations arising in the field of high intensity focused ultrasound (HIFU). Good agreement was found for small-radius transducers; but as the source radius was increased to dimensions equivalent to practical HIFU sources the KZK approximation leads to errors in the location of the predicted focus. Examples for media with multiple scattering centers also show the utility of the FC approach in modeling complex media. In general the FC method produces solutions with significantly reduced discretization requirements over those associated with finite-difference and finite-element methods, and they lead to improvements in computing times by factors of hundreds and to thousands over competing approaches. [Work supported by NSF Grant No. 0835795.]

4:15

4pBA13. Numerical modeling of photoacoustic imaging of brain tumors. Kamyar Firouzi and Nader Saffari (Dept. of Mech. Eng., Univ. Coll. London (UCL), Torrington Pl., London WC1E 7JE , United Kingdom, kamyar.firouzi.09@ucl.ac.uk)

Photoacoustic imaging has shown great promise for medical imaging, where optical energy absorption by blood haemoglobin is used as the contrast mechanism. A numerical method has been developed for the *in silico* assessment of the photoacoustic image reconstruction of the brain. Image segmentation techniques were used to prepare a digital phantom from MR images. Then, light transport through brain tissue was modeled using the finite element approach. The resulting acoustic pressure was then estimated by pulsed photoacoustics considerations. The forward acoustic wave propagation was modeled by linearized coupled first order wave equations and solved by the acoustic *k*-space method. Since skull bone is an elastic solid and strongly attenuates ultrasound (due to scattering and absorption), a *k*-space method was developed for elastic media. To model scattering effects, a new approach was introduced based on propagation in random media. In addition, absorption effects were incorporated using a power law. Finally, the acoustic pressure was reconstructed using the *k*-space time reversal technique. The simulations were run in three dimensions to produce the photoacoustic tomogram of a brain tumor. The results show that relying on optical energy absorption by blood hemoglobin as the contrast mechanism, as is the current practice, leads to poor image quality.

4:30

4pBA14. A first-order *k*-space model for elastic wave propagation in heterogeneous media. Kamyar Firouzi (Dept. of Mech. Eng., Univ. Coll. London (UCL), Torrington Pl., London WC1E 7JE, United Kingdom, kamyar.firouzi.09@ucl.ac.uk), Benjamin Cox (Univ. Coll. London (UCL), London WC1E 7JE, United Kingdom), Bradley Treeby (Comput. Sci. Australian Natl. Univ., Australia), and Nader Saffari (Univ. Coll. London (UCL), London WC1E 7JE, United Kingdom)

A pseudospectral model of linear elastic wave propagation is described based on the first order elastodynamic equations. A *k*-space adjustment to the spectral gradient calculations is derived from the dyadic Green's function solution to the second-order elastic wave equation and used to ensure the solution is exact for homogeneous wave propagation for time-steps of arbitrarily large size. This adjustment also allows larger time-steps without loss of accuracy in weakly heterogeneous media. Along with an appropriate smoothing function, the model is applied for media with high-contrast inhomogeneities. An absorbing boundary condition has been developed to effectively impose a radiation condition on the wavefield. The staggered grid, which is essential for accurate simulations, is described in detail, along with other practical details of the implementation. The model compares favorably to exact solutions for canonical examples and to the conventional pseudospectral and finite difference time-domain codes. The numerical results show the accuracy of the method. It is also very efficient compared to alternative numerical techniques due to the use of the FFT to calculate the gradients in k space leading to reduced number of point per-wavelength and larger time-steps are made possible by the k-space adjustment.

4:45

4pBA15. Optimizing transient beamforming with FOCUS ultrasound simulation software. Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2220 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu) and Dustin E. Kruse (Univ. of California, Davis, CA 95616)

The speed and accuracy of FOCUS relative to other programs provide a significant advantage for the evaluation of transient beamforming algorithms. In the farfield of individual array elements, effective transmit focusing for transient beams is achieved when the time delays are determined from the distance between the center of the element and the desired focal point. However, when focusing in the paraxial region near the phased array, the farfield approach for calculating time delays is suboptimal. To address this problem, an optimal beamforming algorithm is derived for transient nearfield simulations in FOCUS. This beamforming algorithm utilizes an approach that rapidly converges to the optimum value by computing the transient pressure at three or four equally spaced time points for each element. The beamforming algorithm sequentially evaluates the optimal time delay for each array element, achieving an additional reduction in the computational effort by applying the computed offset as the initial offset for the next element. After quickly determining the location of the pressure peak in each transmitted waveform, the algorithm selects the time delays that achieve the maximum value of the peak pressure at the intended focus. [Work supported in part by NIH Grant No. EB012079].

THURSDAY AFTERNOON, 26 MAY 2011 DIAMOND, 1:30 TO 4:15 P.M.

Session 4pEA

Engineering Acoustics: Material Properties and Applications

Joseph F. Vignola, Chair
Catholic Univ., Mechanical Engineering, 620 Michigan Ave., Washington, DC 20064

Contributed Papers

1:30

4pEA1. Disorder in subordinate oscillators arrays used to shape the response of dynamic systems. Aldo A. J. Glean, Joseph F. Vignola, John A. Judge, Teresa J. Woods, and Patrick F. O'Malley (Mech. Eng., Catholic Univ., 620 Michigan Ave., Washington, DC 20064, 10glean@cardinalmail.cua.edu)

Earlier work has shown [J. Acoust. Soc. Am. **126**, 1 (2009)] that the time and frequency responses of a lightly damped oscillating structure can be tailored by attaching an array of much smaller subordinate oscillators. Exact analytic governing equations of motion are used to describe the behavior of the coupled system that is composed of the primary system and the subordinate array. Further, it is shown that this tailoring can be achieved using a relatively small number (<100) of attached oscillators whose total mass is small (~<1%) relative to the primary structure. This presentation will show how disorder in the property distributions within the array of attachments affects the tailored responses. Then with this general understanding, applications including time domain step response manipulation, bandpass filtering, and multianalyte detection based on dynamic response of microcantilevers are discussed. These examples represent case studies where sensitivity to disorder is either an asset or an encumbrance.

1:45

4pEA2. Acoustic systems for detecting particle impacts in space. Robert D. Corsaro (Global Strategies Group, 2200 Defense Hwy., Crofton, MD 21114, bob.corsaro.ctr@nrl.navy.mil), Frank Giovane (Virginia Polytechnic Inst.), Jer-Chyi Liou (NASAnOrbital Debris Program Office), Mark J. Burchell (Univ. of Kent, Canterbury, United Kingdom), Nickolus Lagakos (Global Strategies Group), James Tressler (U. S. Naval Res. Lab.), and Vincent Pisacane (U.S. Naval Acad.)

This paper briefly describes acoustic systems currently under development for measuring the flux of micrometeorite and orbital debris in space, and detecting damaging impacts when they occur on spacecrafts or habitats. While the particles of interest here are typically small (less than 5-mm diameter) their high speed (greater than 5 km/s) makes an impact quite energetic. At these speeds an impact typically leaves a hole or crater ten times the diameter of the particle, so particles as small as 50 μm can disable a system or create a hazard. The systems described here span a range of purposes. One system is focused on astronaut safety and uses an array of piezoelectric sensors to detect, localize, and assess damage on habitats. Another is focused on scientific data collection and uses a large impact-area diaphragm instrumented with fiber optic sensors for measuring particle size distribution. Other systems use combinations of acoustic and non-acoustic sensors to provide additional information about the nature of the impacting particles. Laboratory tests of each system are described, and the range and limitations of each are briefly discussed. [This work is being supported by the NASA Orbital Debris Office.]

2:00

4pEA3. Audio system requirements for remote operation of unmanned ground systems. Daniel J. Domme, Jr. (The Graduate Program in Acoust., The Penn State Univ., Appl. Sci. Bldg., University Park, PA 16802, domme@psu.edu) and Karl M. Reichard (The Penn State Univ., State College, PA 16801)

Remotely controlled robotic ground vehicles can perform a variety of tasks where a human presence is undesired or dangerous. These unmanned vehicles are capable of assisting in explosive ordinance disposal, search and rescue operations, as well as other tasks in hazardous environments. Due to the often volatile nature of such scenarios, a more comprehensive understanding of the acoustic environment is often desirable or required by a remote operator to aid in critical work. This presentation summarizes the limitations of selected current architectures of acoustic sensors in unmanned ground vehicles. The presentation also reviews the potential benefits to operator immersion in the acoustic environment by using two-channel audio in tandem with the remote video camera feed to more accurately recreate sound directionality.

2:15

4pEA4. Using sound speed to determine volume fractions in a two-phase flow. Anirban Chaudhuri, Curtis F. Osterhoudt, and Dipen N. Sinha (Los Alamos Natl. Lab. (MPA-11), P.O. Box 1663, MS D429, Los Alamos, NM 87545, anirban@lanl.gov)

This paper presents a method of determining the volume fractions of two immiscible fluids in a two-phase flow by measuring the speed of sound through the composite fluid and the instantaneous temperature. Two separate algorithms are developed, based on earlier work by Urick [J. Appl. Phys. (1947)] and & Toksoz [Geophys (1974)]. The main difference between these two models is the representation of the composite density as a function of the individual densities; the former uses a linear rule-of-mixtures approach, while the latter uses a non-linear formulation. Both methods lead to a quadratic equation, the root of which yields the volume fraction (ϕ), subject to the condition $0 \leq \phi \leq 1$. We present results of a study with mixtures of crude oil and processed water, and a comparison of our results with a Coriolis meter. The fluid densities and sound speeds are calibrated at various temperatures for each fluid component, and the coefficients are used in the final algorithm. Analytical and numerical studies of sensitivity of the calculated volume fraction to temperature changes are also presented. [This work was supported by Chevron USA.]

2:30

4pEA5. A multimicrophone tube for measurement of material acoustic absorption. C. H. Oppenheimer (Hewlett-Packard, 18110 SE 34th St., Vancouver, WA 98683)

A multimicrophone approach for measurement of the acoustic absorption of small material samples is presented. Like the two-microphone method of ASTM E1050, the multimicrophone tube involves a material sample at one end of a tube and a loudspeaker at the other end. Unlike the two-microphone method, the multimicrophone tube uses an arbitrary number, greater than two, of microphones and a distinct algorithm to relate microphone pressures to material acoustic properties. A multimicrophone tube with four microphones was fabricated. Using this tube, material acoustic absorption was determined by the two-microphone approach and by the multimicrophone approach. The multimicrophone approach was observed to extend the bandwidth of accurate determination of material acoustic absorption. Accuracy improvements occurred not only at low frequencies, where tube axis phase changes are small, but also at high frequencies beyond the frequency at which a half wavelength spans the microphone pair of the two microphone method of ASTM E1050.

2:45—3:00 Break

3:00

4pEA6. Prediction of sound absorption characteristics of orifice plates with mean flow using the lattice Boltzmann method. Kaveh Habibi, Phoi-Tack Lew, and Luc Mongeau (Dept. of Mech. Eng., McGill Univ., Montreal, PQ H3A 2K6, Canada)

The purpose of this study was to evaluate the sound absorption characteristics of orifice plates in turbulent mean flows using numerical simulations. The production of vorticity at the orifice edge governs the sound absorption phenomena. The LBM-LES methodology was used to perform detailed numerical simulations of the unsteady turbulent field. The lattice Boltzmann method has several advantages over continuum based CFD methods for situations where the configurations of the duct, the orifice, and the flow structures through or in contact with porous/perforated media are geometrically complex. The absorption coefficient and the impedance of the perforated plate were obtained in a virtual three dimensional (3-D) impedance tube apparatus. Both the mean flow velocity profiles and the acoustic characteristics of a simple circular orifice were found to be in good agreement with available experimental data. The model was then exercised to investigate the acoustic properties of orifices with complex shapes over a range of mean flow velocities.

3:15

4pEA7. Analyzing acoustic response of orifices using transfer matrix method. Fei Liu, Matias Oyarzun, and Lou Cattafesta (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, 231 MAE-A, P.O. Box 116250, Gainesville, FL 32611, cattafes@ufl.edu)

The impedance of an orifice is critical in many applications such as sound absorption, fluidic actuation, and stability prediction of hydraulic piping systems. In this work, a transfer matrix analysis of orifices (circular orifice, slit, and perforated plate) is presented, which provides physical insights into the acoustic behavior of orifices under unsteady excitation. Specifically, the acoustic transfer matrix (TM) for an orifice in the presence or absence of mean flow is developed and compared to a lumped element model (LEM). An experimental investigation is conducted in a plane wave tube (PWT) for circular and slit orifices in the absence of mean flow for a variety of frequencies and sound pressure levels. The experimental results are compared with TM and LEM predictions. The TM predictions for circular orifices in the presence of mean flow are also compared with other available experimental data.

3:30

4pEA8. Experimental determination of the complex Poisson's ratio of viscoelastic materials. François M. Guillot and D. H. Trivett (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332-0405, francois.guillot@me.gatech.edu)

An empirical procedure to determine the dynamic, complex Poisson's ratio of viscoelastic materials as a function of temperature and hydrostatic pressure is presented. In this procedure, the Young's modulus and the bulk modulus of the material are measured, and the complex values of these moduli are combined to compute Poisson's ratio. The two independent measurement systems used to obtain these data are described. The Young's modulus system requires a sample cut in the shape of a bar and relies on traditional resonance measurements for low-frequency data as well as on wave speed measurements for higher-frequency data; measurements are performed in air. The bulk modulus system measures the dynamic compressibility of a sample of arbitrary shape immersed in Castor oil. Data can be obtained at frequencies typically ranging from 50 Hz to 5 kHz, at temperatures comprised between -2 and 50° C and under hydrostatic pressures ranging from 0 to 2 MPa (Young's) or 6.5 MPa (bulk). The two moduli can also be combined to compute any other modulus, and the two systems therefore allow the complete elastic characterization of a homogeneous and isotropic material. Data obtained on a nearly incompressible rubber are presented.

3:45

4pEA9. Measurement of planar poisson's ratio for piezoceramic disks. Dmytro Libov and Viatcheslav Meleshko (Dept. of Theoretical and Appl. Mech., Kiev Nat. Taras Shevchenko Univ., 64 Volodymyrska St., Kiev, Ukraine, dmytro.libov@gmail.com)

Previous methods for a determination of a planar Poisson's ratio in ferroelectric ceramic disks were proposed for thin disks. If a thickness of a disk strongly affects the vibrations, we get error in determination of Poisson's ratio in usage 1-D theory of disk vibrations. Planar Poisson's ratio in this work has been determined from an overtone method and a resonant method for a thick piezoceramic disk. The theoretical and the experimental contour modes with different types of vibrations symmetry have been compared in the resonant method. For the first theoretical contour mode the thickness correction was applied. The first order contour resonant frequency was determined in the experiment after the cutting of split electrodes. The overtone method is based on the solution of three-dimensional problem of vibrations of a finite cylinder. Only one experiment is needed in the overtone method. The obtained results have been compared with the handbook data. It may be concluded that the proposed methods provide an accurate determination of planar Poisson's ratio in thick piezoelectric ceramic disks, when all experimentally measurements were made on the single piece of the piezoceramic material.

4:00

4pEA10. Transmission of complex sounding signals in human respiratory system. Vladimir Korenbaum, Anatoly Nuzhdenko, Alexander Tagiltsev, Anatoly Kostiv, Nikita Lopatkin (Pacific Oceanological Inst., FEB RAS), and Alexander Dyachenko (A.M. Prokhorov General Phys. Inst., RAS)

A complexity of respiratory path structure has already caused assumptions on existence of several ways of sound transmission from mouth to chest wall. The objective is a study of these mechanisms. The studied sample included 25 healthy subjects. They breathed with room air, helium-oxygen, and krypton-oxygen gas mixtures which had various sound velocities. Phase manipulated and frequency sweep signals were injected into mouth. Signals transmitted to chest wall sites (bottom part of trachea and basal area of right lung) were recorded by accelerometers. A convolution procedure was carried out. At least two signal arrivals are recognized in each convolution curve recorded above basal area for any gas mixture. One main arrival is recognized above trachea, which times are statistically dependent on gas mixture for all signals. The first arrival times above basal area of lung are also dependent on gas mixture. However, the second arrival times above basal area of lung are independent of gas mixture. Thus the arrival above trachea and the first arrival above basal area of lung, being dependent on filling gas sound velocity, are transmitted through airways lumen at least in part of their path. Alternatively the second arrival recorded above basal area of lung, being independent of filling gas sound velocity, seems to be transmitted through lung parenchyma. [The study was supported by RFBR grant 09-08-00105.]

THURSDAY AFTERNOON, 26 MAY 2011 — ASPEN, 1:00 TO 4:50 P.M.

Session 4pMU

Musical Acoustics and Engineering Acoustics: Optical Methods for Studying Musical Instruments

Thomas D. Rossing, Cochair
Stanford Univ., Dept. of Music, Stanford, CA 94305

Randy Worland, Cochair
Univ. of Puget Sound, Dept. of Physics, 1500 N. Warner, Tacoma, WA 98416-1031

Invited Papers

1:00

4pMU1. Some optical methods for modal analysis of musical instruments. Thomas D. Rossing (26464 Taaffe Rd., Los Altos Hills, CA 94022, rossing@ccrma.stanford.edu)

Because of their high spatial resolution, optical methods have become important for modal analysis of vibrations in musical instruments. Among optical methods for modal analysis are time-average holographic interferometry, pulsed TV holography, disital speckle interferometry, and scanning vibrometry. We review briefly principles and applications of optical methods for modal analysis.

1:20

4pMU2. Speckle imaging techniques for visualizing deflection shapes. Thomas Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789, tmoore@rollins.edu)

Visualizing the motion of musical instruments during play is often crucial to understanding the underlying physics. Yet the equipment required to view deflection shapes of large or oddly shaped objects is typically expensive and difficult to build, and the cost of commercially available systems is typically out of the price range of all but the most well-funded institutions. Over the past several years we have developed imaging techniques based on electronic speckle pattern interferometry that can be used to visualize the deflection shapes of vibrating objects, with the emphasis being on low cost and easy implementation. In this presentation we will review some of these techniques and discuss the details of construction and use. Each apparatus can be built by undergraduate students in a relatively short period of time and the results are suitable for journal publication. Several examples of images of deflection shapes of musical instruments will be presented.

1:40

4pMU3. Pitfalls of time resolved electronic speckle pattern interferometry. Wilfried Kausel (Inst. of Music Acoust., Univ. of Music, Anton v. Webernplatz 1, A-1030 Vienna, Austria, kausel@mdw.ac.at)

Electronic speckle pattern interferometry has turned out to be a method for steady state analysis of vibrating surfaces, which is easy to use and does not require an expensive setup. A new generation of digital high-speed cameras has now become available, which is significantly cheaper, faster, and more sensitive than prior generations, awakening the desire to study transient behavior interferometrically. Although it seems quite straightforward to do a quasi-static ESPI deformation analysis at frame rates of 10–30 fps, it is, in fact, not. From four different types of state of the art digital high-speed cams, all of them with datasheet specifications which should have made them well suited for the intended purpose, only one was able to produce recordings with satisfactory interferences. Recordings of all tested cameras will be presented and possible explanations for the interference problem will be discussed. A successful application of time-resolved ESPI to a plugged or struck musical instrument will also be demonstrated.

2:00

4pMU4. High speed electronic speckle pattern interferometry as a method for studying the strike on a steelpan. Andrew C. Morrison (Dept. of Phys., DePaul Univ., 2219 N. Kenmore Ave., Chicago, IL, 60614), Thomas R. Moore (Rollins College, Winter Park, FL 23789), and Daniel Zietlow (Univ. of Colorado at Boulder, Boulder, CO 80309)

Electronic speckle pattern interferometry (ESPI) is a useful method for characterizing the operating deflection shapes and modes of vibration of musical instruments. Using ESPI in conjunction with a high-speed camera, capable of capturing images at rates of several thousand frames per second, allows for time-resolved examinations of transient motion. High-speed ESPI movies of note strikes of a low-tenor (also called a soprano) steelpan were acquired while simultaneously recording the sound of the strike. The comparison of the time-resolved interferometry data with the analysis of the sound recordings allows for insights into the evolution of coupling between note areas.

2:20

4pMU5. Experimental study of vibraphone pitch bending using electronic speckle-pattern interferometry. Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416-1031)

Pitch bending on the vibraphone is an extended performance technique called for by modern composers dating back to the 1960s. After first striking the middle of a bar with a soft mallet in the normal manner, the pitch bend is obtained by pressing a hard mallet onto the bar at a nodal point and then sliding it away from the node as the note sustains. The audible result is a descending pitch, typically of about one semitone. Experimental data showing frequency vs location of the hard mall*et al.*ong the bar are presented and interpreted with the use of time-averaged electronic speckle-pattern interferograms showing the vibrational modes of the vibraphone bar. Frequency vs mallet mass data are also presented and discussed. In addition, it is shown that for some combinations of mallet type and vibrational mode this technique can produce an *increase* in frequency.

2:40

4pMU6. Vibration modes of clarinet reeds via digital electronic holography. Karl A. Stetson (Karl Stetson Assoc., LLC, 2060 South St., Coventry, CT 06238)

Discovered in 1964, holographic interferometry has provided vibration and deformation analysis of structures through the 1970s and into the 1980s. In the late 1980s, photographic holography gave way to digital electronic holography this greatly advanced its capabilities by making it possible not only to display interferograms in real time, but also to reduce fringe patterns to numerical data. This technology is widely used for verification of vibration modes of such engineering components as jet engine turbine blades. Holography has also been used to study the vibration modes of string instruments since the late 1960s. This paper applies it to the study of clarinet reeds. Although the vibration amplitude of such reeds when played is many times the level that can be studied holographically, examination of their basic vibration modes offers some insight into how they may function. A comparison is shown of the modes a wet cane reed to several synthetic reeds whose fundamental modes are shown to lie at lower frequencies than an equivalent cane reed. The presentation will conclude with some suggestions for studies of reed profiles and for measurement of reed motion under playing conditions.

3:00

4pMU7. Mode studies of plucked stringed instruments: Application of holographic interferometry. Bernard Richardson (School of Phys. and Astronomy, Cardiff Univ., 5 The Parade, Cardiff CF24 3AA, United Kingdom, richardsonbe@cardiff.ac.uk)

The acoustics group at Cardiff have used holographic interferometry for many years to study the vibrations of musical instruments. After a review of the technique and equipment and the particular strengths and weaknesses of this analysis tool, the paper will describe measurements on historic, modern, and experimental guitars and related instruments. These studies highlight the effects of strutting and bracing patterns used on the underside of the soundboard and the size and positioning of the bridge, which give insight how the design and construction of these instruments affect their mechanical vibrational properties and their acoustical function.

3:20

4pMU8. Measurement of mode shapes of musical instruments using a scanning laser Doppler vibrometer. Thomas M. Huber (Dept. of Phys., Gustavus Adolphus College, 800 College Ave., Saint Peter, MN 56082)

In studying musical instruments and other vibrating systems, optical methods, such as ESPI holography and laser vibrometery, have allowed non-contact measurements of operating deflection shapes. This talk will focus on the principles and operation of a scanning laser Doppler vibrometer system, as well as its application for measuring vibration of musical instruments. Examples will be presented for instruments including guitars and organ reed pipes, which show the capabilities of a Polytec PSV-400 scanning vibrometer system to measure steady-state deflection shapes at resonance frequencies of the system. Also demonstrated will be measurements of the vibration of the face of a guitar during the attack transient.

3:40

4pMU9. Optical methods in stringed instrument testing. Mark French and Haley Moore (138 Knoy Hall, 401 N. Grant St., West Lafayette, IN 47907)

Stringed instruments present an experimental challenge for several reasons. Radiated sound is a strong function of dynamic response, so particularly accurate measurements of structural and acoustic resonant frequencies are desirable. However, the structures tend to be very light and sensitive to the additional mass of contacting sensors. Thus, optical methods are attractive. The ready availability of lasers, inexpensive digital cameras, and powerful, inexpensive computers has made optical testing practical. As a result, a range of

methods has been applied to musical instruments. There are many ways to organize the methods, but an attractive one is whether they require coherent light for interference effects. A hierarchy of methods is presented along with representative results. The simplest methods simply track motion of a reflected beam across an optical sensor. Coherent methods start with laser vibrometry and move through holography and speckle pattern interferometry. We pay particular attention to an particular application of laser vibrometry and to a low cost speckle pattern interferometer.

4:00

4pMU10. Mode studies in a triangular bell plate. Uwe J. Hansen and John Garner (Dept. of Chemistry & Phys., Indiana State Univ., Terre Haute, IN 47809)

Triangular bell plates have been used as the poor man's substitute for hand-bells. As in other pitched percussion instruments, pitch identification largely depends on the harmonic relation between the most significant overtones. This work reports on experimental mode studies of a bell plate using electronic holographic interferometry and impact excited modal analysis. Experimental results are also compared with finite element mode calculations. The finite element calculations are in good agreement with the experimental data.

Contributed Papers

4:20

4pMU11. Acoustic effects of holes on commercial cymbals. Brooke R. Peaden and Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416)

In recent years, several cymbal companies have begun manufacturing cymbals with holes cut into them. The 10 -in. cymbal used in this study (Sabian Ozone Splash) contains six symmetrically placed 1.5 in. diameter holes. Electronic speckle pattern interferometry (ESPI) has been used

to collect mode shape and frequency data on brass plates and cymbals containing holes of various sizes. Data taken on a brass plate show that a single hole may cause mode frequencies to increase or decrease depending on the size and location of the hole relative to the modal pattern. As a hole can be treated as a region of both lower mass and lower stiffness, these competing frequency effects are to be expected. To illustrate the effect of the six holes on the Ozone cymbal, ESPI data are presented on a standard 10-in. Sabian splash cymbal (without holes) as six holes are added in a series of increasing diameters until the geometry of the Ozone cymbal is matched.

4:35

4pMU12. Visualization of weak shock waves emitted from a trombone. Kazuyoshi Takayama, Kiyonobu Ohtani, Toshihiro Ogawa, Takamasa Kikuchi, Reiko Takayama (Inst. of Fluid Sci., Tohoku Univ., 2-1-1 Katahira, Aoba, Sendai 980-8577, Japan, k.takayama@mac.com), and Toshinori Takahashi (Tohoku Univ., Sendai 980-8579, Japan)

The coalesce of compression waves propagating in pipes into weak shock waves is one of the topics of shock wave dynamics. We have worked for weak shock waves generated in automobile exhaust pipe lines [Sekine *et al.* (1990)] and train tunnel sonic booms in Japanese high speed train entry into a long tunnel [Takayama *et al.* (1995)]. Hirschberg *et al.* (1996) reported, for the first time, the emission of weak shock waves from a trombone blown in ff and Pandya *et al.* (2003) visualized weak shock waves emitted from brass instruments. So far we understood, these belong to our continuous academic curiosity. It is a highlight of brass instruments to produce dramatizing sounds. In the 631-bar of the fourth movement of Mahler's symphony No. 1, fff is specified for a third trombone. We then visualized, by using a set of 1 m diameter schlieren mirrors in schlieren optics and holographic interferometry, weak shock waves emitted from a trombone blown in ff and fff and measured pressure at the muzzle and horn and velocity at the horn.

Session 4pNS

Noise, Animal Bioacoustics, and Committee on Standards: Measurement and Assessment of the Soundscape in Parks and Wilderness Areas

Kurt M. Fristrup, Cochair
National Park Service, Natural Sounds Program, 1201 Oakridge Dr., Fort Collins, CO 80525

Paul D. Schomer, Cochair
Schomer and Associates, 2117 Robert Dr., Champaign, IL 61821

Invited Papers

2:20

4pNS1. Adapting noise measurement and modeling practices to meet legislative and policy mandates for National Park management. Kurt M. Fristrup (Natural Sounds and Night Skies Div. Natl. Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

Historical community noise studies focused on exposures corresponding to maximum tolerable annoyance and potential health risks. These criteria should rarely be pertinent for National Parks, but applicable knowledge and tools were developed by these studies. A-weighted measurements integrate sound energy across the audible spectrum to account for properties of human hearing. To improve the utility of A-weighted measurements, NPS is investigating band-limited dB(A) measures to exclude natural environmental sounds that have no bearing on the effects of anthropogenic noise. Leq metrics were shown to be reasonable for integrating time histories to predict human responses to noise exposure. NPS utilizes measurements of audibility to parse this measure into the extent of noise-free conditions and the Leq value when noise is audible. This approach distinguishes situations with chronic exposure to modest levels of noise from very infrequent exposures to loud events. Several noise models have been developed to support community noise management. These are used in park settings to model the kinds of individual noise events that can occur (a type of vehicle on a specific route). Interactive noise mapping tools have been developed to compute aggregate exposure from collections of many such events under different planning scenarios.

2:40

4pNS2. Environmental noise measurements at Mount Rushmore National Memorial. Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@schomerandassoc.com)

Assessing noise in national parks remains a subject of research and study for two general problem areas: the reaction to noise by park visitors and the effects of noise on wildlife. Traditional methods for environmental noise assessment such as methods to assess the effects of airport noise on residential communities are inappropriate and inadequate for the National Park issues, and Schomer and Stanley (2009) developed a research plan to gauge the sound quality of a hike, not annoyance. The goal of this research is to develop methods to assess the sound quality of different park soundscapes by being able to rate the acoustical experience of park visitors. This report concentrates on physical measurements of the acoustic environment. Our approach for the measurements was to select a trail that was used regularly, but not particularly heavily, that had regular, distinct anthropogenic noise, and that was readily accessible. This paper provides an analysis of the acoustical measurements. In particular, it provides an analysis of the data collected by the 20 fixed monitors, an analysis of the data collected by the mobile monitor, and an analysis of the GPS data used to establish the mobile monitor position as a function of time.

3:00

4pNS3. Spatial variation of natural ambient sound pressure levels in Rocky Mountain National Park. Daniel J. Mennitt (Natural Sounds and Night Skies Div., Natl. Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel.mennitt@partner.nps.gov)

The soundscape is a critical component of an ecological community, and knowledge of natural ambient sound pressure levels is crucial to assessing impacts of noise. However, the spatial correlation of natural ambient sound pressure levels is largely unknown and a given measurement may not be representative of the locale. Much anthropogenic noise can be considered point or line sources; the spatial correlation of the resulting sound pressure levels can be completely described by directivity and the inverse square law in an isotropic medium. However, most natural sources (wind, birdsong, rain, insects, etc.) may be better described in aggregate as an irregular source with stochastic characteristics. Recently, a 9-day long study collected continuous audio data at 18 locations over a roughly 3.5 s km^1 area in Rocky Mountain National Park. Analysis seeks to investigate the spatial variation of natural ambient sound pressure level measurements and the degree to which they are correlated across space. Results will suggest how sparsely an area can be measured as well as placement of a finite number of microphones to best observe the acoustical environment.

3:20

4pNS4. A subspace tracking method for detection, identification, and estimation of acoustical transient sources. Neil Wachowski and Mahmood R. Azimi-Sadjadi (Dept. of ECE, Colorado State Univ., 1373 Campus Delivery, Fort Collins, CO 80523, nswachow@engr.colostate.edu)

This study introduces a new method for detecting, identifying, and estimating transient signal and interference sources whose signatures may be present in vector-valued observations with additive Gaussian noise. It is assumed that, for each source type, the subspace it lies in and the dependencies between its successive signatures are known and that a maximum of one signal and one interference source is present at a given time. This framework accounts for the possibility of the presence of either no sources, a single source, or both signal and interference and, in the latter case, separate estimates of each source can be extracted from the observations. Results on simulated data and one third octave vector sequences representing real acoustical measurements attest to the effectiveness of the proposed method at performing these different tasks. [This work is sponsored under a cooperative agreement from the National Park Service (NPS) Contract No. H2370094000 (WASO).]

3:40

4pNS5. Automatically separating anthropogenic from natural sounds in parks. Jack Gillette and Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, gillett1@uni.illinois.edu)

The National Park Service needs to determine the extent of both man-made and natural sounds in national parks. To do this, they need to separate anthropogenic noises from natural ones. To be accurate, this requires hundreds of hours of sound data spread out over times of day, days of the week, and seasons, and must be spatially sufficient. With these much data to analyze, manual listening for audible tones and noises can be an impossible task. However, with a program that uses an algorithm to search for anthropogenic sounds, the task becomes much easier. Our hypothesis is that all anthropogenic noises except for jet aircraft will include tones below 1000 Hz. Our main purpose is to write software that flags anthropogenic sounds below 1000 Hz. Because anthropogenic and natural sounds are uncorrelated, we hope to flag virtually all anthropogenic sounds at the cost of falsely flagging some natural sounds. Furthermore, we can very accurately estimate the ALEQ for a given length of time based on a much smaller sample and then subtract that from the total to find the ALEQ for the anthropogenic sound. This paper discusses the development and testing of this process.

4:00

4pNS6. Modeling hikers' exposure to transportation noise in national parks and wilderness. Kenneth Kaliski, Steve Lawson (Resource Systems Group, 55 Railroad Row, White River Junction, VT 05001, kkaliski@rsginc.com), David Pettebone (Sci. Div. Yosemite Natl. Park, Yosemite Natl. Park, CA 95389), Peter Newman (Colorado State Univ., Fort Collins, CO 80521), Eddie Duncan, Brett Kiser, Eric Talbot, and Emily Eros (Resource Systems Group, Inc., White River Junction, VT 05001)

Results of research conducted in a variety of national park settings suggest that the quality of visitors' experiences is tied to the naturalness of the area's soundscape. Yet, visitor use in national parks tends to be concentrated in areas that are accessible or close to roads. As a result, surface transportation noise can significantly affect visitor experience quality. This paper describes tools and methods used to estimate the effects of various transportation planning options on hikers' noise exposure in national parks. These tools combine transportation noise mapping with GPS tracking and computer modeling of visitor use to map and quantify the effects of transportation planning alternatives on visitors' soundscape experiences. Examples of how these methods have been implemented in national parks are discussed.

4:20

4pNS7. Aircraft noise-dose-visitor-response relations for national parks. Grant S. Anderson (76 Brook Trail, Concord, MA 01742, gndrsn@comcast.net), Amanda S. Rapoza, and Aaron L. Hastings (Acoust., Volpe Natl. Transp. System Ctr., Cambridge, MA 02142)

The U.S. National Parks Overflights Act and the Air Tour Management Act require management of air tours over national parks. Unfortunately, past noise-dose-response relations do not suffice for predicting response of national park visitors who are not at home, are not exposed for months/years, and are not subject to jets on approach or departure, but instead to low-flying tour aircraft. Toward filling that insufficiency, this study was designed to understand the complex relationship between in-park noise exposure and park-visitor response. The study derives from a decade of visitor surveys and simultaneous sound measurements at ten frontcountry sites (scenic overlooks and short hikes) within four scenic national parks. This talk describes the analysis of these measurements and documents the six resulting dose-response relations—-two responses (annoyance and interference with natural quiet and sounds of nature) paired with three response dichotomizations (slightly to extremely, moderately to extremely, and very to extremely). In these relations, acoustic-dose predictors are augmented with visitor-specific percentages (visited site before, only adults in visitor group, natural quiet, and sounds of nature are very important). This augmentation further reduced site-to-site variability, suggesting that the current model is more applicable to other sites and settings.

4:40

4pNS8. A pilot survey of visitor perception of sounds along a trail. Arnav R. Pamidighantam and Paul D. Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, pmdghntm@gmail.com)

A pilot survey was conducted to determine the perception of noise by visitors along a hike at a national park by asking visitors to take a written survey on all or a large portion of predetermined route. The visitors were asked to stop at their discretion, several times, and rate the acceptability and their interpretation of the noises heard at the stop on a nine point bipolar scale. These questions were asked for a wide variety of sounds that may be heard on a trail. Also, a cumulative question was asked in order to rate the acceptability and

personal interpretation of noise at each stop as well as the hike as a whole. Analysis of the completed surveys was done to establish the effectiveness of the survey and any changes that would make the survey more user-friendly. This paper discusses the effectiveness and user-friendliness of the survey as well as the methods determined to increase these characteristics in the hope of better understanding how the visitor perception of a trail is determined by the noises along the trail.

5:00—5:30 Panel Discussion

THURSDAY AFTERNOON, 26 MAY 2011 WILLOW A, 1:30 TO 4:45 P.M.

Session 4pPA

Physical Acoustics and Engineering Acoustics: Violent Cavitation Activity in Water and Other Liquids II

Lawrence A. Crum, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Thomas J. Matula, Cochair
Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Invited Papers

1:30

4pPA1. Transient cavitation in high-quality factor resonators at high static pressures. D. Felipe Gaitan, Yuri A. Pishchalnikov (Impulse Devices, Inc., 13366 Grass Valley Ave., Grass Valley, CA 95945, gaitan@impulsedevices.com), Thomas J. Matula (Univ. of Washington, Seattle, WA 98105-6698), Charles C. Church (Univ. of Mississippi, Oxford, MS 38677), Joel Gutierrez, Corey Scott (Impulse Devices, Inc., Grass Valley, CA 95945), R. Glynn Holt (Boston Univ., Boston, MA 02215), and Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105-6698)

Cavitation collapse can generate intense concentrations of energy, sufficient to erode even the hardest metals and to generate light emissions visible to the naked eye [sonoluminescence (SL)]. The phenomenon of "single bubble sonoluminescence" (SBSL) in which a single stable cavitation bubble radiates light flashes each acoustic cycle typically occurs near 0.1 MPa static pressures. Impulse Devices, Inc. has developed a new tool for the study of SL and cavitation: a high quality factor, spherical resonator capable of achieving acoustic cavitation at ambient pressures in excess of 30 MPa. This system generates bursts of violent inertial cavitation events lasting only a few milliseconds (hundreds of acoustic cycles). Cavitation observed in this system is characterized by flashes of light with intensities up to 1000 times brighter than SBSL flashes as well as spherical shock waves with amplitudes exceeding 100 MPa (1000 bars) at 1 cm from the cavitation center. Computer simulations indicate shock wave amplitudes near the collapsing bubble around 1–10 TPa (10–100 mbars) and liquid temperatures on the order of 5000 K, possibly causing the liquid to become opaque. The implications of these extreme conditions on SL emission will be discussed. [Work funded by Impulse Devices, Inc. ACPT Contract No. W9113M-07-C-0178.]

1:50

4pPA2. Comparison of single bubble collapse and cluster collapse in a high pressure vessel. Thomas Matula, Brian MacConnaghy, Lawrence Crum (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), and Felipe Gaitan (Impulse Devices, Inc., Grass Valley, CA 95945)

Details of the collapse of single bubbles and bubble clusters leading to the emission of a shock wave under high overpressures will be presented. Ultrahigh speed photography captures the events at various stages of bubble (or cluster) growth and collapse. Shock waves are observed millimeters from the collapse center, suggesting very violent conditions at the source. For example, shock waves emitted by single-bubble sonoluminescence can reach 3 mm/μs about 5 μm from the bubble center. With a cluster, we observe shock waves at this speed over 500 μm from the center. The strength of the collapse is estimated by measuring the emitted shock wave velocity from images taken with the ultrahigh speed imaging system. Cluster collapses can be much stronger than from single bubbles. Cluster collapse appears to be initiated by the collapse of outer bubbles.

2:10

4pPA3. Shock-controlled bubble cloud dynamics and light emission. R. Glynn Holt, Phillip A. Anderson, Ashwinkumar Sampathkumar, Jonathan R. Sukovich (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215), and D. Felipe Gaitan (Impulse Devices, Inc., Grass Valley, CA 95945)

Cavitation bubble collapse can generate intense concentrations of mechanical energy, sufficient to erode even the hardest metals and to generate light emissions visible to the naked eye. In this talk we describe cavitation bubble cloud experiments carried out in spherical resonators at ambient and acoustic pressures up to 30 MPa. Key to our system is the ability to nucleate with temporal and spatial

controls, which we achieve using dielectric breakdown in water from pulsed focused laser beams. Our observations show that the cloud dynamics are controlled by the repetitive emission of shock waves, which propagate outward from the inertial cloud collapse, reflect off of the sphere wall, and then converge on the resonator center. Shock convergence phenomena and light emission from compact cloud collapse will be discussed. [Work supported by the Impulse Devices, Inc.]

2:30

4pPA4. The effects of hydrostatic pressure on conditions in and near a collapsing cavitation bubble. Charles C. Church (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS), D. Felipe Gaitan, Yuri A. Pishchalnikov (Impulse Devices, Inc., Grass Valley, CA), and Thomas J. Matula (Univ. of Washington, Seattle, WA)

It has long been understood that the conditions within a collapsing cavitation bubble become more extreme as hydrostatic pressure increases, but quantification of these conditions requires estimating the temperature, pressure, and density of the plasma in the bubble, a difficult task. To provide this information, we conducted numerical simulations using the plasma physics hydrocode HYADES, a 1-D, three-temperature, Lagrangean hydrodynamics, and energy transport code. The contents of a bubble at the center of a sphere of water at 1–3000 bars were specified at time = 0, and the bubble was driven by a spherically converging pressure wave of various frequencies (2.5–26 kHz) and amplitudes (10–3000 bars). Results were obtained for temperature, pressure, and density within and immediately outside the bubble. Calculations for bubble radius and the velocity and amplitude of the radiated shock wave compared well with experimental measurements at modest hydrostatic pressures (1–300 bars). At higher pressures, the maximum temperature within the bubble increases above 100 eV, the shock amplitude becomes greater than 1 Gbar, and its propagation speed is up to Mach 100. Also, the shock front heats the fluid, stimulating photon emissions in the liquid [Impulse ACPT contract No. W9113M-07-C-0178.]

2:50

4pPA5. Simultaneous measurements of shock waves, sonoluminescence flashes, and high-speed video of cavitation in high-quality factor resonators at high static pressures. Yuri A. Pishchalnikov, D. Felipe Gaitan, Mark S. Einert (Impulse Devices, Inc., 13366 Grass Valley Ave., Grass Valley, CA 95945, yuri@impulsedevices.com), R. Glynn Holt (Boston Univ., Boston, MA), Charles C. Church (Univ. MS, Oxford, MS), and Lawrence A. Crum (Univ. Washington, Seattle, WA)

Violent cavitation activity has been observed in water under hundreds of bars static pressure using Impulse Devices spherical resonators. To better understand the extreme conditions inside and in the immediate vicinity of the collapsing bubbles, we simultaneously recorded multi-frame shadowgraph images, acoustic pressure, and sonoluminescence (SL) flashes from the bubbles. Images of bubbles and shock waves were captured using a V710 Phantom high-speed camera (400 000 frames/s). The tip of a fiber-optic probe hydrophone was positioned in the field of view of the camera to correlate acoustic pressure with shadowgraph images of shock waves and bubble dynamics. SL flashes were collected with two photomultiplier tubes (PMTs, Hamamatsu, 1-ns rise time). The PMTs had identical ultraviolet filters but different sensitivities to extend the dynamic range from a single to thousands of photons. Typically a single bubble was spontaneously nucleated at the center of the sphere. After the first collapse, the bubble reemerged as a cluster of bubbles. The relationships among the static pressure, driving acoustic amplitude, the maximum bubble size, SL flashes, and shock wave amplitude will be discussed and compared with numerical results obtained using the hydrocode HYADES. [Funded by Impulse Devices, Inc. ACPT contract W9113M-07-C-0178.]

3:10—3:30 Break

Contributed Papers

3:30

4pPA6. Characteristics of high-quality factor resonators operating at high static pressures. D. Felipe Gaitan, Joel Gutierrez, Yuri A. Pishchalnikov, Michael Coffin, and Henry Tardif (Impulse Devices, Inc., 13366 Grass Valley Ave., Grass Valley, CA 95945, gaitan@impulsedevices.com)

Cavitation collapse can generate intense concentrations of mechanical energy, sufficient to erode even the hardest metals and to generate light emissions visible to the naked eye, e.g., single bubble sonoluminescence. We describe a high-quality factor, spherical resonator capable of achieving acoustic cavitation at ambient pressures in excess of 30 MPa. In this presentation, the dynamics of the resonator and the cavitation in general will be discussed, e.g., driver configuration, quality factor, resonance modes, power and pressure amplitude threshold for cavitation, etc., as a function of static pressure, as well as other parameters. [Funded and directed by Impulse Devices, Inc., ACPT Contract W9113M-07-C-0178.]

3:45

4pPA7. Modeling time delay in clusters of interacting bubbles. Derek C. Thomas, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Modeling the dynamics of large clusters of interacting bubbles requires that the effects of fluid compressibility be taken into account. Compressibility manifests itself through radiation damping and bubble-bubble interactions due to time delays associated with the finite sound speed. The time delays convert the dynamical equations for interacting bubbles in an incompressible fluid from a system of nonlinear ordinary differential equations to one of delay differential equations (DDEs). Special care must be taken when integrating DDEs numerically to maintain acceptable bounds on errors. The dynamical equations determined to be most suitable for solving as DDEs were obtained using Hamiltonian mechanics [Ilinskii *et al.*, J. Acoust. Soc. Am. **121**, 786 (2007)]. These first-order differential equations were augmented to include time delays in the bubble interaction terms and then solved numerically using a sixth-order Runge–Kutta method with a continuous interpolant (DDE_SOLVER). The same equations were also solved without the time delays but with correction terms that account for mutual radiation damping. Comparison of the results reveals the importance of time delay in bubble-bubble interactions as a function of the size and density of a bubble cluster. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and NIH Grant No. DK070618.]

4:00

4pPA8. Shock-driven growth of bubble clouds. Phillip Anderson, A. Sampathkumar, and R. G. Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Laser-nucleated bubble clouds in a high-pressure spherical resonator have previously been reported to develop into tight clusters after many acoustic cycles. The formation of these organized clusters is largely driven by the reconvergence of shocks from earlier cloud collapses. Highly tempo-

rally (40 Mfps) and spatially (3 μm/pixel) resolved images reveal that the expansion of the shock-nucleated clusters is initially very fast (> 8 km/s), much faster than the shocks themselves. However, this explosive growth of the cluster does not begin until hundreds of nanoseconds after the shock passes, and so the cluster never surpasses the shock itself. The phase diagrams of the cluster and shocks are mapped out and the shock-driven nucleation is discussed. [Work supported by the Impulse Devices, Inc.]

4:15

4pPA9. Complex demodulation of broadband cavitation noise. Pascal Clark, Les Atlas (185 Stevens Way, Dept. of Elec. Eng., Univ. of Washington, Seattle, WA 98195), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI 02841)

Cavitation is a prominent source of ship noise resulting from the collapse of bubbles in the wake of rotating propeller blades. This same rotation also induces an audible rhythm, which is often modeled as a periodiclike modulator multiplying a broadband noise carrier signal. Conventional signal models restrict the modulator term to be non-negative and real-valued, accounting only for long-term energetic fluctuations in the time series data. However, recent empirical observations [Versluis *et al.*, Science, 2000] and explanatory simulations [Ida, Phys. Rev. (2009)] reveal that interactions between multiple bubbles can lead to phase inversions in the form of negative pressure spikes. In this context we propose a more general complex-valued modulator that can account for energetic as well as phase-coherence patterns in the signal model. Through the use of synthetic signals and ship data examples, we will briefly discuss issues of detection and maximum-likelihood estimation related to the new complex signal model. For the purpose of fostering scientific understanding, however, our primary focus will be motivation for and estimation of the significance of possible complex modulation in both modeled and observed cavitation noise. [This research was supported by the Office of Naval Research.]

4:30

4pPA10. Analysis of acoustic quasi-Gaussian beams and scattering by spheres at the focus of this type of beam. Philip L. Marston (Dept. Phys. and Astronomy, Washington State Univ., Pullman, WA 99164-2814)

A paraxial Gaussian beam is an approximation sometimes used to describe the focusing of ultrasonic waves. In this research an exact solution of the linear Helmholtz equation is examined that is similar to a Gaussian beam when weakly focused. This superposition has advantages for relatively tightly focused beams not describable with a paraxial approximation. The quasi-Gaussian beam considered is an appropriate superposition of zero-order Bessel beams [P. L. Marston (unpublished)]. This way of representing a focused beam has the additional attribute that an analytical result is known for the scattering by an isotropic sphere placed on the axis of a Bessel beam [P. L. Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)]. In the focused case the scattering by a sphere centered on the focal point of the quasi-Gaussian beam follows by superposition. In comparison with the usual case of plane-wave illumination, mode amplitudes in the scattering are modified by a function depending on a ratio of incomplete gamma functions and the size of the focus of the beam. The analysis is useful for predicting the level of increase in scattering when a sphere is placed in a strongly focused beam. [Work supported by ONR.]

THURSDAY AFTERNOON, 26 MAY 2011 GRAND BALLROOM C, 1:20 TO 4:35 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Psychophysical and Physiological Sensitivity to Interaural Level Differences

G. Christopher Stecker, Chair
Univ. of Washington, Dept. of Speech and Hearing Sciences, 1417 N.E. 42nd St., Seattle, WA 98105

Chair's Introduction—1:20

Invited Papers

1:25

4pPP1. Neural sensitivity to interaural level differences determines virtual acoustic space minimum audible angles for single neurons in the lateral superior olive. Daniel J. Tollin (Dept. of Physio., Univ. of Colorado Sch. of Medicine, 12800 E. 19th Ave., Aurora, CO 80045, daniel.tollin@ucdenver.edu)

The minimum audible angle (MAA), the smallest angle separating two sound sources that can be reliably discriminated, is a psychophysical measure of spatial acuity. In humans and cats, MAAs for tone and noise stimuli range from 1–5 deg. For high-frequency (>1.5 kHz) tones the predominant cue for azimuth is the interaural level difference (ILD). Neurophysiologically, ILDs are first encoded in the lateral superior olive (LSO). Here, we examined the ability of LSO neurons in cats to signal changes in the azimuth of noise sources. Using measurements of head related transfer functions, the virtual acoustic space technique was used to manipulate source azimuth in the physiological experiments. For each neuron detection theory was used to compute the smallest increment in azimuth necessary to discriminate that change based on discharge rate and response variability. Minimum neural MAAs were 2.3 deg for midline sources (median = 4.5 deg, n = 32 neurons). The good neural acuity for spatial location will be explained in terms of the changes in the acoustic ILD cue with changes in source location along with the underlying sensitivity of the LSO neurons to the ILD cue. LSO neurons can signal changes in sound azimuth that match or exceeded behavioral capabilities. [NIH DC006865.]

1:45

4pPP2. Empirical and theoretical findings concerning how interaural intensitive disparities affect lateralization for high- versus low -frequency stimuli. Leslie Bernstein and Constantine Trahiotis (Dept. of Neurosci. and Surgery (Otolaryngol.), Univ. of Connecticut Health Ctr., 263 Farmington Ave., Farmington, CT 06030)

This presentation concerns empirical and theoretical findings concerning the potency of interaural intensitive disparities (IIDs) at low versus high spectral frequencies. In one context, an acoustic pointing task was used to measure how IIDs affect extents of laterality. Laterality was measured for 4-kHz-centered stimuli with reinforcing or opposing combinations of IIDs and interaural temporal disparities (ITDs). The goal was to assess whether IIDs act as scalars or "weights" within the putative "binaural display" at high spectral frequencies (where the envelopes convey ITD-information) as they have been repeatedly shown to operate at low spectral frequencies (where the waveforms convey ITD-information). They do. The data were accounted for via an augmentation of the cross-correlation-based "position-variable" modeling approach developed by Stern and Shear [J. Acoust. Soc. Am. **100**, 2278–2288 (1996)] to account for ITD-based lateralization at low spectral frequencies. In a second context, laterality produced by IIDs (ITD=0) was measured for stimuli centered at either 4 kHz or 500 Hz. The novel finding was that IIDs consistently produced larger extents of laterality at 4 kHz than they did at 500 Hz. [Work supported by research Grant No. NIH DC-04147 from the National Institute on Deafness and Other Communication Disorders, National Institutes of Health.]

2:05

4pPP3. Physiology of the interaction between intensity and time at monaural and binaural stages. Philip X. Joris (Lab. of Auditory Neurophysiol., Univ. of Leuven, Herestraat 49 Bus 1021, BE-3000 Leuven, Belgium)

The two cues for azimuthal sound localization, interaural level differences (ILDs) and interaural time differences (ITDs), interact perceptually. According to the latency hypothesis, this interaction arises peripherally: Increases in intensity for the ear nearest to a sound source cause a decrease in response latency relative to the other ear. ILDs would thus shift the pattern of activation in an array of coincidence detectors that processes ITDs. Psychophysical studies have accumulated evidence against the latency hypothesis, but it has received support in several physiological studies that studied transient responses in a variety of binaural structures. We examined the basic premise of peripheral time-intensity trading in the auditory nerve with a coincidence analysis of responses to broadband noise. We found that changes in intensity cause small but systematic shifts in the ongoing timing of responses, generally but not always resulting in shorter delays between stimulus onset and neural response for increasing intensity. Overall, the results show that ongoing timing is remarkably invariant with intensity at the most peripheral neural level. The implication of temporal coding in monaural pathways for binaural ILD processing will be discussed in a wider context. [Supported by the Fund for Scientific Research—Flanders and Research Fund K.U. Leuven.]

2:25

4pPP4. Interaural level differences: Diffraction and localization by human listeners. William M. Hartmann (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824) and Brad Rakerd (Michigan State Univ., East Lansing, MI 48824)

Because long-wavelength tones are diffracted around the head, interaural level differences (ILDs) are thought to be too small to provide useful localization cues at low frequency. ILDs are unquestionably useful at high frequency. The boundary between low and high for human listeners has been set as high as 3000 Hz [Stevens, and Newman, Am. J. Psych. **48**, 297 (1936)]. In fact, however, the frequency range for useful ILD cues depends on the azimuth of the source. ILDs are physically present at 500 Hz and below but they are sensitive to changes in azimuth only near the forward direction. As the frequency increases to 1000 Hz, ILDs clearly play an essential role in sound localization, but the non-monotonic azimuth dependence caused by an approach to the Poisson/Arago bright spot leads to confusion for azimuths greater than 40 or 50 deg. At 2000–3000 Hz, the non-monotonic bright-spot structure moves out to larger azimuths, and ILDs become a mostly monotonic function of azimuth for an ever larger range of azimuths. Above 3000–4000 Hz, the bright spot affects ILDs only at the largest azimuths, but there can be idiosyncratic structure at smaller azimuths. [Work supported by the NIDCD, Grant No. DC 00181.]

2:45—2:55 Break

2:55

4pPP5. Processes for interaural time and level differences: To what extent are they independent and where are they integrated? Shigeto Furukawa (NTT Commun. Sci. Labs., NTT Corp., Atsugi, Kanagawa 243-0198, Japan, shig@avg.brl.ntt.co.jp)

Two psychophysical experiments examined the degree to which the processes for interaural and level differences (ITD and ILD) are independent and the central-processing stage at which the information about the two cues is combined. The first experiment measured the detectability of individual or simultaneous changes of ITD and ILD. The results showed greater interaction of the two cues for high-frequency stimuli ("transposed stimuli" centered at 4 kHz) than for low-frequency stimuli (tones at 125 or 500 Hz). An analysis adopting the framework of signal detection theory indicated that the two cues are processed by partially independent "channels" at low frequencies. Focusing on low-frequency stimuli, the second experiment measured the detectability of simultaneously modulated ITD and ILD, with a varying rate and phase relationship between the modulations. Although the pattern of the results varied among listeners, there was evidence that the detectability depends on the relative phase even at a modulation rate as high as 100 Hz. This indicates that a cue integration stage in the auditory system, in which ITD and ILD interact additively or subtractively, is located before a stage where temporal modulation information is lost, which could be the source of the "binaural sluggishness."

3:15

4pPP6. Temporal weighting in binaural hearing: Distinct contributions of interaural time and level differences following sound onset. G. Christopher Stecker and Andrew D. Brown (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, cstecker@uw.edu)

The ability of a listener to make use of interaural level differences (ILDs) or envelope interaural time differences (ITDs) carried by modulated high-frequency sounds depends critically on the modulation period. Specifically, periods shorter than 4–5 ms substantially reduce the availability of ongoing cues, with the consequence that localization and discrimination increasingly rely on onset cues as the period decreases. Over the past several years, the authors have studied this dependence by measuring listeners' detection of dynamic ITD and ILD, their temporal weighting of both cues during localization tasks, and their perception of simulated sources and echoes (i.e., precedence effects) carrying one or both cues. This presentation will review the results of those studies, which suggest that (a) onsets dominate processing of both cues at high rates; (b) postonset ILD sensitivity is better maintained, compared to postonset ITD, at high rates; (c) sensitivity to postonset ILD, but typically not ITD, exhibits a temporally increasing profile consistent with binaural temporal integration of that cue; and (d) maintained sensitivity to postonset ILD allows changes in that cue, but not ITD, to cause "breakdown" of precedence effects. [Work supported by NIH Grant Nos. R03 DC009482 and F31 DC010543, and NSF Grant No. IOB-0630338.]

Contributed Papers

3:35

4pPP7. Attentional tracking in real-room reverberation. Simon J. Makin, Anthony J. Watkins, and Andrew P. Raimond (Sch. of Psych. and Clinical Lang. Sci., Univ. of Reading, Harry Pitt Bldg., Reading, RG66AL, UK, s.j.makin@reading.ac.uk)

A listener can attend to one talker in a mixture of sound sources, and a difference in bearing between "target" and interfering sources is thought to aid this "tracking." Interaural cues are degraded by reverberation however, raising the question of what cues are actually used to track sources in rooms? Listeners can also use differences in voice characteristics, so this study asks how well cues arising from location compete with talker differences when tracking messages in real-room reverberation. Real room measurements of binaural room impulse responses (BRIRs) were used to spatialize the stimuli and listeners decided which of two simultaneous target words belonged in a target phrase played simultaneously with a "distracter" phrase. Location differences were in competition with talker differences so that listeners' responses indicated which cue-type they were tracking. Further experiments used processed BRIRs to eliminate temporal cues. Location was dominant over talker difference in dichotic but not diotic conditions, and results with processed BRIRs indicated that this "binaural advantage" is almost entirely due to ILD. Furthermore, an analysis of spectral distances between BRIR channels revealed that the probability of location difference overriding talker difference correlates with the distance between ILD patterns but not with monaural spectral distances.

3:50

4pPP8. Effects of dynamically changing interaural level differences brought about by amplitude compression. Ian M Wiggins and Bernhard U Seeber (MRC Inst. of Hearing Res., Nottingham NG7 2RD, England, ian@ihr.mrc.ac.uk)

Amplitude compression is a feature of most hearing aids and cochlear implant processors. When compression acts independently at each ear in a bilateral fitting, interaural level differences are altered dynamically, potentially affecting spatial perception. A lateralization task was used to measure the position of sounds processed with a simulation of hearing-aid compression. Normal-hearing listeners indicated the leftmost and rightmost extents of the sound image(s) and selected from three response options according to whether they heard (1) a single, stationary image; (2) a moving image; or (3) a split image. Fast-acting compression applied at high frequencies significantly affected the perceived position of sounds. For sounds with abrupt onsets and offsets, compression shifted the entire image to a more central location. For sounds containing gradual onsets and offsets, including speech, compression shifted only the innermost extent toward the center, resulting in a wide separation between the leftmost and rightmost extents. In such cases, compression increased the occurrence of moving and split images by up to 57 percentage points. The severity of the effects was reduced when undisturbed low-frequency binaural cues were available to listeners, indicating the importance of preserving these cues in bilaterally fitted hearing devices.

4:05

4pPP9. The contribution of intrinsic amplitude modulation to the precedence effect at high frequencies. Bernhard U. Seeber (MRC Inst. of Hearing Res., Sci. Rd., University Park, Nottingham NG7 2RD, United Kingdom, seeber@ihr.mrc.ac.uk)

The precedence effect (PE) demonstrates our ability to locate sounds correctly at the source despite the presence of interfering sound reflections. It was shown to function with broadband noises of long duration even when lead and lag had simultaneous onsets, i.e., when information was restricted to the ongoing sound part. In a localization dominance task participants indicated the perceived location of lead-lag stimuli played from loudspeakers in an anechoic chamber. Stimuli were harmonic complex tones (HCTs) and Gaussian noise bandlimited to 2500–5500 Hz. Lead dominance existed for all stimuli despite lead and lag overlapping in time and having simultaneous onsets. This demonstrates that information from the ongoing sound part alone can evoke the PE at high frequencies. The amount of intrinsic modulation affected echo thresholds (ETs), which were slightly larger for HCTs with Schroeder positive than negative phase. Localization dominance was weak for the noise; ETs were short and a broad image was frequently reported. This indicates that the fast amplitude modulation inherent in HCTs is crucial for the PE at high frequencies, while the shallow modulation depth of the noise was likely detrimental. The PE should thus be possible for selected sounds with cochlear implants, which encode envelope information.

4:20

4pPP10. Quantifying dynamic interaural level differences introduced in modulated electrical stimuli. Matthew J. Goupell, Alan Kan, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI 53705)

Clinical mapping of bilateral cochlear implants (CIs) is often performed by measuring individual single-electrode currents that elicit threshold and comfortable loudness. This is done one ear at a time, where considerations for simultaneous bilateral stimulation are often disregarded. In contrast, synchronized bilateral stimulation of CIs in controlled research experiments often use constant-amplitude pulse trains to measure sensitivity to interaural level differences (ILDs) and interaural time differences. This is typically done using a single pair of subjectively pitch-matched electrodes, where the amplitudes of the electrical pulses elicit a near-comfortable loudness and produce a binaurally centered auditory image. Current clinical mapping of the processors do not consider possible ILDs introduced across the ears during simultaneous bilateral stimulation. Additionally, because loudness growth curves are often extremely variable across electrodes and ears, dynamic ILDs will possibly be introduced for modulated stimuli, like speech signals. Data will be shown that quantify the ILDs introduced by a typical bilateral clinical mapping, along with incoherence detection and binaural masking level difference threshold measurements, all in bilateral cochlear-implant listeners. Results will be considered in the context of theories regarding binaural processing and potential applications to clinical mapping strategies. [Support provided by the NIH Grant Nos. K99-DC010206 (J.G.) and R01-DC003083 (R.Y.L.).]

Session 4pSC

Speech Communication: Perception and Production in Children and Infants (Poster Session)

Melissa A. Redford, Chair
Univ. of Oregon, Dept. of Linguistics, Eugene, OR 97402-1290

Contributed Papers

All posters will be on display from 1:00 p.m. to 3:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m. and contributors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

4pSC1. Stop consonant acoustics in children with fetal alcohol syndrome. Christopher Bolinger and James Dembowski (Dept. of Speech Lang. Hearing Sci., Texas Tech Univ. Health Sci. Ctr., 3601 4th St., Lubbock, TX 79430-6073, james.dembowski@ttuhsc.edu)

Speech of children with fetal alcohol syndrome (FAS) has been little studied compared to language. Becker *et al.* (1990) found a relationship between prenatal alcohol exposure, oral motor control, and speech articulation. The current presentation is part of a larger study attempting to further examine the relationship between speech and motor control in children with FAS. Behavioral tests suggest deficits in focal oral motor control specific to children with FAS [Bolinger and Dembowski (2010)]. The current project attempts to extend this investigation through acoustic measures. Peak and mean frequencies of stop consonant releases were used to infer control of place of articulation. Voice onset time (VOT) was used to infer articulatory-laryngeal coordination. Preliminary measures on three experimental speakers and two matched neurotypical controls suggest higher stop consonant frequencies in the experimental group, with a poorer distinction between alveolar and velar stops than in the control group. Voiced VOT values were significantly longer for FAS children than for controls. Mean voiceless VOTs were similar across groups, but substantially more variable for the FAS children. Values may be interpreted as acoustic evidence for specific speech motor control deficits in FAS children relative to matched neurotypical children.

4pSC2. Getting the timing right: Meaningful variation in children's verbal response times. Marisa A. Tice (Dept. of Linguist., Margaret Jacks Hall (460-113), Stanford Univ., Stanford, CA 94305-2150, middyp@stanford.edu)

Adults take turns in conversation with minimal gaps between speakers and almost no vocal overlap. Good conversational timing is a crucial skill all speakers must develop to enter into efficient and successful conversation. Children begin taking turns in conversation from infancy, supported heavily by their caregivers. Research on the acquisition of conversational timing, however, has been limited in scope, overlooking longitudinal data, parent-child interactions, and processing factors contributing to children's response latencies. This study addresses these issues directly for question-answer pairs over the course of ages 1–3. Several factors were found to affect children's response latencies in replying to their mothers' questions, including interlocutor, age, question function, and answer complexity. In contrast to previous findings, children's timing is, on average, nearly aligned with adult averages before the age of 4. Their timing is *at* adult ranges for certain early-mastered question functions and answer types. These effects interact with the child's age, and the developmental trajectory hinges on both changes in the child's response abilities and also in the kinds of questions the parent asks. Children's timing and turn-taking abilities are more advanced than has been previously assumed, suggesting that children master turn-projection skills and pragmatically relevant factors early on.

4pSC3. Factor analyzes of critical-band-filtered infant babbling. Yoshitaka Nakajima (Dept. of Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, nakajima@design.kyushu-u.ac.jp), Yoko Shimada (Kyoto Univ., Kyoto 606-8501, Japan), Hirotoshi Motomura, Kazuo Ueda, and Takeharu Seno (Kyushu Univ., Fukuoka 815-8540, Japan)

In order to investigate the development of human speech communication, babbling of Japanese infants was analyzed through 19 critical band filters below 7 kHz. Smoothed power fluctuations derived from these filters were submitted to factor analyzes, i.e., principal component analyzes followed by varimax rotation. The first two factors explained about 25% of variance. One of these factors appeared in a stable manner in a frequency range above 2500 Hz. This factor is different from any factors obtained in similar analyzes of adult speech in which 500, 1700, and 3300 Hz were typical boundaries to separate factors across eight different languages/dialects including Japanese.

4pSC4. Oral receptive and expressive language skills and word recognition in children with cochlear implants. Rachael Gilbert (Linguist., Univ. of Texas–Austin, 1 University Station B5100, Austin, TX 78712-0198), Douglas Sladen, and Rajka Smiljanic (Univ. of Texas–Austin, Austin, TX 78712)

Compared to children with normal hearing, children with cochlear implants (CIs) have a higher degree of difficulty performing speech recognition in noise [Litovsky *et al.*, (2004); Sc. Thibodeau, (2006)]. The combined contribution of the peripheral-auditory and the cognitive-linguistic skills that underlie this difficulty is not well understood. This study examines the relationship between oral language abilities and the effects of phonetic and semantic enhancements on speech perception in children with and without cochlear implants. Specifically, we compare their performance on the oral and written language scale (OWLS), oral receptive and oral expressive tests, with the results of a sentence-in-noise perception test. The sentence in-noise-test was composed of 120 sentences in which the final words varied in predictability (i.e., high versus low semantic context) and which were produced in conversational and clear speech by one female and one male talker. The relationship between the OWLS scores and word recognition in noise was investigated for ten children with CI and ten children with hearing between the ages of 6 and 13. This research will further our understanding of the lower-level sensory and higher-level cognitive factors that affect speech understanding in children.

4pSC5. Longitudinal formant analysis of Mandarin-learning children before 7 years of age. Li-mei Chen (Dept. of Foreign Lang. and Lit., Natl. Cheng Kung Univ., 1 University Rd., Tainan City, Taiwan)

The present study is the seventh year of a longitudinal observation of vowel production in two Mandarin-learning children. In addition to gender differences, both subjects at 7 years of age experienced a slow growth stage after a rapid growth period in previous stages. Major findings are (1) girl subject used vowels with nasal endings more frequently than boy subject; (2) decrease of F1 F2 values is not so evident as in the previous stages in boy subject. As to girl subject, up to 7 years of age, no obvious change in formant values was found; (3) F1 values are more stable than F2 values in

both subjects, especially in corner vowels. They appeared to acquire jaw movement sooner than tongue movement; (4) the earlier trend of shrinkage in F1-F2 vowel area was not found at this stage, and extension of vowel space was even found in both subjects; (5) there is no obvious trend of f0 decline in both subjects. Longitudinal data of vowel formant values from the same group of subjects provide important references for assessment and treatment of articulation disorders in children. [This investigation was supported through funds from National Science Council in Taiwan (NSC 96-2411-H-006 -023).]

4pSC6. Patterns of development in stop-release cues to coda contrasts in American-English-speaking 2-year-olds. Jae Yung Song (Dept. of Commun. Sci. and Disord., Univ. of Wisconsin-Milwaukee, Milwaukee, WI 53201, songjy@uwm.edu), Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA 02139), and Katherine Demuth (Macquarie Univ., Sydney, New South Wales 2109, Australia)

A distinctive characteristic of early speech is "exaggerated" release of stop consonants including multiple bursts and heavy post-release noise [Imbrie (2005)]. This study provides an in-depth investigation of changes in the acoustic characteristics of stop coda release from 1;6 to 2;6 years, focusing on their interaction with emerging linguistic contrasts. We examined how various non-spectral coda-release-related cues, such as the presence and duration of post-release noise, varied with coda voicing and place (alveolar versus velar) in spontaneous speech from six American-English-speaking mother-child dyads. Starting as early as 1;6, children produced many of the same types of correlates of voicing and place as adults did, but they exhibited more frequent occurrences and longer durations of these cues compared to mothers; this exaggeration decreased by 2;6. In utterance-medial position, where the decrease was particularly striking, it occurred in a selective, stepwise fashion rather than simultaneously in all contexts. That is, the decrease was larger for voiceless codas than for voiced codas, and for place contrasts, the values decreased first for velars, and only later for alveolars. This observation raises the question of whether such selective fine-tuning is characteristic of phonological development across languages, and if so, what the underlying mechanisms might be. [Work supported by NIH R01HD057606.]

4pSC7. The development of neutral tone in Mandarin-speaking children. Jie Yang and Barbara Davis (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712)

Besides the four citation tones in Mandarin stressed syllables, neutral tone usually occurs in unstressed syllables. The fundamental frequency ($F0$) contour and height of neutral tone are determined by the preceding tones. Neutral tone was also considered to have lower intensity and shorter duration compared to citation tones [Cao (1986); Chao (1968)]. Li and Thompson [(1977)] suggested that neutral tone was not fully acquired by Mandarin-speaking children by age 2. However, the acoustic characteristics of neutral tone in the extended period of acquisition were not explored. The present study compared acoustic realizations of neutral tone in children's production with adults. Eight 5-year-old and eight 8-year-old monolingual Mandarin-speaking children and young adults participated. Bisyllabic target words containing neutral tone in the second syllable were elicited by picture-naming tasks. $F0$, duration, and intensity of neutral tone syllables were measured. The ratios of these acoustic parameters between the first and second syllables were calculated. Results indicated that 5-year-old children started to produce $F0$ contour and height of neutral tone according to the preceding tone. But they still showed significant differences in the above-mentioned measures compared to 8-year-olds and adults.

4pSC8. Longitudinal changes of raw and normalized vowel space in children with cerebral palsy. Jimin Lee, Gary Weismer, and Katherine Hustad (Waisman Ctr. and Dept. of Communicative Disord., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, jiminlee @pitt.edu)

The current study compares longitudinal changes of raw and normalized vowel space in 22 children with cerebral palsy in four different severity groups and over four time sampling points spanning a total of 18 months. 13 children had a clinical diagnosis of dysarthria and 9 children had CP, but no diagnosed speech disorders. The average chronological age of children at the first time sampling point was 50 months ($SD= 9.8$). Severity was operationally defined by orthographic transcription-based word intelligibility scores obtained from 176 listeners. Nearey's normalization formula was employed to calculate normalized vowel space. Results showed both similarities and differences in longitudinal change patterns of raw and normalized vowel space in children with cerebral palsy. The difference between the two vowel space measures was most noticeable in children without speech disorders. Results will be discussed in terms of application and interpretation of normalization procedures in longitudinal vowel space data in pediatric population, comparison between raw and normalized vowel space measures, and comparison with previously reported data on developmental vowel space.

4pSC9. Acoustic and articulatory characteristics of clinically resistant /r/. Suzanne E. Boyce, Sandra Combs, and Ahmed Rivera-Campos (Dept. of Comm. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH 45267-0379, suzanne.boyce@uc.edu)

Acoustically, there is strong evidence to suggest that achievement of an appropriately low third formant for American English /r/ depends on successful simultaneous production of two vocal tract constrictions by the tongueone constriction by tongue tip or dorsum at the palate and one constriction by tongue root in the pharynx [Tiede *et al.* (2004); McGowan *et al.* (2003); Boyce (2008)]. The timing and extent of movement for these constrictions also appear to differ according to pre- versus postvocalic word position. In a study of older children with persistent difficulty producing /r/, clinic notes at their initial clinic visit indicate that two-thirds were missing a pharyngeal constriction during /r/ for at least one of these prosodic conditions. Clinic notes also indicate that improvement in /r/ production was accompanied by achievement of simultaneous palatal and pharyngeal constrictions. In this paper, we report an analysis of the correlation between these subjective clinical observations and more objective ultrasound imaging and acoustic data collected simultaneously on the same subjects during therapy for /r/ production.

4pSC10. Children's perception of clear-speech vowels. Dorothy Leone (Iona College, 715 North Ave., New Rochelle, NY 10801), Sih-Chiao Hsu, Miriam Baigorri, Gemma Moya-Gale, and Erika Levy (Teachers College, Columbia Univ., 525 West 120 St., Box 180, New York, NY 10027)

Clear speech is an intelligibility-enhancing mode of speech that increases identification accuracy in various listening populations [e.g., Picheny *et al.* (1986)]. Children identify words more accurately in clear-speech sentences in noise than in conversational-speech sentences in noise [Bradlow *et al.* (2003)]. At the segmental level, adults identify clear-speech vowels in noise significantly more accurately than conversational vowels in noise, with some vowels benefiting more from clear speech than others [Ferguson and Kewley-Port (2002)]. However, little is known about the clear speech intelligibility benefit for vowels in noise as perceived by children. This study examined children's identification of clear-speech vowels in noise. Four female adult speakers produced (/i-ɛ-æ-ʌ/) in carrier phrases in conversational and clear speech. Stimuli were then presented in speech-shaped noise to school-aged children at two signal-to-noise ratios. Children's repetition of key words were recorded and coded by naive adult listeners. Results revealed children's poorer vowel identification with decreased signal-to-noise ratio, and a greater clear-speech advantage in the poorer SNR, although listener-variability was evident. These data provide preliminary support for adults' use of clear speech when communicating with children in adverse listening conditions.

4pSC11. A cross-linguistic developmental study of vowel spectral movement patterns. Hyunju Chung (Dept. of Communicative Disord., Univ. of Wisconsin-Madison, Goodnight Hall, 1975 Willow Dr., Madison, WI 53706, hchung23@wisc.edu), Eun Jong Kong, Jan Edwards, and Gary Weismer (Univ. of Wisconsin-Madison, Goodnight Hall, 1975 Willow Dr., Madison, WI 53706)

Most studies of cross-linguistic differences in vowel production have focused on static formant values. However, Chung, *et al.* (in press) found systematic differences in spectral movement patterns between vowels produced by adult American English and Korean speakers. American English vowels showed greater magnitude and more consistent direction of movement, as compared to Korean vowels. The purpose of this study was to investigate the development pattern of these cross-linguistic differences. The first two formant frequency values of eight English vowels (/ɑ/, /e/, /ɛ/, /i/, /ɪ/, /o/, /u/,

and /ʊ/) and five Greek and Korean vowels (/a/, /e/, /i/, /o/ and /u/) from word-initial fricative-vowel sequences produced by ten adults and twenty children (2-year-olds and 5-year-olds) were extracted. The duration of vowels was normalized to have seven different proportional time points. In order to minimize the effect coming from the preceding or following consonants, only the formant frequency values measured from the second time point to the vowel midpoint were used for the subsequent analysis. Results will be discussed with respect to the language-specificity in the vowel spectral movements and possible gestural differences across age groups. [Work supported by a Fulbright fellowship, NIDCD Grant No. 02932 and NSF Grant No. 0729140.]

4pSC12. Physiological constraints explain order of Mandarin tone acquisition in 3-year-old children. Puisan Wong (Dept. of Otolaryngol.—Head and Neck Surgery, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212)

Careful analyzes of childrens' lexical tones revealed a more protracted developmental course than previously described. This study examined reasons for that late acquisition by performing acoustic analyzes on 289 monosyllabic Mandarin tones that were produced by 13 children and 4 adults and, subsequently, judged by 10 native listeners. Eight acoustic parameters that yielded strong correlations with the judges categorization of the tones were compared among adult correct productions, child correct productions, and child incorrect productions. Results revealed that childrens' tone 1 (T1, high level) errors involved reduced $f0$ height and inability to sustain a level $f0$. Childrens' tone 2 (T2, rising) errors reached minimum $f0$ later in the syllable and had either reduced rising or falling $f0$ slopes. Childrens' tone 3 (T3, dipping) errors involved reduced syllable length, failing to reach a low $f0$, and having a much higher mean $f0$. Childrens' incorrect tone 4 (T4, falling) productions had reduced negative $f0$ slopes. Even childrens' correctly identified T1, T2, and T3 productions were not adultlike. The order of the four tones from the most to the least adultlike was T4, T2, T1, and T3, corresponding to the order of ease of speech motor control for the production of the tones.

4pSC13. A test of formant frequency analyzes with simulated child-like vowels. Kate Bunton and Brad H. Story (Dept. Speech, Lang., and Hear. Sci., Univ. of Arizona, Tucson, AZ)

Speech production by children is typically characterized by a fairly high fundamental frequency of phonation and a short vocal tract length that produces high formant frequencies. Together these two characteristics contribute to the difficulty of making accurate measurements of the formants because the vocal tract transfer function may be undersampled by the voice source harmonics. In addition, the close proximity of the low-numbered harmonics (including the fundamental) to a formant may lead to strong nonlinear interaction of the acoustic pressures in the vocal tract and the glottal airflow. The purpose of this study was to use standard spectrographic and LPC techniques, as well as new pitch-synchronous method, to measure formant frequencies of child-like vowels that have been simulated with a speech production model. Each vowel will be simulated with two different representations of the voice source: a glottal area model that allows for nonlinear source-tract interaction and a glottal flow model in which vocal tract characteristics cannot affect the source. The results of the analyzes will be compared to the actual formant frequencies calculated directly from the known vocal tract area functions used to generate the vowels. [Research supported by NIH R01-DC04789.]

4pSC14. Repetition of words from dense and sparse phonological neighborhoods in children with hearing loss and normal hearing. Mark VanDam (Boys Town Natl. Res. Hospital, 555 N 30 St., Omaha, NE 68131, mark.vandam@boystown.org), Noah H. Silbert (Univ. of Maryland, College Park, MD 20742, nsilbert@umd.edu), and Mary Pat Moeller (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

4 and 7 year-old children with normal and impaired hearing performed a listen-and-repeat task with words from dense and sparse phonological neighborhoods. Response accuracy was measured as a function of age, hearing loss, and neighborhood density. Accuracy was higher for older children, children with normal hearing, and for words from dense phonological neighborhoods. Age was evaluated cross-sectionally. Older children with normal hearing were consistently more accurate than younger children with normal hearing; children with hearing loss showed much smaller age-related differences. For children with normal hearing, accuracy for sparse neighborhoods was significantly higher for 7 year-olds than for 4 year-olds, but no such effect was observed for children with hearing loss; a similar pattern holds for words from dense neighborhoods. Relationships between accuracy, productive phonology (GFTA-II), expressive vocabulary (EVT-II), and degree of hearing loss (PTA dBHL) are also explored. These results suggest that phonological predictability, as measured by neighborhood density, is an important factor in the ability of children to perceive and reproduce words, interacting with both age and presence or absence of hearing loss. Absence of age-related performance improvement for children with hearing loss has theoretical implications for the role of auditory experience in development. [Work supported by NIH/NIDCD Grant Nos. R01DC006681 and P30DC04662.]

4pSC15. A nonword is a word is a word: Perceptual evidence for neighborhood density effects in preschool-aged children. Melinda D. Woodley (Linguist. Dept., Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720)

Work with infants has suggested that children as young as 9 months are sensitive to phonotactic probability [Jusczyk *et al.* (1994)], yet there is little direct evidence for phonotactic facilitation effects in preschoolers [Munson *et al.* (2005)]. One possible explanation for this discrepancy is that as infants begin to assign meaning to lexical items, low level phonetic processing is essentially discontinued until competition among lexical items necessitates finer grained phonological representations [Werker and Stager (2000), among others]. However, another possible reason is methodological; work with toddlers has almost exclusively focused on production tasks. Adults clearly exhibit phonotactic facilitation effects in production [Vitevitch *et al.* (2004)], but any processing effects in children could be masked by noisiness in motor command implementation. The present study therefore examines the effects of phonotactic probability/neighborhood density on lexical access in preschoolers using a purely perceptual ("same/different") task. Whereas adults exhibit a lexical competition effect for words and a phonotactic facilitation effect for nonwords in speech perception [Vitevitch and Luce (1999)], the present results demonstrate a significant lexical competition effect for both words and nonwords, suggesting that the lexical level is indeed the primary mode of processing for young children.

Session 4pSPa

Signal Processing in Acoustics and Underwater Acoustics: Detection, Localization, and Noise

Colin W. Jemmott, Chair
Applied Operations Research, 420 Stevens Ave., Solana Beach, CA 92075

Contributed Papers

1:30

4pSPa1. Improved signal detection in non-Gaussian deep ocean noise. Colin W. Jemmott (Appl. Operations Res., 420 Stevens Ave., Ste. 230, Solana Beach, CA 92075, colin.jemmott@appliedor.com), David R. Barclay, and Michael J. Buckingham (Univ. of California, San Diego, La Jolla, CA 92093-0238)

DeepSound is an autonomous, high-bandwidth acoustic recording system designed to profile ambient noise to depths of 9 km. Recent ambient noise measurements recorded by the DeepSound probe well below the conjugate depth of the channel exhibit significant non-Gaussianity (the hypothesis of Gaussianity is rejected for 97% of 5 s samples using the large sample size corrected Anderson–Darling test at a 95% confidence level). A common sonar signal processing task is the automatic detection of a signal corrupted by ambient ocean noise. The standard solution is matched filtering, which is the optimal detector in the maximum likelihood sense assuming additive white Gaussian noise. Because matched filter is derived under the assumption of Gaussian noise, its performance suffers when implemented in real world non-Gaussian noise. This paper describes a signal detection algorithm for non-Gaussian noise based on the generalized Gaussian probability density function. The generalized Gaussian detector is locally optimal for small SNR, and the processing chain resembles a matched filter with a non-linear preprocessing stage. The detection performance of the generalized Gaussian detector is shown to exceed that of matched filtering for synthetic signals injected into the noise measurements recorded by the DeepSea probe. [Work supported by ONR.]

1:45

4pSPa2. Bayesian histogram filter localizer performance in a multiple contact, continental shelf environment. Alexander W. Sell and R. Lee Culver (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802)

Model-based recursive Bayesian state estimation provides a straight forward way of including *a priori* environmental knowledge in a localization framework. Previous work by Jemmott *et al.* demonstrates successful range and depth localization of an acoustic source using this method and data from a single hydrophone. However, that work utilized a relatively benign environment from the SWellEx 96 experiment in which only two source signals were present. This talk will extend the work of Jemmott *et al.* and discuss the performance of the localization scheme in a multiple contact, continental shelf environment. [Work supported by ONR Undersea Signal Processing.]

2:00

4pSPa3. Cramer–Rao bounds for vector sensors. Arthur B. Baggeroer (MIT, 77 Massachusetts Ave., Rm. 5-206, Cambridge, MA 02139)

New formulations of the Cramer–Rao bounds for vector sensors are presented. They have the following advantages compared to previous expressions: (i) the formulation is simple and easy to evaluate involving the calculation of just three terms, (ii) arbitrary Green's functions connecting source and observer can be used instead of plane waves, (iii) correlated noise vice white noise is easily introduced permitting the evaluation of the impact of interference, (iv) passive and active problems can be addressed, and (v) tomographic parameters can be introduced. In this presentation, these results are applied to source localization using towed arrays of vector sensors. Specific results include (i) the performance improvement of a vector array compared to a pressure array, (ii) comparison of vector arrays to twin line arrays, and (iii) the performance limits on a vector array to resolve right/left ambiguities and mitigate backlobe interference. [Work supported on ONR Code 321, Sonar Signal Processing.]

2:15

4pSPa4. The influence of ambient field directionality on wave coherence. Shane Walker and W. A. Kuperman (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093)

The directionality of the ambient noise field influences the expectation value of the wave coherence measured between a pair of spatially separated sensors. It is demonstrated that diffraction acts to suppress the coherence associated with wave components that do not causally visit both sensors. Conversely, wave components that do causally pass between the sensors are not suppressed. As it is the latter that carries information about the point-to-point propagation of waves between the sensor locations, the tomographic relationship between the measured wave coherence and sensor-to-sensor propagation is robust to even highly directional features of the ambient field. While this robustness is helpful for tomographic extraction applications (with a caveat on the required observation time), it can prove detrimental to applications designed to characterize the ambient field distribution. Practical experimental implications are discussed.

2:30

4pSPa5. Quantifying the emergence rate of spatio-temporal correlations of the ambient wave field from finite duration sample realizations. Shane Walker and W. A. Kuperman (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA)

The spatiotemporal correlations measured between receivers in a diffuse wave field convey information about the point-to-point propagation of waves between the receiver locations. An issue of practical experimental importance is how much data must be accumulated to generate a useful estimate of the expectation value of the spatiotemporal correlations. The uncertainty associated with a finite duration realization of the sample cross correlation function for an arbitrary environment is derived. Using this, a method for modeling the emergence rate of the expectation value of the spatiotemporal correlations is discussed. As examples, the cases of free space and the ocean waveguide are considered. Practical experimental implications are discussed.

2:45

4pSPa6. Impulse signal reconstruction using bi-receiver data. H. Z. Wang (College of Information Sci. and Eng., Ocean Univ. of China, 238 Songling Rd., Qingdao 266100, China, coolicejiao@hotmail.com), N. Wang, and D. Z. Gao (Ocean Univ. of China, 238 Songling Rd., Qingdao 266100, China)

Signal reconstruction is used widely in target identification and communication in underwater. A novel method for impulse signal reconstruction, using the observed data of two receivers which are arranged in the same depth with a certain horizontal interval, is proposed in this talk. This method needs no a prior environmental information but the ranges between the source and the receivers. Although Green function depends on the range, frequency, and weakly on source/receiver depth, spectrum of signal, however, is only dependent on the frequency. On the other hand, dispersion characteristic in shallow water environment provides a compensation mechanism between the frequency and range shift, using this mechanism, the amplitudes and phases of Green function spectral ingredients can be extracted, respectively. Then the impulse signal is obtained by deconvolution processing to the received data of one receiver with Green function. The method is validated using numerical simulation and the correlation coefficient between the reconstructed and original signals is above 0.95 in the cases of high signal-to-noise ratio.

3:00

4pSPa7. Sensitivity analysis of estimating bullet trajectories using ballistic shock wave measurements. Wm. Garth Frazier (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38655, frazier@olemiss.edu)

Estimation of the trajectory of supersonic projectiles using measurements of the shock cone is subject to several sources of measurement error as well as array geometry and miss distance. The microphone position measurement errors are especially important to consider in the case of a moving array. This investigation examines the sensitivity of the solution to these errors and parameters of the array geometry. Cramer–Rao bound analysis is performed in addition to Monte Carlo simulations using maximum likelihood estimation and a more computationally efficient alternative estimator. The results indicate that for several combinations of array geometry and microphone location error, solutions in practical applications can be of very limited value.

THURSDAY AFTERNOON, 26 MAY 2011 METROPOLITAN A, 3:30 TO 5:15 P.M.

Session 4pSPb

Signal Processing in Acoustics and Animal Bioacoustics: Biosonar, Time-Frequency Methods, and Bioacoustic Processing

Charles F. Gaumond, Chair
Naval Research Lab., 4555 Overlook Ave., S.W., Washington, DC 20375-5350

Contributed Papers

3:30

4pSPb1. A deterministic filterbank compressive sensing model for bat biosonar. David A. Hague and John R. Buck (Dept of Elec. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747)

The big brown bat (Eptesicus fuscus) uses FM echolocation calls to accurately estimate range and resolve closely spaced objects in clutter and noise. They resolve glints spaced down to 2 μs in time delay which surpasses traditional signal processing techniques. The matched filter for these calls maintains 10 μs resolution while the inverse filter (IF) achieves higher resolution at the cost of significantly degraded detection performance. Recent work by Fontaine and Peremans [J. Acoust. Soc. Am.(2009)] demonstrated that a sparse representation of bat echolocation calls coupled with a random FIR filter sensing method facilitates distinguishing closely spaced objects over realistic SNRs. Their work raises the intriguing question of whether sensing approaches structured more like the bat's auditory system contain the necessary information for the hyper-resolution observed in behavioral tests. This research estimates sparse echo signatures using a gammatone filterbank closer to the bat auditory system. The filterbank outputs are decimated then processed with ℓ_1 minimization. Simulations demonstrate that this model maintains higher resolution than the MF and significantly better detection performance than the IF for SNRs of 5–45 dB while downsampling the return signal by a factor of 6. [Work supported by ONR and the SMART Program.]

3:45

4pSPb2. Extended Fourier for multi-component chirp analysis: Application to bat signals. Said Assous, Laurie Linnett (Fortkey Ltd., 18/1 Vitoria Terrace, Edinburgh EH1 2JL, United Kingdom), David Gunn, Peter Jackson, John Rees (Br. Geological Survey, Keyworth, Nottingham NG12 5GG, United Kingdom, dgu@bgs.ac.uk), and Mike Lovell (Univ. of Leicester, Leicester LE1 7RH, United Kingdom)

Much progress has been made recently for improving the time-frequency analysis. Typically, signals have frequencies that change with time. Even, in bio-acoustic, creatures such as bats and dolphins rely on time varying signals for echolocation and target detection. With the advent of pulse compression techniques in radar and sonar technologies, the need for time-frequency representations has become more important. The underlying basis of many of the techniques is the Fourier transform. In this paper, we present a technique, which relies on Fourier transform, exhibiting both greater resolution and conceptual simplicity. We have termed the technique Fourier extension analysis since it extends the use of Fourier analysis to signals with time varying components. Furthermore, we demonstrate how the analysis extends naturally to a time-frequency representation using the image processing Hough transform tool without the need for windowing, as it is the case for short time Fourier transform, for example, and we also discuss how the resolutions are obtainable with regard to separation of multi-component chirp signals. Examples will be presented for the analysis of some bat signals showing the capabilities of the developed technique compared to other methods in terms of resolution and cross term free advantages.

4:00

4pSPb3. Analysis/synthesis of sonar echoes as impact sounds. Charles F. Gaumond, Derek Brock, Christina Wasylyshyn (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, charles.gaumond@nrl.navy.mil), Paul C. Hines, and Stefan Murphy (Defence Res. & Development Canada, P.O. Box, 1012 Dartmouth, NS B2Y 3Z7, Canada)

Active sonar performance is sometimes limited by clutter that generates an unacceptable false alarm rate (FAR). High FAR is overcome through the use of signal classification, which is treated here using a sequence of techniques that mimic human perception. The techniques are applied to a corpus of signals that were measured during the experiment Clutter 09, which took place on the Malta Plateau in the spring of 2009. First, techniques for foreground/background separation are presented using whitening and thresholding in a time-frequency representation adapted from computation techniques from acoustic scene analysis. The effects of thresholding are demonstrated with a few signals from the corpus. Modifications, suitable for the noisy sonar-echoes in the corpus, of the natural sound paradigm of Aramaki is presented [Aramaki, *et al.*, Comp. Mus. Mod. Retr. CMMR 2009 (2009)]. Preliminary results of this representation are presented aurally. [Research funded by the Office of Naval Research.]

4:15

4pSPb4. Narrow frequency analysis for periodic broadband noisy signals. Alexander Ekimov (NCPA, The Univ. of Mississippi, University, MS 38677, aekimov@olemiss.edu)

The Wigner–Ville transform was used twice for analysis of periodic broadband noisy signals. The first time use is for an extraction of informative part (part in future) from the signal when time window is sliding across a signal from a beginning to the end. Time window overlap *N*% is applied. This corresponds to *N*% time of existing window for a part of analyzed signal. The time window size is equal to the part duration time T1. That allows increasing the signal to noise ratio for this part within the window. Then the narrow Fourier filtration (which corresponds to the most informative part of the signal) is applied for the output of time window. Broadband signal becomes narrow band after that. Second time the Wigner–Ville approach is used to extract a periodicity in filtered signal. In this case second window was applied. This window size is at least two times more parts' repetition time T2. Overlap is *M*%. This signal processing was successfully used for acoustic and electromagnetic signals of human motion due to footsteps and for underwater clicks of beaked whales in passive acoustic identification. [This work was supported by Army Armament Research, Development, and Engineering Center Contract No. W15QKN-09-C-0163.]

4:30

4pSPb5. Digital audio watermarking using psycho-acoustic frequency masking and Gaussian mixture model. Yejin Seo, Sangjin Cho, and Uipil Chong (School of Elec. Eng., Univ. of Ulsan, 93 Daehak-ro, Nam-gu, Ulsan, Korea, upchong@ulsan.ac.kr)

Most of papers related to the watermarking techniques are currently focused on the development of a blind watermarking detection algorithm and a watermark embedding algorithm based on the psychoacoustics. This paper proposes a blind watermark scheme based on psycho-acoustic frequency masking (PAFM) and Gaussian mixture model (GMM). PAFM is used for watermark embedding into each frame and GMM is used for detecting watermarks. This paper compares robustness and sound quality between the proposed method and informed watermark scheme. Informed method is implemented by using errors-in-variables (EIV) and linear predictive coding (LPC) and is also based on spread spectrum. In this method, target frames are chosen by comparing the average energy for whole signals with the energy of every frame and watermarks are embedded into principal peaks around estimated formants computed by EIV and LPC. In order to evaluate robustness of proposed method, we attacked the sound by using LPF, cropping, and additive noise. [This work was supported by the National Research Foundation of Korea (NRF) grant funded by the Korea Government (MEST) (No. R01-2008-000-20493-0).]

4:45

4pSPb6. Iterative motion compensation algorithm for heart rate detection with ultrasound data. Asif Mehmood and Geoffrey Goldman (U.S. Army Res. Lab., Adelphi, MD, 20783–1197, asif.kyani@gmail.com)

Noncontact active ultrasound sensors can be used to detect human physiological signals such as respiration and heart rate using Doppler processing. Target motion such as a person swaying can reduce the performance of these algorithms. To mitigate the effect of target motion or respiration on estimating the heart rate, we developed an iterative motion compensation algorithm. We low pass filter the demodulated data, estimate the motion of the target, and then compensate the data for the slow moving motion. This procedure can be repeated at different cutoff frequencies for different scenarios. Now, standard Doppler processing techniques can be used to analyze the motion compensated data. The algorithm was tested on people standing in a laboratory illuminated with a 40-KHz continuous-wave ultrasound sensor. Results for estimating the respiration rate and heart rate will be presented.

5:00

4pSPb7. Two dimensional characterization of skeletal muscle's natural vibrations during voluntary contractions. Akibi A. Archer (Woodruff School of Mech. Eng., Georgia Inst. of Technolgy, 775 Ferst Dr. NW, Atlanta, GA, aarcher6@mail.gatech.edu), Perry Atangcho, Minoru Shinohara, and Karim G. Sabra (Georgia Inst. of Technol., Atlanta, GA 30332)

During voluntary muscular contractions, low frequency mechanical vibrations ($<$100 Hz) are naturally generated by skeletal muscles as a result of the muscle fiber activity and the overall change of the muscle-tendon geometry. Determining the temporal and spatial variations of muscle vibrations could reveal fundamental features of skeletal muscles' physiology and activation mechanisms. These mechanical vibrations were recorded over a 3 $\times$ 5 grid of skin mounted accelerometers on the biceps brachii muscle for ten healthy subjects for various contraction levels during isometric elbow flexion. The spatial origin and propagation directionality of these natural muscle vibrations were characterized across frequencies using standard cross correlation techniques and back projection algorithms (akin to time-reversal imaging). Additionally cross-correlating these accelerometer recordings provided travel-time measurements of these natural muscle vibrations between multiple sensor pairs. Furthermore, travel-time tomographic inversions yielded spatial variations of their propagation velocity from which the local stiffness of the muscle could be estimated using elastography principles. This passive tomographic elastography technique could potentially lead to simpler, low-cost, and noninvasive monitoring of musculoskeletal and neuromuscular disorders, without using an external mechanical or radiation excitation as commonly required by conventional elastography techniques.

4p THU. PM

Session 4pUW

Underwater Acoustics: Scattering and Reverberation

David M. Fromm, Chair
Naval Research Lab., 4555 Overlook Ave., S.W., Washington, DC 20375-5350

Chair's Introduction—12:55

Contributed Papers

1:00

4pUW1. Generalized Bragg scattering for uncertain sea surfaces. Roger M. Oba (Acoust. Div. Naval Res. Lab., Washington, DC 20375, roba@wave.nrl.navy.mil)

Bragg's law and related concepts based on acoustic scattering due to surface resonances can provide a framework for computational sea surface scattering. Studies for periodic surfaces [Oba, J. Acoust. Soc. Am. **128**, 39 (2010)] demonstrate that variation of acoustic resonance with the surface phase allows Fourier analysis separation of Bragg scattering orders. More generally, consider one-dimensional, undulating, sea surfaces composed of several Fourier components of known wavelengths, but of uncertain amplitude and phasing. A combination of polynomial chaos (PC) and Fourier expansion for the acoustic field consists of sums of products of Laguerre polynomials and complex exponentials, corresponding to parameters for uncertainty in surface amplitude and phase, respectively. Each single PC-Fourier index quantifies the acoustic response to an individual order in Bragg's law. The Laguerre polynomial terms represent the non-linear acoustic variation with surface amplitude. A particular PC-Fourier expansion coefficient corresponding to several surface wavelength components permits evaluation of cross-component coupling. The PC-Fourier coefficient decay rate indicates the degree of Bragg scattering coupling between components. The robustness of the PC-Fourier coefficients in the presence of a larger set of random rough surface components is explored. [This research is sponsored by the Office of Naval Research.]

1:15

4pUW2. Three dimensional analyzes of scattering by pressure release sinusoidal surfaces. I. Theory. Patrick J Welton (1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

Many previous analyzes of scattering by sinusoidal surfaces are based on the Helmholtz integral equation in two dimensions and assume that the incident energy is a plane wave. Some of these solutions are exact insofar as they do not use the Kirchhoff boundary value approximation. However, all of the analyzes are based on one or more commonly used scattering approximations. Analyses presented here are carried out in three dimensions and are based on an integral equation which is analogous to the Rayleigh integral formula. The scattering is restricted to the plane of incidence, but includes receiver grazing angles ranging from 0 to 180 deg. Both the Fresnel phase approximation and Gaussian source and receiver beam functions are used. One of the objectives of this study is to explore the validity of the Kirchhoff approximation, so shadowing and second order scattering are explicitly included. The surface heights and slopes are not restricted in any way, and the surface slope treatment is exact. The resulting expressions are used in the companion paper to explore various approximations and limiting cases that are often encountered in the scattering literature.

1:30

4pUW3. Three dimensional analyzes of scattering by pressure release sinusoidal surfaces. II. Results. Patrick J. Welton (1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

Theory from the preceding paper is compared with available experimental scattering measurements, and good agreement is found. The Fraunhofer and Fresnel phase approximations are examined, and limitations of validity are discussed. The interaction of the phase and beam approximations is examined as well as variations in the size of the scattering surface. Various slope approximations are compared. The significance of the Kirchhoff approximation and shadowing is explored within the limitations of geometrical acoustics.

1:45

4pUW4. Backscattering from a pressure-release rough surface. Sumedh M. Joshi and Marcia J. Isakson (Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758)

The backscattering of an incident Gaussian-tapered acoustic plane wave is modeled using finite elements, the Kirchhoff approximation, and perturbation theory, with the aim of quantifying the validity of the approximate models. von Karman-type power spectra describing the bottom roughness are sampled from the literature with rms roughness from 3–15% of the acoustic wavelength. Realizations of each type of spectrum are made assuming a Gaussian surface height distribution, and an average backscattering cross section is obtained by ensemble averaging. The pressure-release bottom is used instead of a penetrable sediment, as the focus in this study is to quantify the differences between finite elements and the approximate models due to roughness. [Work supported by ONR.]

2:00

4pUW5. A study on subcritical penetration into rough seafloor. Linhui Peng and Jianhui Lu (Dept. of Marine Technol., Ocean Univ. of China, Qingdao, 266071 China, penglh@ouc.edu.cn)

Bragg scattering from rough interface of seafloor is one of the main causes of subcritical penetration of sound into seafloor. In this paper, the mechanism of subcritical penetration into rough seafloor is analyzed by Bragg scattering from sinusoidal fluctuation surface, and the condition that subcritical penetration can be induced is discussed. Because the refraction angle of minus order Bragg scattering waves is always smaller than refraction angle of Snell's penetration wave, the subcritical penetration attributes to the minus order Bragg scattering waves which propagate as normal plane wave below the critical grazing angle. So, in order to obtain the subcritical penetration, the minus order Bragg scattering wave should be considered detailed. The first-order perturbation approximation is used for rough surface scattering usually. In this paper the validity of first-order perturbation approximation for rough surface scattering is also checked by comparing the first-order Bragg scattering wave with high-order Bragg scattering waves.

2:15

4pUW6. A review of recent developments in underwater acoustic modeling. Paul C. Etter (Northrop Grumman Corp., P.O. Box 1693, Baltimore, MD 21203, paul.etter@ngc.com)

This is the fifth paper in a series of reviews presented at 8-year intervals starting in 1979 [J. Acoust. Soc. Am. **65**, S42 (1979); **82**, S102 (1987); **97**, 3312 (1995); **114**, 2430 (2003)]. All surveys cover basic acoustic models and sonar-performance models. Basic acoustic models include propagation, noise, and reverberation models. Propagation models are categorized according to ray theory, multipath expansion, normal mode, fast field, and parabolic approximation formulations; further distinctions are made between range-independent and range-dependent solutions. Noise models include ambient noise and beam-noise statistics models. Reverberation models include cell and point-scattering formulations. Sonar-performance models include active sonar models, model operating systems, and tactical decision aids. Active sonar models integrate basic acoustic models, signal processing models, and supporting databases into cohesive operating systems organized to solve the sonar equations. Model-operating systems provide a framework for the direct linkage of data-management software with computer-implemented codes of acoustic models. Tactical decision aids represent a form of engagement-level simulation that blend environmental and sonar-performance information with tactical rules. The current inventory of underwater acoustic models comprises 128 propagation models, 21 noise models, 28 reverberation models, and 35 sonar-performance models. Since 1979, approximately five models have been added to the inventory each year.

2:30

4pUW7. A computationally efficient multistatic reverberation algorithm. David M. Fromm (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5350, david.fromm@nrl.navy.mil)

The multistatic reverberation algorithm (MURAL), which supports trainer development, is presented. The algorithm can be used with any propagation model that produces range-sampled grids of transmission loss, travel time, launch, and grazing angles. The flexibility of MURAL with functions that calculate the propagation, scattering, and beam patterns is shown. A reverberation model should produce high-fidelity accurate results in real-time. In reality, the calculation almost always involves the trade-off between computational speed and accuracy. The operation of MURAL couples the algorithm controls to the requested resolution of the prediction with the goal of self-optimizing its performance for the requested resolution. Details are presented that address the technical issues of speed, accuracy, fidelity, and flexibility through several interrelated approaches including (1) the utilization of large amounts of memory available on today's computers, 2) carefully tying the resolution of the look-up tables to the integration, and (3) reordering the nesting of the loops so that the pulse duration is applied after the receiver beamforming. [Work supported by the Office of Naval Research.]

2:45

4pUW8. Scattering in a Pekeris waveguide from a rough bottom using a two-way coupled mode approach. Steven A. Stotts, David P. Knobles, and Robert A. Koch (Appl. Res. Labs, Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758, stotts@arlut.utexas.edu)

Scattering from a rough surface is described numerically with a two-way coupled-mode formalism that contains scattering effects to all orders and provides, in principle, an exact solution to the wave equation. Both scattered field and direct blast components are computed within the framework of the formalism. In contrast to finite elements, each modal component can be calculated separately. A comparison between the scattered component from the coupled mode and Born approximation solutions is presented for scattering from a rough bottom Pekeris waveguide. Total field time series are constructed, and the transition from direct blast to scattered field dominance is identified. For a Gaussian source pulse, the frequency band needed to resolve the scattered component is shown to be much less than is required to resolve the direct blast. Convergence tests to validate the accuracy of the solution are discussed. The importance of the nonlinear contributions to the scattering is identified. The usual method of time averaging the solution produces lower intensity levels than those obtained from the envelope of the time series.

3:00—3:15 Break

3:15

4pUW9. Bottom reverberation modeling with limited bottom interaction information. Xavier Zabal and Michael Porter (HLS Res., Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA)

Frequently, the only available information on bottom interaction is the reflection loss as a function of grazing angle. Here we explore a method of incorporating this information into a normal mode code, such as KRAKEN, via the mode cycle distance. We compare this approach with the perturbation method used by KRAKEN with a full geoacoustic description of the bottom. Results are also compared to an exact solution using a full-bottom characterization. This approach using the cycle distance generalizes to full reverberation calculations, allowing one to predict the spatial and temporal structure of the active sonar reverberation field. The resulting model provides a small trade-off in accuracy but provides the reverberant field extremely rapidly.

3:30

4pUW10. Toward benchmarks of low frequency reverberation level in a Pekeris waveguide: Insight from analytical solutions. Michael A. Ainslie (TNO, Stieltjesweg 1, 2628 CK Delft, The Netherlands), Dale D. Ellis (DRDC Atlantic, Dartmouth, NS B2Y 3Z7, Canada), and Chris H. Harrison (NURC, La Spezia, Italy)

The requirement by modern navies to predict sonar performance in shallow water, whether for use in research, planning, or operations, led to an initiative for the validation of reverberation models in the form of two reverberation modeling Workshops at the University of Texas at Austin [J. S. Perkins and E. I. Thorsos, J. Acoust. Soc. Am. **126**, 2208 (2009)]. The scenario considered here (Problem XI, from the first workshop) requires the computation of reverberation versus time in a Pekeris waveguide with Lambert scattering from the seabed. Simple analytical methods are presented that provide insight into the dominant propagation paths giving rise to the reverberation and hence establish regimes of validity for various computer models making different approximations or assumptions. Results from eigenray, normal mode, and hybrid continuum methods are compared with each other and with the analytical solutions. Numerical predictions are shown to overlap to within a few tenths of a decibel in regions where the different assumptions made by the various models are valid. These overlapping solutions are proposed as "benchmarks" in the sense of a baseline against which future model improvements can be assessed and quantified.

3:45

4pUW11. Integrating the energy flux method of shallow-water reverberation with physics-based seabed scattering models. Ji-Xun Zhou, Xue-Zhen Zhang, Lin Wan (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405), Zhaohui Peng, Zhenglin Li, and Lianghao Guo (Chinese Acad. of Sci., Beijing 100190, China)

The energy flux (angular spectrum) method for shallow-water (SW) reverberation was first presented 3 decades ago in archived Chinese journals. These included closed form expressions for reverberation in SW with a down refraction or isovelocity profile. [Zhou, Acta Acust. **5**, 86–99 (1980); Chin. J. Acoust. **1**, 54–63 (1982)]. More papers on SW reverberation using this method have recently been published in JASA and IEEE JOE. However, most of these papers are based on semiempirical seabed scattering models, which do not have a physical basis. In the past 30 years, one of the major accomplishments in ocean acoustics is the improvement in our understanding of seabed scattering, resulting from a significant effort of both at-sea measurement and theoretical modeling. [Jackson and Richardson, *High-Frequency Seafloor Acoustics*, (2007)]. Benefiting from this accomplishment, this paper integrates the energy flux model of SW reverberation with the physics-based seabed scattering models in the angular domain. This integration directly and intuitively results in a general expression for SW re-

4p THU. PM

verberation in terms of seabed physical/scattering parameters, and this expression is almost identical with the expression derived from the boundary perturbation method. The results may be used to estimate the bottom roughness or sediment inhomogeneity from SW reverberation measurements .[Work supported by the ONR and CAS].

4:00

4pUW12. The modeling and analysis of the reverberation on the continental shelf off of New Jersey. Youngmin Choo and Woojae Seong (Dept. of Naval Architecture and Ocean Eng., College of Eng., Seoul Natl. Univ., San 56-1, Sillim-dong, Kwanak-gu, Seoul 151-744, Korea)

As interest of operating sonar system in the littoral area increases, study of reverberation on a continental shelf is needed. In this study, an incoherent reverberation on a continental shelf was simulated using a 2-D propagation model based on ray theory with scattering strength formula based on various theories. Each scattering cross section was modified for application to von Karman roughness spectrum, which was observed during an experiment on the continental shelf off of New Jersey. Various reverberation patterns were shown according to each scattering cross section. By comparing the results from the model and experimental data, an adequate scattering strength was found for each source frequency, and the dominant mechanism of the reverberation on the continental shelf was identified for each time step. In case of the monostatic reverberation measurement, the source and receiver were located on the border between slopes. So both slopes affected the reverberation. In case of the bistatic reverberation, scattered signal from each slope affected the reverberation at different times. The characteristic of bistatic reverberation was different from that of monostatic reverberation. The effect of source-receiver separation on the reverberation of a continental shelf was studied through simulating monostatic and bistatic reverberations.

4:15

4pUW13. The impact of bottom backscatter strength modeling on active sonar optimal frequency calculations. Colin W. Jemmott and William K. Stevens (Appl. Operations Res., 420 Stevens Ave., Ste. 230, Solana Beach, CA 92075, colin.jemmott@appliedor.com)

For many active sonar problems, there exists an optimal frequency—one at which detection range maximized. The optimal frequency can be calculated using a frequency dependent analytical sonar equation to solve for the detection range, defined as the largest range with positive signal excess. The frequency with the maximum detection range is the optimal frequency. An advantage of this approach is that it combines all of the sonar equation terms into a single quantity. It might seem that such a frequency will not exist because propagation loss tends to increase with increasing frequencies, so performance might decrease monotonically with frequency. However, other terms in the sonar equation are also frequency dependent. For example, Knudsen or Wenz curves show that noise levels generally decrease with frequency, while source level and target strength may be complicated functions of frequency. In shallow water, active sonar may be reverberation limited rather than noise limited, but standard Lambert scattering strength models are frequency independent. New, more accurate bottom backscatter strength models have frequency dependence, which will impact the optimal frequency calculation. This approach demonstrates one operational impact of improved reverberation modeling, and quantifies the performance degradation caused by a suboptimal frequency selection.

4:30

4pUW14. Surface scattering rejection for clear sidescan images. Stephen K. Pearce and John S. Bird (School of Eng. Sci., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A-1S6, Canada, skp6@sfu.ca)

Surface scattering is a major source of interference for sidescan sonar systems operating in high traffic areas or in choppy water. This surface scattering from wakes, as a result of boat traffic or from the chop of the sea surface, obscures the bottom return. A sonar system with a multi-element array can separate surface signals from bottom signals using beamforming, thereby creating clear images of the seafloor. This multi-element array can have as few as six elements and still effectively remove surface scattering. In this paper, different beamforming techniques are applied and their impact on suppressing surface returns is shown. Comparisons between beamforming methods are made by comparing the relative path levels with and without beamforming applied. The ability of a multi-element array to successfully discriminate between bottom returns and surface returns using beamforming is then shown using both simulated and experimental data. It is concluded that sonar systems employing a multi-element array can produce clear images of the seafloor even in the presence of strong surface interference when beamforming is used to create a beam that has a broad main lobe pointed toward the bottom and low sidelobes pointed toward the surface.

4:45

4pUW15. Simulations of multibeam sonar echos from schooling individual fish. Arne Johannes Holmin (Dept. of Mathematics, Univ. of Bergen, Johannes Bruns Gate 12, P.O. Box 7803, 5008 Bergen, Norway), Nils Olav Handegard, Rolf Korneliussen (Inst. of Marine Res., Nordnes, 5817 Bergen, Norway), and Dag Tjøstheim (5008 Bergen, Norway)

Echo sounders and sonars are important tools in fisheries research, and interpretation of fish school data can be used to extract information about fish behavior. For such interpretations to be reliable, accurate acoustic modeling of the targets and the acoustical instrument is needed. A simulation model for multibeam and multifrequency echo sounders and sonars is presented, in which synthetic echograms of fish schools are generated based on known individual dynamic and acoustical variables and beam configuration of the specific acoustical instrument. Using the simulation model, acoustic observations of fish school behavior patterns can be generated and analyzed along with observations of real fish schools. This provides a tool for validating both the interpretation of real data and the interpretation of behavior models based on individual interactions. Examples of the effect of local and global orientation fluctuations are presented and compared to data of real fish schools, specifically for the SIMRAD MS70 multibeam sonar.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. On Tuesday and Thursday the meetings will be held starting immediately after the Social Hours at 8:00 p.m. On Wednesday, two technical committees will meet at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

Architectural Acoustics	Willow A
Animal Bioacoustics	Issaquah
Speech Communication	Cirrus
Underwater Acoustics	Metropolitan A

Session 5aAA

Architectural Acoustics: Acoustics of Healthcare Spaces I

Kenneth P. Roy, Cochair
Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

Erica E. Ryherd, Cochair
Georgia Inst. of Technology, Mechanical Engineering, 771 Ferst Dr., Atlanta, GA 30332-0405

Chair's Introduction—8:00

Invited Papers

8:05

5aAA1. Noise in the surgery area of a four hundred beds hospital. Sergio Beristain (E.S.I.M.E., IPN, IMA. P.O. Box 12-1022, Narvarte, 03001 Mexico, pistaito Federal Mexico, sberista@hotmail.com)

The studied hospital has eight surgery rooms in a U shaped installation: several sources and relevant conditions have been identified and evaluated. Some rooms are normally allocated for specific types of surgery such as eyes, brain, or heart procedures, while some others are employed for general purpose surgery, which depend on the day to day needs and programming of the hospital: none new borne surgery is performed in this surgery area, and the last one of them is reserved for all kinds of emergency surgery, colloquially known as the dirty surgery room, where patients enter directly from the street, where they have had an accident or a sudden illness. The whole surgery area represents a special noise condition because some surgeries are simple, programed, and require no special noisy tools, but, for instance, in the traumatology or bone surgery, hammers and saws are commonplace during the procedure. General noise measurements were performed during some surgeries in order to find out the prevalent noise condition in this section of the hospital.

8:25

5aAA2. Comparing novel absorptive treatments in operating rooms. Timothy Y. Hsu, Erica E. Ryherd (George Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, gth776e@mail.gatech.edu), Colin Barnhill, James West (Johns Hopkins Univ., Baltimore, MD 21218), and Natalia Levit (DuPont, Richmond, VA 23234)

Hospital noise levels have been rising over the past several decades, and there have been few viable noise control solutions. The hospital environment can have special health requirements due to strict infectious control criteria. In order to meet these requirements, a novel absorbing panel was developed for a previous study in this series and installed in hematological intensive care units at the Johns Hopkins University Hospital Weinberg Building. This current study focuses on the impact of installing the panels in an operating room in the Johns Hopkins University Hospital. The impacts of the addition of sound absorbing materials in these operating rooms were analyzed both acoustically and perceptually. Sound level recordings and reverberation times were measured before and after the installation of the absorbing panels. Comparisons of these acoustic measurements will be presented. Additionally, a before/after questionnaire was administered to the operating room medical staff. Statistically significant differences in the survey results will be presented. These preliminary results in acoustic and survey data illustrate the effectiveness of the use of novel hospital-approved absorbing panels in the operating room.

8:45

5aAA3. Isolating above-ground-floor magnetic resonance imaging (MRI) facilities: A case study. Dennis Noson, Daniel Bruck (BRC Acoust., 1734 1st Ave. South, Ste. 401, Seattle, WA 98134, dnoson@brcacoustics.com), and Dave Forrest (Forrest Sound Products, 14158 177th Ave. NE, Redmond, WA 98052)

A successful MRI installation requires careful coordination of design and construction disciplines to produce a noise isolation result satisfying the medical staff and their need for low distraction in the work environment. One of Seattle's largest medical centers recently constructed a new facility designed to meet stringent requirements for its second floor location, chiefly that sound levels during MRI operation were required to be less than 34 dBA at any adjacent staff area. With separate clinics on the floors above and below the MRI, sound isolation details at perimeter beams, area partitioning, and resilient fastening methods were all important to achieving the noise goal. Design details will be presented, along with measurements of MRI noise reduction spectra collected after completion of the project, and comparison will be made with predicted performance of the architectural assemblies. Vibration isolation was important to the MRI operation. Vibration measurements will also be presented and discussed relative to the MRI manufacturer's design criteria.

9:05

5aAA4. Case study: Innovative acoustical design of central plant. Peter K. Holst and Robert P. Alvarado (Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104)

This presentation will provide insights from a successful project that included a central plant located at the ground floor of a new hospital tower in a residential neighborhood in Burlingame, California. Design considerations at the onset of the project included establishing acceptable noise criteria for nearby residences. Ambient noise surveys were conducted and appearances at public hearings ensured the noise levels of the equipment would not exceed the existing noise environment. Solutions for reducing equipment noise include the use of ultraquiet cooling towers, sound-absorbing concrete block, and an inventive acoustical silencer design at the intake and discharge of a generator room. Within the hospital, space planning would not afford a buffer floor separating the generator room from vibration-sensitive eye-surgery operating rooms directly above. A heavy-duty spring-isolated ceiling, built in sequence with an independent spring-isolated equipment-support grid, required significant levels of coordination between the acoustical consultant and construction team. This presentation will include a summary of project concerns, innovative design solutions to meet the applicable noise criteria, and postconstruction measurement results. This valuable information could be used in the design of future medical facilities in residential neighborhoods, and in the design of unavoidable juxtapositions at sensitive uses.

9:25

5aAA5. Effects of noise on emergency department staff. Arun Mahapatra, Selen Okcu, Erica Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30332), Jeremy Ackerman (Emory Univ. School of Medicine, Atlanta, GA 30322), and Craig Zimring (Georgia Inst. of Technol., Atlanta, GA 30332)

The hospital sound environment is complex. The impact of a poor soundscape is becoming an increasing concern for both staff and patients, particularly in emergency departments (EDs). While there is growing evidence of the negative impacts of a poor soundscape, there is surprisingly little rigorous evidence about exactly what acoustic characteristics impact hospital staff. This paper presents a detailed acoustic study of two EDs in Atlanta, GA. Equivalent sound pressure levels (Leq), speech intelligibility index (SII), spectral distribution, and other acoustic metrics were calculated for four locations in each emergency department over 24 h. Results show Leq's between 50–63 dBA, which severely exceeds the 30 dBA guideline level set by the World Health Organization. Additionally, no area observed held a "good" SII rating. These acoustic measurements will be compared to results of staff questionnaires focusing on staff stress and job satisfaction. Findings from this work will advance the understanding of how various characteristics of the ED soundscape impact occupants, how to best measure and quantify these characteristics, and how to improve the ED soundscape. [Work supported by Health Systems Institute (HSI).]

9:45

5aAA6. Measuring the effects of acoustical environments on nurses in health-care facilities: A pilot study. Hind Sbihi, Murray Hodgson, George Astrakianakis (School of Env. Health, Univ. of British Columbia, 3rd Fl., 2206 East Mall, Vancouver, BC, V6R3J5, Canada, murray.hodgson@ubc.ca), and Pamela Ratner (Univ. of British Columbia, Vancouver, BC, V6T1Z3, Canada)

This paper summarizes the methods and results of a pilot ecological study conducted in four health-care facilities (acute-care, community-care, and long-term-care). The objective was to consolidate and test tools for exposure assessment and the investigation of study outcomes, in particular, stress. Area and personal monitoring was performed. Nurse noise exposures were monitored. Full-shift monitoring of sound levels was performed, and conventional acoustical parameters derived; new acoustical descriptors including occurrence rate and peakiness were also determined. Two questionnaire scales were developed: a study questionnaire to assess perception of the acoustical environment and of work- and noise-related stresses, and a daily diary to capture variations in the perceived stress and document aggressive events. The study questionnaire was found to measure disturbance, impaired communication, and mental fatigue. Biological markers of noise-related stress (salivary cortisol and heart-rate variability) were collected. Exposure measures were correlated with outcomes; while the results were often not statistically significant due to small sample sizes, they identified interesting relationships and validated the measurement tools for future use. Long-term-care was identified as the most acoustically-critical environment both from a physical-acoustical perspective and from the perspective of workers.

10:05—10:15 Break

10:15

5aAA7. Acoustics of interconnected nursing unit corridors. Selen Okcu (College of Architecture, Georgia Tech, Atlanta, GA 30332), Erica Ryherd, and Craig Zimring (Georgia Tech, Atlanta, GA 30332)

Nursing units are composed of proportional (e.g., patient room) and nonproportional spaces (e.g., race track and hallway design nursing unit corridors). Acoustic qualities of these spaces can have a significant impact on sound task performance of caregivers such as localization of auditory cues. For example, it is well known that in reverberant environments, the human auditory system's ability to localize auditory cues is very limited. Various nonhospital studies documented the acoustic qualities of proportional spaces. However, the number of studies linking the design and acoustics of long enclosures in hospitals is very limited. By conducting acoustic simulation analysis, impulse response measurements, and heuristic design analysis, this study statistically analyzed the link between design and acoustics of interconnected nursing unit corridors.

10:35

5aAA8. Psychoacoustic measures and their relationship to patient physiology in an intensive care unit. Timothy Y. Hsu, Erica E. Ryherd (George Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, gth776e@mail.gatech.edu), Jeremy Ackerman (Emory Univ. School of Medicine, Atlanta, GA), and Kerstin Persson Waye (The Sahlgrenska Acad. of Gothenburg Univ., 405 30 Gothenburg, Sweden)

Many researchers have attempted to characterize the soundscape of hospital wards using traditional acoustic metrics. These traditional metrics, such as average sound level, are readily measured using sound level meters and have been the primary results reported in previous studies. However, it has been shown that these traditional metrics may be insufficient in fully characterizing the wards. This is a continuation of a larger study that evaluates the relationship between hospital soundscape and the effects on the hospital occupants in a medical-surgical intensive care unit. The previous study in this series showed statistically significant relationships between patient physiology and the traditional sound level meter metrics. This current study aims to expand beyond traditional sound level meters and use digital recordings to describe the ICU rooms with psychoacoustics metrics. Thus, a comprehensive catalog of these metrics for ICUs is compiled and presented. In addition to descriptive results with these measures, statistical analysis is used to reveal preliminary relationships between psychoacoustic metrics and the patient physiological responses. These results continue to reveal the importance of understanding the relationships between hospital acoustics and patient physiological arousal. [Work supported by ASA and Swedish FAS.]

10:55

5aAA9. In defense of sleep. Jo M. Solet (15 Berkeley St., Cambridge, MA 02138)

In defense of sleep environments of care that deliver 24-h service to patients and residents ideally provide a level of quiet compatible with sleep. Yet, historically the noise levels in hospitals have been rising, leading the Facilities Guidelines Institute, for the first time, to include minimum acoustic standards in the 2010 cycle of the Guidelines for the Construction of Health Care Facilities. This presentation will provide the most recent information about the nature of sleep, the health risks associated with limited sleep, and the U.S. epidemiological evidence supporting the conclusion that sleep is a critical pillar of health, in parallel with nutrition and exercise. Listeners will be equipped with scientific evidence in defense of sleep, which will enable them to advocate for the critical role of acoustics expertise as part of the design process. In addition, practitioners will discover new incentives to better protect their own sleep, as part of effective self-care and to enhance performance.

11:15

5aAA10. Harmonizing national and worldwide acoustical guidelines for healthcare. David M. Sykes (Remington Partners, 31 Baker Farms Rd., Lincoln, MA 01773, david.sykes@remingtonpartners.com)

In the absence of leadership from the U.S. Federal Government, the process of developing (and continuously improving) consistent acoustical guidelines for healthcare facilities is like herding cats since many independent authorities compete with each other. Nevertheless, progress continues. 1 year after the release of the FGI 2010 Guidelines—the first code-level, comprehensive U.S. guidelines in sixty years for noise and vibration in healthcare facilities—several groups have agreed to cite the Guidelines as their sole "reference standard": the Department of Health and Human Services (responsible for HIPAA), the Veterans Administration, the Centers for Disease Control, 38 regional AHJ's (Authorities having Jurisdiction), the U.S. Green Building Council's LEED for Health Care, the Green Guide for Health Care, 48 State code authorities, and most recently, the International Green Code Council. What is left? The Joint Commission, the Department of Defense, the World Health Organization, and others. Meantime, work on the 2014 edition of the Guidelines is already underway. Since the FGI Guidelines are actively used in 16 countries, it is imperative to involve the global acoustical community in the research and development needed to strengthen them. Can ICBEN help with this task? What is the best way forward?

11:35

5aAA11. Practical considerations for helicopter noise control in hospitals. Chris Papadimos and Roman Wowk (Papadimos Group, 818 Fifth Ave. Ste. 207, San Rafael, CA 94901)

Medical transport helicopters produce very high noise levels including pronounced low-frequency content that has the potential of inducing perceptible vibration. Based on recent experiences helicopter noise and vibration can be very challenging to control inside hospital buildings, especially if the helipad is located on the roof above occupied floors. Although such events may be tolerable in noncritical areas, it should be carefully addressed in patient rooms or any other healing places to prevent harmful effects on patient sleep and recovery. Current standards and guidelines for the design of healthcare facilities do not provide specific criteria for helicopter noise and vibration inside hospitals; however, appropriate limits may be selected based on past sleep disturbance studies and other available information. Appropriate helicopter noise control is particularly challenging with the current trend of exterior curtain wall constructions with a high percentage of glazing, thus requiring early evaluation including proper space planning and budgeting prior to committing to such designs. Helicopter noise studies will be presented to communicate recent experiences for large-scale academic healthcare projects.

Session 5aAB

Animal Bioacoustics: General Topics in Passive Acoustic Monitoring of Animals I

Catherine L. Berchok, Chair
NOAA, Alaska Fisheries Science Center, Seattle, WA 98115

Contributed Papers

8:30

5aAB1. Comparison of visual and acoustic detection rates for cetaceans off Washington. John Calambokidis (Cascadia Res., 218 1/2 W 4th Ave., Olympia, WA 98501), Erin Oleson (Pacific Islands Fisheries Sci. Ctr., Honolulu, HI 96814), Erin Falcone, Greg Schorr (Cascadia Res., Olympia, WA 98501), Sean Wiggins, and John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92037)

From 2004–2010, we examined marine mammal occurrence off Washington acoustically and visually. Acoustic monitoring was conducted by Scripps Institution of Oceanography using high-frequency acoustic recording packages (sample rate 80–200 kHz) in Quinault Canyon, part of the Navy's Northwest Range. Visual surveys were conducted using small 6 m rigid-hull inflatable boats covering out to the location of the HARPs during favorable weather year-round. Visual surveys detected over 500 sighting of 12 species of cetaceans. Humpback whales were the most common baleen whale; harbor and Dall's porpoise were the most common odontocete detected visually. Species detected acoustically included Pacific white-sided and Risso's dolphins, and beaked, killer, sperm, humpback, blue, and fin whales. While there were similarities in the detections by both methods there were also some Stark contrasts. Overall, there were more acoustic detections than visual sightings and these included sperm whales, which were detected frequently acoustically but never visually. For humpback whales, acoustic detections were highest in fall and winter (and low in summer), while visual sightings showed highest numbers in summer and early fall. These differences can be explained by the dive and calling behavior of these species and demonstrate how the strengths and weaknesses of these approaches complement each other.

8:45

5aAB2. North Atlantic right whale seasonal presence off the coast of New Jersey: Confirmation by passive acoustic monitoring and ship survey data. Kathleen M. Dudzinski, Amy Whitt, and Jennifer Laliberte (2201 K. Ave., Ste. A2, Plano, TX 75074, kdudzinski@geo-marine.com)

North Atlantic right whales (NARW) are one of the most critically endangered marine mammals with abundance estimates for the North Atlantic population at about 438 individuals cataloged in 2008. Data presented here were part of a larger, long-term study to assess presence of marine mammals, sea turtles, and birds along the New Jersey coast in advance of wind farm development. Seasonal presence of NARW off New Jersey was characterized using static passive acoustic monitoring (PAM) and line-transect shipboard surveys. NARW upcalls were detected on 115 days over 21 months of deployment (March 2008 to December 2009) with a significant difference in number of upcalls detected between PAM stations by month (F-ratio = 3.1292, df = 22, p = 0.000). NARW upcalls were detected from March to June and September to December 2008 and from January to March and in June 2009, with the greatest number of calls detected during spring months annually. Presence of NARW was confirmed by sighting data, with sightings all seasons except summer. Four sightings were recorded: three during November, December, and January when right whales are on calving grounds farther south or in the Gulf of Maine and the fourth sighting was a cow-calf pair in May.

9:00

5aAB3. Relationships between gray whale (Eschrichtius robustus) calling rates, group size, and ambient noise levels in Laguna San Ignacio, Mexico. Diana Ponce-Morado, Aaron M. Thode, Melania Guerra (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA 92093-0205, dponcemo@ucsd.edu), Jorge Urban (Laboratorio de Mamxxed;feros Marinos de la Universidad Auto de Baja California Sur (UABCS), Baja California Sur Mxico), and Steven Swartz (Laguna San Ignacio Ecosystem Program, Darnestown, MD 20874)

Determining relationships between calling activity and group size for marine mammal species is challenging, in part due to difficulties in obtaining reliable independent visual censuses of animals in open waters. In this study, acoustic calling rates of eastern Pacific gray whales were measured over a 4-week period during their 2008 breeding season in the sheltered lagoon of Laguna San Ignacio of Baja Mexico. Visual counts were conducted for 6 days during the deployment. It was found that the lagoon population more than doubled over the observational period, with much of the increase occurring over a 7-day period. Acoustic data collected during those 6 days were manually reviewed to yield counts of various gray whale call types during each day. All call rates showed peaks in early morning and evening, with minimum rates generally detected in the early afternoon, a time of low ambient noise but high tourist panga activity. The number of $S1$-type calls counted over 24 h increased roughly as the square of the number of the animals in the lagoon, when call counts were adjusted for variations in background ambient noise levels. An exception to this trend occurred during a time of rapid population increase in the lagoon.

9:15

5aAB4. Variations in the number of fin whale calls recorded at different locations in the Northeast Pacific Ocean. Michelle J. Weirathmueller, Dax C. Soule, and William S. D. Wilcock (School of Oceanogr., Univ. of Washington, Box 357940, Seattle, WA 98195-7940, michw@u.washington.edu)

A large number of fin whale calls have been observed in a 3-year ocean bottom seismometer dataset (2003–2006) over the Endeavor Ridge (48°N /129°W), a hydrothermally active area in the Northeast Pacific Ocean. Most of the vocalizations were detected during the winter months. Because zooplankton constitute an important part of fin whales' diets, and enhanced populations of zooplankton have been observed at all depths above the Endeavor hydrothermal vents, it has been hypothesized that the fin whales could be near the vents specifically for feeding. As part of the analysis of the Endeavor vent field data set, algorithms have been developed, which utilize the absolute and relative spectral energy levels in the frequency band of the whale vocalizations. In order to test whether the concentrations of whale vocalizations are unusually high over the hydrothermally active area, the detection algorithm is being applied to data from individual ocean bottom seismometers at other nearby locations including the center of Explorer Plate (49.5°N, 129°W), and the base of the continental slope off Nootka Sound (49.3°N, 127.6°W).

9:30

5aAB5. Passive acoustic monitoring of cetacean at Josephine Seamount, Portugal. Giacomo Giorli (Dept. of Oceanogr., Univ. of Hawaii, 1000 Pope Rd., Honolulu, HI 96822, giacomog@hawaii.edu), Whitlow W. L. Au (Univ. of Hawaii, Kaneohe, HI 96744), and Ronald P. Morrissey (Naval Underwater Warfare Ctr. Div. Newport, RI)

In the past years, the development of a variety of passive acoustics recorders has provided a unique way of acquiring information about marine mammal species in remote regions of the ocean. During the NATO Undersea Research Centre Sirena 10 cruise an ecological acoustic recorder (EAR) was deployed to monitor the presence of marine mammal. The EAR was deployed on May 13, 2010 at a depth of 944 m in the vicinity of Josephine seamount (37 deg; 02.087'N, 013¢ª 51.733W, south west Portugal) and programed for a recording time of 40 s every 2 min. For the detection and classification of echolocation clicks we integrated the visual analysis of spectrograms with the analysis performed by an automatic detector/classification system called M3R, developed by the Naval Undersea Warfare Center Division. M3R has been successfully operated for the analysis of recording from different US Navy ranges. Cetaceans detected include beaked whales, sperm whales, pilot whales, risso's dolphins, and dolphins. Hourly and daily distributions of signals were calculated. Beaked whales have been regularly detected everyday and the echolocation activity has been found to be higher at night. This suggests that Josephine seamount represents a good habitat for these deep diving cetaceans. [NATO Undersea Res. Ctr. funded and operated the sea trial Sirena 10.]

9:45

5aAB6. A comparison of the song of humpback whales (Megaptera novaeangliae) wintering in the Northwestern and Main Hawaiian Islands. Jessica Chen, Marc O. Lammers, and Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, jchen2@hawaii.edu)

An extensive study of the population structure of North Pacific humpback whales (SPLASH) has revealed that many of the whales feeding in the Aleutian Islands and the Bering Sea are not observed in known wintering and breeding areas in the Pacific. Recent findings suggest that the Northwestern Hawaiian Islands (NWHIs) represent a previously undocumented wintering area that may be distinct from the Main Hawaiian Islands' (MHIs') wintering ground. To examine this issue further, data from Ecological Acoustic Recorders (EARs) deployed at five locations in the NWHIs and MHIs were analyzed to compare the structure of songs produced by whales in both areas. Previous research suggests that songs from different breeding groups differ from one another, providing a method to evaluate whether whales in the NWHIs are part of the same breeding group as those in the MHIs. Recordings from each site were randomly selected, and song units were classified and counted to compare between sites. Preliminary results indicate that there are differences between the NWHIs and MHIs. These findings suggest that further work using photo ID should be undertaken to confirm whether whales in the NWHIs are indeed those feeding in the Aleutians and the Bering Sea.

10:00

5aAB7. Analyzing potential acoustic differences between different bowhead demographics in the Beaufort Sea. Dawn M. Grebner, Aaron M. Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093, dgrebner@ucsd.edu), Susanna B. Blackwell, Charles R. Greene (Greeneridge Sci., Inc., Santa Barbara, CA 93117), William R. Koski (LGL Environ. Res. Assoc., 22 Fisher St., King City, ON L7B 1A6, Canada), Dale Funk (LGL Alaska Res. Assoc. Inc., 1101 East 76th Ave., Ste. B, Anchorage, AK 99518), and Michael Macrander (Shell Exploration and Production Co., Anchorage, AK 99503)

Most bowhead whales from the Bering-Chukchi-Beaufort stock migrate westward from Arctic Canadian waters through the Alaskan Beaufort Sea each year from late August to late October. Since 2007 both aerial sighting and acoustic call data have been collected during these fall migrations, as part of a long-term monitoring effort to assess the potential impacts of North Slope oil industry activities on bowhead whales. The aerial sighting efforts recorded the position of bowhead individuals or groups, designating the animals as subadults or adults, and noted whether a calf was present. A five-site array of seven directional autonomous seafloor acoustic recorders (DASARs) each recorded the acoustic data. An automated detection algorithm was used to isolate individual bowhead whale calls and estimate their locations. Here simultaneous acoustic and visual data from 2007, 2008, and 2010 were merged to determine whether potential variations in calling behavior exist between subadults, adults, and adults with calves. Visual measurements of animal course and direction were used to place bounds on call times and locations that may be associated with that animal versus other animals sighted nearby. Potential differences in call rate and call type were statistically examined. [Work supported by the Shell Exploration and Production Company.]

10:15—10:30 Break

10:30

5aAB8. Multi-year use of unique complex songs by western arctic bowhead whales: Evidence from three years of overwinter recordings in the Chukchi Sea. Julien Delarue (JASCO Appl. Sci., 1496 Lower Water St., Ste. 432, Halifax, NS B3J1R9, Canada, julien.delarue@jasco.com)

The current understanding is that singing is part of bowhead whales' breeding behavior and that songs change every year. The latter is based on the analysis of songs recorded in the springtime off Barrow, AK, but it is also accepted that spring songs may be somewhat degraded in comparison to those produced in winter when bowheads are most sexually active. A year-long passive acoustic monitoring program in place since July 2007 in the Chukchi Sea provided an opportunity to compare songs recorded during three consecutive fall migrations. One unique song was detected in 2007, 2008, and 2009 and another was detected in 2007 and 2008. Minor variations are likely the result of a difference in song maturity caused by bowheads' different departure times from the instrumented area in all 3 years. This finding strongly suggests that, when starting to sing in the fall, western arctic bowheads display the same songs every year. Variations occurring throughout the winter could explain the previously observed differences in spring songs at Barrow. However, some songs we recorded during the spring migration in all 3 years showed striking similarities. Songs recorded in the Bering Sea in winter should be examined to fully validate this finding.

10:45

5aAB9. Statistical analysis of fin whale vocalizations recorded by a seismic network at the Endeavour Segment of Juan de Fuca Ridge, N. E. Pacific Ocean. Dax C. Soule (School of Oceanogr., The Univ. of Washington, Box 357940, Seattle, WA 98195, daxsoule@u.washington.edu), William S. D. Wilcock (Univ. of Washington, Seattle, WA 98195), and Richard E. Thompson (Inst. of Ocean Sci., Sidney, BC V8L 4B2, Canada)

From 2003–2006, an eight-station seafloor seismic network was deployed along the Endeavour Segment of the Juan de Fuca ridge that recorded an extensive data set of 20-Hz fin whale calls. Algorithms have been developed to detect and track vocalizing whales that swim near the seismic network. During the first year of operation, more than 100000 fin calls that include ~100 whale tracks were identified. Tracks comprise both single whales distinguished by a stereotyped ~25 s interpulse interval and inferred multiwhale tracks characterized by more complex interpulse intervals. Whale tracks vary from individuals or groups that cross the network in a few hours to those that meander for up to 24 h. The call rates vary seasonally with the highest rates in winter and exhibit an apparent weak diurnal variation. The center frequencies range from 17–34 Hz, with the primary population centered at 20 Hz and a secondary population centered at 25 Hz. Statistical analysis of observed bandwidths and center frequencies, interpulse intervals, seasonality, and diurnal patterns will be presented. Additionally, the ~100 whale tracks will be used for migration analysis and to quantify the swimming patterns in the network. [Funding from the ONR.]

11:00

5aAB10. Characterizing and classifying humpback whale (Megaptera novaeangliae) song units. Adrienne M. Copeland, Whitlow W. L. Au (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734), Marc O. Lammers (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), Adam A. Pack (Univ. of Hawaii at Hilo, Hilo, HI 96720), and Julie N. Oswald (Oceanwide Sci. Inst., Honolulu, HI 96839)

Humpback whales, Megaptera novaeangliae, are one of the most recognizable and investigated marine mammals. However, little progress has been made in automatically distinguishing and classifying individual units of their song. A Matlab script has been developed to characterize the different song units and to apply the appropriate statistics to separate and categorize each unit. The Matlab program measures 48 parameters from each song unit. The songs were recorded by a swimmer snorkeling above vocalizing humpbacks in the waters off Maui, HI. A 16 bit, digital tape recorder with the automatic gain control disabled and a sample rate of 44.1 kHz was used to record songs from different whales. The swimmer determined the range of the whale using a portable handheld fathometer. Singing whales typically suspended themselves in the water column at depths varying from 15 to 30 m, which was contingent on the bottom depth. Song units were separated into distinct categories using a principle component analysis (PCA) based on the 48 parameters describing each unit. A classification algorithm was then developed based on the categories determined by the PCA. The classifier will be integrated into a modified version of the real-time Odontocete call classification algorithm (ROCCA) for future analyzes.

11:15

5aAB11. Seeing the species through the trees: Using Random Forest classification trees to identify species-specific whistle types. Julie N. Oswald (Bio-Waves, Inc., 517 Cornish Dr., Encinitas, CA 92024, julie.oswald@bio-waves.net), Jim V. Carretta (NOAA Fisheries, Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Court, La Jolla, CA 92037), Michael Oswald (Bio-Waves, Inc., 517 Cornish Dr. Encinitas, CA 92024), Shannon Rankin (NOAA Fisheries, Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Court, La Jolla, CA 92037), and Whitlow W.L. Au (Hawaii Inst. of Marine Biology, P.O. Box 1106, Kailua, HI 96374)

Acoustic identification of delphinid species is hampered by high variability in whistle characteristics. It is possible that not every whistle contains species-specific information and that there are "species-specific" whistle types. Random forest analysis was used to examine whistles of 8 species recorded in the eastern tropical Pacific Ocean (*Delphinus* species, *Globicephala macrorhynchus, Pseudorca crassidens, Stenella attenuata, S. coeruleoalba, S. longirostris, Steno bredanensis, Tursiops truncatus*). Fifty-one variables were measured from 2176 whistles. The number of trees within a random forest that "voted" for the predicted species was used as a measure of the strength of classification. A whistle was considered strongly classified if the predicted species received at least 40% of the votes, even if the prediction was incorrect. The percent of whistles that were strongly classified ranged from 33% (*S. longirostris*) to 73% (*G. macrorhynchus*). Overall, 62% of strong whistles were correctly classified, ranging from 22% (*S. longirostris*) to 87% (*P. crassidens*). Overall correct classification for weakly classified whistles was 33% and ranged from 17% (*Delphinus* spp. and *G. macrorhynchus*) to 55% (*S. bredanensis*). Results suggest that while there may be "species-specific" whistle types, the distinctiveness of these whistles and the frequency with which they are produced varies among species.

11:30

5aAB12. Integration of real-time odontocete call classification algorithm into PAMGUARD signal processing software. Michael Oswald, Julie N. Oswald (Bio-Waves Inc., 1474 Tennis Match Way, Encinitas, CA 92024, mike.oswald@bio-waves.net), Marc O. Lammers (Hawaii Inst. of Marine Biology, Kailua, HI 96374), Shannon Rankin (SWFSC, La Jolla, CA 92037), and Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Kailua, HI 96374)

Real-time odontocete call classification algorithm (ROCCA) is a tool for real-time acoustic species identification of delphinid whistles. Introduced in 2006 as MATLAB-based software, ROCCA is currently being incorporated into PAMGUARD, a freely-available, open source software package. ROCCA provides automated extraction of whistle contours from a spectrogram. It measures 54 whistle contour features including frequencies, slopes, duration, and variables related to the positions of inflection points and steps. ROCCA currently classifies whistles of seven species and one genus: *Globicephala macrorhynchus, Pseudorca crassidens, Steno bredanensis, Stenella attenuata, S. coeruleoalba, S. longirostris, Tursiops truncatus*, and *Delphinus* species. The classifier is a Random Forest trained on 2231 whistles collected over six cruises and 7 years in the eastern tropical Pacific Ocean. The original ROCCA classifier used a combination of discriminant function analysis and CART algorithms on 13 whistle contour features for an overall correct classification score of 35%, which was significantly greater than random (12%). The current Random Forest scheme, trained on 54 whistle contour features, yields an overall correct classification score of 62%. Feedback from at-sea beta testing has been incorporated into the latest version of ROCCA. Additional species, automated detection, and alternate classification schemes are being explored to extend ROCCA into different geographic areas with greater accuracy.

11:45

5aAB13. Challenges of using passive acoustic monitoring for marine mammals during anthropogenic activities. Shane Guan (Office of Protected Resources, NOAA/NMFS, 1315 E-West Hwy., Silver Spring, MD 20910), Claudio Fossati, Gianni Pavan, and Giovanni Caltavuturo (Univ. of Pavia, Via Taramelli 24, 27100, Pavia, Italy)

Monitoring the presence of marine mammals in the vicinity of an anthropogenic activity using passive sonar can greatly improve detection rates by visual monitoring, and it is the only way to detect marine mammals at large distances during nighttime monitoring. Therefore, passive acoustic monitoring (PAM) is sometimes required by regulatory agencies as a mean to supplement visual monitoring during anthropogenic activities that may potentially adversely affect marine mammals. However, there are many critical aspects that need to be taken into consideration when prescribing PAM to support mitigation measures. These challenges include (1) training for shipboard observers to operate and maintain sophisticated PAM hardware and software; (2) proper design of PAM system that works well during industrial operations (such as seismic vessels); (3) reliable bearing and ranging of calling of animals, thus providing basis for mitigation measures; and (4) the affordability of PAM system to small businesses. This presentation provides a comprehensive analysis on the above aspects that are essential for marine mammal passive acoustic monitoring during anthropogenic activities, and highlights future needs to improve and expands PAM as a standard technique to support mitigation measures to reduce anthropogenic impacts.

Session 5aBA

Biomedical Acoustics and Physical Acoustics: Photons and Phonons: Diagnostic and Therapeutic Applications I

Parag V. Chitnis, Cochair
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Ronald Silverman, Cochair
Riverside Research Inst., F. L. Lizzi Center for Biomedical Engineering, 156 William St., New York, NY 10038

Invited Papers

8:00

5aBA1. Photoacoustic tomography: Ultrasonically breaking through the optical diffusion limit. Lihong V. Wang (Dept. of Biomedical Eng., Washington Univ., Campus Box 1097, One Brookings Dr., St. Louis, MO 63130-4899, lhwang@wustl.edu)

Photoacoustic tomography (PAT) has been developed for functional and molecular imaging by physically combining optical and ultrasonic waves via energy transduction. Key applications include early-cancer and functional imaging. Light provides rich contrast but does not penetrate biological tissue in straight paths as x-rays do. Consequently, high-resolution pure optical imaging (e.g., confocal microscopy, two-photon microscopy, and optical coherence tomography) is limited to depths within one optical transport mean free path ($\sim$1 mm in the skin). Ultrasonic imaging, on the contrary, provides good image resolution but suffers from poor contrast in early-stage tumors as well as strong speckle artifacts. PAT—embodied in the forms of computed tomography and focused scanning—overcomes the above problems because ultrasonic scattering is $\sim$1000 times weaker than optical scattering. In PAT, a pulsed laser beam illuminates the tissue and generates a small but rapid temperature rise, which induces emission of ultrasonic waves due to thermoelastic expansion. The short-wavelength ultrasonic waves are then detected to form high-resolution tomographic images. PAT broke through the diffusion limit for penetration and achieved high-resolution images at depths up to 7 cm in tissue. Further depths can be reached by thermoacoustic tomography using microwaves or rf waves instead of light for excitation.

8:20

5aBA2. Functional-anatomical imaging of tumors and vasculature using optoacoustic-ultrasonic system. Alexander Oraevsky (TomoWave Labs., Inc., 675 Bering Dr., Ste 575, Houston, TX 77057, aao@tomowave.com)

This lecture will discuss the current status and prospectives of optoacoustic imaging and advantages of combined optoacoustic-ultrasonic imaging systems and their biomedical applications. Optoacoustic system provides optical contrast in tissue while mapping tissue structures with ultrasonic resolution. The main advantage of imaging using optical contrast is the possibility to map distribution of blood concentration and its oxygen saturation (functional imaging) in the vasculature or other tissue of interest (such as malignant tumors). High optical contrast of blood (specifically hypoxic blood) makes visualization of the tumor angiogenesis possible, thereby providing functional information for differentiation of malignant and benign tumors. The idea that drives present developments in the field of optoacoustic imaging is that coregistration of optoacoustic and ultrasonic images would be a useful enhancement of almost every application of medical ultrasound.

8:40

5aBA3. Noninvasive optoacoustic platform for multiparameter patient monitoring. Rinat O. Esenaliev, Yuriy Petrov, Irene Y. Petrov (Ctr. for Biomedical Eng., Univ. of Texas Medical Branch, 301 University Blvd., Galveston, TX 77555-1156, riesenal@utmb.edu), and Donald S. Prough (Univ. of Texas Medical Branch, Galveston, TX 77555-0591)

Our objective is to improve patient care by developing a noninvasive, optoacoustic diagnostic platform for accurate and continuous monitoring of important physiological parameters including total hemoglobin concentration, venous oxyhemoglobin saturation (both cerebral and mixed), cardiac output, circulating blood volume, cardiac index, systemic oxygen delivery, and hepatic function. Currently, invasive measurements of these parameters are routinely used in the care of large populations of patients. We built optoacoustic systems for monitoring of these parameters and tested them in animal and clinical studies. The systems include portable, light-weight, inexpensive, pulsed laser diodes, or tunable OPOs operating in the near infrared spectral range. We developed patient interfaces with highly sensitive, wide-band optoacoustic probes designed and built in our laboratory for these diagnostic applications. In some studies, ultrasound imaging systems were used for accurate optoacoustic probing of specific blood vessels. A software package was developed for automatic, real-time, continuous monitoring using measurements at different wavelengths. We will report results of the animal and clinical tests performed by our group to study the capabilities of the optoacoustic platform for noninvasive monitoring of these physiological parameters. [Dr. Esenaliev and Dr. Prough are co-owners of Noninvasix, Inc., a UTMB-based startup that has licensed the rights to optoacoustic monitoring technology.]

9:00

5aBA4. Combined application of photoacoustic and acousto-optic imaging for model-free quantitative optical absorption mapping. Wiendelt Steenbergen, Robert Molenaar, and Khalid Daoudi (Biomedical Photonic Imaging Group, MIRA Inst., Univ. of Twente, P.O. Box 217, NL-7500 AE Enschede, The Netherlands, w.steenbergen@utwente.nl)

Quantitative absorption mapping is a major challenge in photoacoustic tissue imaging due to the unknown local fluence inside the tissue. We present a photoacoustic method that is inherently quantitative and does not rely on a model for light transport. It is a well-designed combination of photoacoustic imaging and acousto-optic modulation, where the latter provides information regarding the fluence if certain rules are followed. In our method, photoacoustic (PA) imaging is performed with successive light injection in the tissue at two different positions on the tissue surface. Next an acousto-optic (AO) scan is performed with light injection in the first position, and detection of modulated light at the second position. We will show how the resulting two PA scans and the AO scan can be combined to yield the local absorption coefficient. Here we invoke the principle that in turbid media, light trajectories can be traveled in both directions with equal probability. We will give a brief outline of the theory, leading to an expression for the local absorption coefficient in terms of PA and AO measurements and geometrical parameters of the instrument. We will show results of a validation with the Monte Carlo simulation model and of experiments on phantoms.

9:20

5aBA5. Coupled contrast for photoacoustic imaging and detection. Matthew O'Donnell (Dept. of Bioengineering, Univ. of Washington, Seattle, WA 98195-2180)

Nanoparticle agents with high optical absorption in the near infrared have been used to target diseased cells and increase specific contrast in photoacoustic (PA) imaging. However, background signals from tissue can severely limit the specificity of these agents. This is especially problematic for detecting targeted cells circulating in the vasculature where the large PA signal from blood can completely mask the contrast agent signal. For sensitive detection of targeted cells in the vasculature, their PA signal must be greatly enhanced compared to the blood background. Using a new class of contrast agents called coupled particles, we have developed technologies that can potentially accumulate and concentrate targeted cells while simultaneously enhancing their specific contrast compared to background signals. This approach leverages the high optical absorption and magnetic properties of Au-shell-encapsulated magnetic nanoparticles. By manipulating these coupled particles with an applied magnetic field, targets can be accumulated within the flow field and differentiated photoacoustically through motion processing algorithms providing high contrast specificity. In this talk we will discuss the basic coupled agent technology, procedures for manipulating these agents for cellular accumulation and differentiation, and challenges remaining to translate this technology into clinical use.

9:40—10:00 Break

Contributed Papers

10:00

5aBA6. Acoustical method of cleaning lungs. Sanford Hawkins and Andrew Morrison (DePaul Univ., Chicago, IL 60614 shawkin@mac.com)

The muco-ciliary system in the lungs uses cilia asymmetrically beating at 16 Hz to remove lung secretions. As lung secretions are non-Newtonian fluids with a critical frequency between 16Hz and 20HZ, vibrating the lung at 16 thins the secretions. A unique 16 Hz user-powered sound generator, which recently received FDA approval for diagnosis and treatment of respiratory diseases, will be described. The device provides a simple and effective means of acoustically driving the muco-ciliary system in the appropriate frequency range and may possibly synchronize the cilia for improved efficiency. The device may be the first effective way to treat the third leading cause of death. Additionally, a new and more efficient way to cough will be demonstrated.

10:15

5aBA7. Optical multiplexed operation of nanomechanical systems. A. Sampathkumar, K. L. Ekinci (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215), and T. W. Murray (Univ. of Colorado, Boulder, CO 80309)

An emerging application area for nanoelectromechanical systems (NEMS) is in the next generation of biological and chemical sensors. One of the key technological challenges in the development of sensors using large-scale arrays of NEMS beams lies in designing a measurement system with multiplexing capability to detect simultaneously the flexural motion of a large number of nanomechanical beams. In the latest work, a full-field optical interferometry system making use of a photorefractive crystal is developed to obtain the displacements of the NEMS array. In this system, a coherent imaging approach using an adaptive photorefractive holography technique and a parallel detection scheme employing a charge-coupled device is used to obtain full-field displacement images of the entire NEMS array. Full-field out-of-plane displacement images of an array of 60 nanomechanical beams excited using an optical transduction scheme are presented. The full-field system is characterized in terms of both efficiency of excitation of nanomechanical vibration and sensitivity of the interferometric displacement detection and is found to be well suited for the measurement of vibrations in large-scale NEMS array.

10:30

5aBA8. Acousto-optic imaging using a powerful long pulse laser and digital holography. Emilie Benoit, Salma Farahi, Emmanuel Bossy, and François Ramaz (Institut Langevin, ESPCI ParisTech,10 rue Vauquelin, F-75231 Paris Cedex 05, France, emilie.benoit@espci.fr)

Acousto-optic tomography is a technique that couples ultrasounds and light in order to measure local optical properties through thick and highly scattering media, e.g. human breast tissues. Thanks to the acousto-optic effect, we can get the optical contrast information given by light and get the spatial localization from the ultrasound longitudinal waves. The use of a powerful long pulse laser (1 ms, 200 mJ) helps to improve the sensitivity of the technique by raising the optical peak power and collect more ultrasound tagged photons. Moreover, its tunable wavelength around 780 nm is appropriate for biological imaging since it is where light has its maximum penetrating depth in tissues. Detection of acousto-optic signals is done by off-axis heterodyne digital holography on a high speed CMOS camera. This enables us to make a tunable spatio-temporal filter with a high signal to noise ratio. Experimentally, this technique enables to image optical absorbers embedded within a thick scattering media (a few centimetres). It is also theoretically possible to use a conventional pulsed ultrasound scanner to generate acousto-optic signals that can at the same time record ultrasound images, which will be a step toward multimodal imaging for breast cancer detection.

10:45

5aBA9. Investigating new photoacoustic contrast agents. Kamyar Firouzi (Dept. of Mech. Eng., Univ. Coll. London (UCL), Torrington Pl., London WC1E 7JE, United Kingdom, kamyar.firouzi.09@ucl.ac.uk), Eleanor Stride, and Nader Saffari (Univ. Coll. London (UCL), London WC1E 7JE, United Kingdom)

Numerical studies on the detection of tumors using photoacoustic imaging have shown that relying on optical energy absorption by blood hemoglobin as the contrast mechanism leads to poor image quality. This provides a strong incentive for seeking suitable manufactured photoacoustic contrast agents (PACAs). We present a theoretical comparison of three different photoacoustic contrast agents, all incorporating dye particles with suitable optical absorption characteristics. The three different designs are as follows: (1) stabilized droplet: a droplet of dye; (2) stabilized bubble: a bubble filled with gas and coated by dye as the shell; and (3) phase shift droplet: a droplet of volatile dye, all pulsating in a homogenous incompressible fluid. For each case, the governing equations describing the dynamics of a single PACA and the radiated pressure are derived. Each case is expressed in terms of four coupled sets of equations as the pressure radiated from the bubble, equation of motion, photoacoustic energy equation and equation of state, by virtue of simplifying assumptions. The derived equations are numerically solved by the Runge–Kutta method, using appropriate photoacoustic properties. The numerical results predict a much stronger radiated acoustic signal for the same optical source energy in the case of the stabilized bubble (case 2).

11:00

5aBA10. Detection of macrophages and lipid using ultrasound guided spectroscopic intravascular photoacoustic imaging. Bo Wang (Dept. of Biomedical Eng., Univ. of Texas at Austin, Austin, TX 78712), Pratixa Joshi (Univ. of Texas at Austin, Austin, TX 78712), Nadine Matthias, James Amirian (Univ. of Texas Health Sci. Ctr., Houston, TX 77030), Silvio Litovsky (Univ. of Alabama Birmingham, Birmingham, AL 35249), Konstantin Sokolov (Univ. of Texas, Houston, TX 77030), Richard Smalling (Univ. of Texas Health Sci. Ctr., Houston, TX 77030), and Stanislav Emelianov (Univ. of Texas at Austin, Austin, TX 78712)

Macrophages and lipid are two key components in the development and characterization of atherosclerotic plaques. Imaging the activity and distribution of macrophages and lipid in the vessel wall would help identify the vulnerable plaques. In this study, spectroscopic intravascular photoacoustic (sIVPA) imaging was used to detect phagocytically active macrophages and deposits of lipid simultaneously. Polyethylene glycol coated spherical gold nanoparticles (Au NPs) were intravenously injected *in vivo* into an atherosclerotic rabbit allowing phagocytically active macrophages within the plaques to be labeled with Au NPs. Atherosclerotic aorta was harvested and scanned with a bench-top sIVPA imaging system. Au NP-labeled macrophages were detected in the 710–770 nm wavelength range because of the plasmon resonance coupling effect. Lipid was imaged in 1210–1230 nm wavelength range based on the characteristic optical absorption spectrum of fatty acids. An image processing method was developed to identify Au-NP-labeled macrophages and lipid deposits from the multiwavelength IVPA data. Coregistered sIVPA images were combined with corresponding IVUS images to demonstrate the location of Au NP-labeled macrophages and lipid within the vessel wall. Finally, histochemistry stains confirmed that combined sIVPA and IVUS imaging can successfully detect the distribution of phagocytically active macrophages and lipid deposits.

11:15

5aBA11. Simultaneous high-frequency ultrasound and optoacoustic imaging of *in vivo* mouse embryos. Parag V. Chitnis (F. L. Lizzi Ctr. for Biomedical Eng., Riveside Res. Inst., 156 William St., 9th Fl., New York, NY 10038, pchitnis@rri-usa.org), Orlando Aristizábal (New York Univ. School of Medicine, New York, NY), Jonathan Mamou (Riverside Res. Inst., New York, NY), Daniel H. Turnbull (New York Univ. School of Medicine, New York, NY), and Jeffrey A. Ketterling (Riverside Res. Inst., New York, NY)

An integrated optoacoustic (OA) and high-frequency ultrasound (HFU) system that provides detailed images of anatomical structure and molecular contrast in small animals and transgenic embryos is presented. Volumetric imaging was achieved by raster scanning a vertically oriented imaging probe. The imaging probe was a five-element, 40-MHz, PVDF-TrFE-based annular array. A PBS-filled Petri-dish with a center hole was placed on the abdomen of an anesthetized mouse, and a laparotomy was performed to expose an intact uterus. A bifurcated beam from a 532-nm pulsed laser illuminated the embryos from opposing directions normal to the image plane. The central element of the array was excited with a high-voltage impulse synchronized with the light pulse. The resulting ultrasound echo and OA signals from each scan location were digitized from all five array channels and postprocessed. The anatomy of the embryonic head (HFU image) was coregistered with the embryonic vasculature (OA image). Delay-and-sum beamforming was performed and the resulting HFU and OA images exhibited a notable improvement in the depth of field and signal-to-noise ratio in comparison to the images constructed using signals acquired from the central array element alone. The feasibility of real-time, spatially coregistered, dual-modality *in vivo* imaging of mouse embryos was demonstrated.

11:30

5aBA12. Design and applications of photoacoustic nanodroplets in imaging and therapy. Katheryne Wilson, Alexander Hannah, Kimberly Homan, and Stanislav Emelianov (Dept. of Biomedical Eng., Univ. of Texas at Austin, Austin, TX 78712, emelian@mail.utexas.edu)

A unique contrast agent, termed photoacoustic nanodroplet (PAnD), has been developed for several biomedical and clinical applications. A PAnD consists of several plasmonic nanoparticles encapsulated within a nanosized perfluorocarbon droplet. PAnDs act as contrast agents for ultrasound and photoacoustic imaging through different mechanisms. Optical absorption by plasmonic nanoparticles triggers vaporization of the perfluorocarbon droplet, yielding two effects: first, strong photoacoustic transients are generated, and second, a gas microbubble is formed. Furthermore, subsequent laser pulses interacting with expelled plasmonic nanoparticles result in photoacoustic transients due to thermoelastic expansion of tissue. To demonstrate these contrast enhancement mechanisms, a homogenously laden polyacrylamide phantom (108 PAnD/ml) underwent pulsed laser irradiation (5 mJ/cm^2, 10 Hz PRF, 5–7 ns pulse duration), while photoacoustic and ultrasound images were captured with a 40 MHz linear array transducer. Photoacoustic signal from vaporization was one to two orders of magnitude stronger than that from thermoelastic expansion, and ultrasound contrast due to presence of gas microbubbles was significantly increased. In addition to serving as contrast agents, PAnDs with applied ultrasound irradiation cause temporary disruptions of the cell membrane and increase intracellular uptake of nanoparticles, thus improving accumulation and retention of nanoparticles within targeted tissue. A sevenfold increase in cellular uptake of nanoparticles has been demonstrated.

Session 5aEA

Engineering Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Animal Bioacoustics: Acoustical Sensor and Array Technology I

Dehua Huang, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841-1708

Thomas R. Howarth, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841-1708

Invited Papers

7:50

5aEA1. Comparison of design methods to achieve wideband operation in tonpilz transducers. Michael B. Wilson, Stephen C. Thompson, and Thomas B. Gabrielson (The Appl. Res. Lab., The Penn. State Univ., P.O. Box 30, State College, PA 16804)

Wide sonar bandwidth with advanced processing enables improved system capabilities in sensors and arrays. However, transmitting and receiving bandwidth greater than one octave remains an emerging technology. Multiple resonances and high coupling materials are now viable options to increase the bandwidth of a device, but there is no study to date of a multiply resonant device that uses high coupling material. A set of guidelines is presented for the design of elements for wideband sonar arrays having bandwidth greater than one octave using multiple resonances, high coupling materials, or some combination.

8:10

5aEA2. Low frequency acoustic sensor or array calibration waveguides of finite length. Dehua Huang and Anthony Paolero (NAVSEA Newport, Newport, RI 02841)

Calibration acoustic transducers at very low frequencies in confined and well understood environment require special equipment. The Underwater Sound Reference Division has three cylindrical tubes for low frequency calibration. This paper addresses a rectangular cross section of finite length waveguide as calibration environment. The acoustic fields in the new waveguide for standing and traveling waves are both analyzed. The mathematical model and the numerical simulation results will also be presented. [This work is supported by the U.S. Navy.]

8:30

5aEA3. Practical and mathematical aspects of pairing microphones for hearing aid directional arrays. Daniel M. Warren and Charles B. King (Knowles Electrons, 1151 Maplewood Dr., Itasca, IL 60143)

Directional microphone arrays in hearing aids are short baseline, end-fire arrays, which require tight matching of response characteristics between the microphones, as previously reported [J. Acoust. Soc. Am. **113**(4,) 2219 (2003)]. Pairing tolerances for good directional performance are much tighter than normal manufacturing tolerances on individual microphones, so it is infeasible to produce entire production lots within these tolerances. Instead, microphones are selected in pairs based on their measured performance. In some cases, the pairing tolerance is on par with the measurement uncertainty of even a well-designed manufacturing test system. Standard gauge analysis is not well suited to this pairing measurement. An analysis of acoustical measurement error as it relates to the pairing problem, the statistics of successful pairing from a large quantity of microphones, and the underlying graph theory of optimum pairing will be discussed in this paper.

Contributed Papers

8:50

5aEA4. Performance of tourmaline hydrophones at low- and midfrequencies. Juan Morales, Akash Kale, and Mardi Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Tourmaline hydrophones are well suited for measurement of underwater blasts and other impulsive type sounds having extremely fast rise times and high peak pressures. The physical characteristics of these small, lightweight sensors, however, make them an attractive choice for other applications. To determine their performance at low- and midfrequencies, substitution calibrations of two tourmaline sensors with a calibrated laboratory hydrophone were completed using a J-ll underwater sound transducer in a 100-gal tank. The calibration signals consisted of tone bursts containing 5–20 cycles at frequencies ranging from 250 to 10 000 Hz and having sound pressure levels from 110 to 180 dB *re* 1 μa. Results indicate that a tourmaline hydrophone coupled to an in-line charge amplifier and preamplifier to provide fixed voltage gains ranging from 100× to 10 000× has excellent performance over this range of frequencies. [Work supported by the National Oceanographic Partnership Program (NOPP) through ONR Award No. N000140710992.]

9:05

5aEA5. Design of an ideal spherical hydrophone using a plasmonic acoustic cloak. Matthew D. Guild, Michael R. Haberman (Dept. of Mech. Eng., and Appl. Res. Labs. The Univ. of Texas at Austin, Austin, TX 78712), and Andrea Alu (The Univ. of Texas at Austin, Austin, TX 78712)

An ideal acoustic sensor provides an electrical output that is proportional to and in phase with an incident acoustic pressure field, without significantly altering the impinging field. The common approach to producing such a sensor is simply to minimize the size of the sensing elements relative to the wavelength of the measured field. Unfortunately, this approach has the drawback of reducing sensitivity. Recent work on plasmonic acoustic cloaking [Guild *et al.*, J. Acoust. Soc. Am. **128**, 2374 (2010)], however, suggests the possibility of creating an ideal acoustic sensor with dimensions on the same length scale as the incident wavelength, obtained by drastically sup-

5a FRI. AM

pressing its scattering without affecting its ability to measure the impinging signal. Unlike other cloaking methods that reroute the incident field around the cloaked object, plasmonic cloaks allow the cloak interior to interact with the incident field, thereby permitting the realization of highly noninvasive acoustic sensors. This work presents an investigation on the application of a plasmonic acoustic cloak for spherical ceramic hydrophone shells. The resulting scattering reduction and the effects on hydrophone sensitivity and bandwidth due to presence of the cloak are analyzed and discussed.

9:20

5aEA6. Locating multiple incoherent sound sources in three-dimensional space. Na Zhu and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detoit, MI 48202)

A new technology for locating incoherent sound sources in three-dimensional space is developed. The underlying principle of this technology is a model based approach that assumes acoustic radiation from a point source in free space, together with triangulations and signal pre-processing to enhance the signal to noise ratio. A prototype developed for this technology consists of six microphones, multi-channel high-accuracy data acquisition module, and a web camera. This device allows for capturing and visualizing target sources in three ways: (1) through the camera viewing angle that shows the actual sources and their Cartesian coordinates directly, (2) through projection of the source positions on a horizontal plane and their traces as they move in space, or (3) automatically switching between the camera and top view displays, depending on whether the target falls inside the viewing angle of the camera or outside. To ensure a real-time display of the results, four microphones are used. The disadvantage of using four microphones is that the accuracy in source localization may be reduced when the input data are contaminated by interfering and background signals. Under this condition, redundancy checks using six microphones can be used to enhance the accuracy and spatial resolution of source localization.

9:35

5aEA7. Distributed vertical line array receiver. Peter F. Worcester, Matthew A. Dzieciuch, Lloyd L. Green, David D. Horwitt, Jacques C. Lemire, Scott D. Carey, and Matthew Norenberg (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA 92093-0225, pworcester@ucsd.edu)

A distributed vertical line array (DVLA) receiver able to span the water column in water up to 6000 m deep has been developed to allow both modal and ray-based analyzes of acoustic propagation. The DVLA is made up of distributed, self-recording hydrophones with timing and scheduling provided by a small number of central controllers, called D-STARs. The enabling technologies for this approach are (i) flash memory modules that can store gigabytes of data in a small pressure case at each hydrophone and (ii) inductive modems that allow low-bandwidth communication between the D-STAR controllers and the hydrophone modules over standard oceanographic mooring wire for control and time synchronization. The DVLA consists of sub-arrays with a nominal length of 1000 m. The hydrophone modules are clamped to the mooring wire during deployment, making the DVLA readily configurable. It is navigated using acoustic transponders on the seafloor. The hydrophone modules make precision temperature measurements to provide the sound-speed profiles needed for beamforming. A DVLA consisting of two 1000-m sub-arrays, one spanning the sound-channel axis and the other spanning the surface conjugate depth, was successfully deployed in the Philippine Sea for 1 month during spring 2009. [Work supported by the Office of Naval Research.]

9:50—10:00 Break

10:00

5aEA8. A virtual receiving array using a time-reversal chaotic cavity. Jae-Wan Lee and Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 262 Seongsanno, Seodaemun-gu, Seoul 120-749, Korea)

It has been demonstrated that a virtual transmitting array can be constructed using a small number of transducers glued to a chaotic cavity (or a random reverberator) in conjunction with time reversal acoustics [Montaldo *et al.*, IEEE Trans. Ultrason., Ferroelec., Freq. Contr. **52**, 1489–1497 (2005)]. However, a virtual receiving array based on the concept has been neither realized nor reported yet. This paper presents an underwater virtual receiving array using a time-reversal chaotic cavity. The prototype array consists of a single immersion transducer and a chaotic cavity and is made to have a total of 8×8 virtual receiving elements. The performance of the virtual array is compared with a commercial 2-D hydrophone array. [Work supported by the National Research Foundation of Korea Grant funded by the Korean Government (MEST) (NRF-2009- 0077588).]

10:15

5aEA9. Horns as particle velocity amplifiers. Dimitri M. Donskoy (Dept. of Ocean Eng., Stevens Inst. of Tech., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu) and Benjamin Cray (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI, 02841)

Horns are simple but effective means for improved performance of acoustical systems. Nowadays they are primarily used for sound emission (in loudspeakers) providing better impedance matching between the source and the medium. Applications of receiving horns as sound amplifiers are dated back to the beginning of the last century with the advent of telephones. At the present time, however, highly sensitive pressure sensors such as microphones and hydrophones rarely require additional amplifications for common applications. Recent progress in the development of the acoustic particle velocity sensors opened up many new opportunities in their utilization in air and fluids. Compared to the pressure sensors, however, the particle velocity sensors are less sensitive especially at low frequencies and may benefit from additional amplification provided by horns. Here we revisit theory and numerically analyze horn performance as an amplifier of particle velocity rather than impedance matching device or pressure concentrator.

10:30

5aEA10. Analytical comparison of spherical multimicrophone probes. Curtis P. Wiederhold (Dept. of Mech. Eng., Brigham Young Univ., 435 Crabtree Bldg., Provo, UT 84602, curtis.wiederhold@gmail.com), Kent L. Gee, Scott D. Sommerfeldt, and Jonathan D. Blotter (Brigham Young Univ., 435 Crabtree Bldg., Provo, UT 84602)

Probes with microphones flush-mounted in hard spheres have been used to measure energy-based acoustic quantities by estimating the pressure gradient using finite-difference approximations. The number of microphones, their orientation in the sound field, and the frequency content of an incident acoustic wave affect the probe's accuracy. This study analytically compares errors in measuring acoustic intensity and energy density of plane waves incident on ideal point microphones located on a rigid sphere. The difference between implementing such a probe as a series of three orthogonal 1-D probes or using a least-squares approximation using all microphones is considered. Estimation errors as a function of incidence angle and frequency have been calculated, thereby providing insight into the relative merits of each probe configuration.

10:45

5aEA11. Experimental analysis of multimicrophone probes for measurement of rocket noise. Jarom H. Giraud, Kent L. Gee, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., 453 Crabtree bldg., Provo, UT 84602)

Although the use of multimicrophone acoustic probes can be useful in characterizing rocket noise source regions, one challenge is that both high-bandwidth pressure and intensity data are desirable. The high-frequency bandwidth limitations arise from the microphone size, type, orientation, spacing, and the design of preamplifier holders. To explore their broadband responses, various tetrahedral four-microphone probes were placed in the far field of a loudspeaker and rotated in an anechoic chamber to determine their pressure magnitude and intensity magnitude/angle errors as a function of orientation and frequency. Results are presented comparing the advantages and limitations of each design, particularly as they apply to obtaining rocket

noise data. One probe design considered consists of microphones flush-mounted on a hard sphere. Other designs consist of sets of microphones arranged in tetrahedral configurations with their preamplifiers set at different angles.

11:00

5aEA12. Diver monitoring using the Hawaii Experimental Acoustic Range. Tom Fedenczuk and Eva-Marie Nosal (Ocean Resources Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822)

We will present the preliminary data and results from the first full stage experiment conducted with the newly built Hawaii Experimental Acoustic Range in Oahu, HI. The range is a multipurpose facility that allows for various receiver configurations, hydrophone models, and locations. Data access and control is permitted in three modes: shore/ship, autonomous, or underwater observatory, with a custom designed 16 channel, 24 bit, 192 kHz simultaneous sampling data acquisition system. For this experiment, the receiver configuration included two volumetric arrays of five hydrophones each, and four additional individual hydrophones. The purpose of the experiment was to detect and track divers in a shallow water harbor environment. Two divers were deployed along predefined paths. The divers stopped at control points that were georeferenced by a survey grade total station. A low amplitude transducer was activated at each control point and transmitted a single 20–40kHz sweep to calibrate the tracking algorithms. This talk will present details of the range, experiment, and preliminary results.

11:15

5aEA13. Tonpilz type underwater vector sensor with directional sensitivity to acoustic waves. Yongrae Roh and Youngsub Lim (School of Mech. Eng., Kyungpook Natl. Univ., Daegu 702-701, Korea, yryong@knu.ac.kr)

Typical Tonpilz-type underwater acoustic transducers making use of piezoceramics detect the magnitude of an acoustic pressure, a scalar quantity, and convert this pressure into a proportional output voltage. The scalar sensor has no directional sensitivity. In this paper, a new Tonpilz transducer structure is proposed to measure both the magnitude and the direction of an incoming acoustic wave with a single transducer unit, which is accordingly referred to as a vector sensor. The piezoceramic stack clamped between the head mass and the tail mass of a Tonpilz transducer is divided into four radial segments, each segment polarized alternately to its neighboring one. An acoustic pressure wave coming from outside causes positive or negative signed electric voltage signals in the piezoceramic segments in accordance with their polarization directions as well as a certain amount of time delay between the electric signals. Proper manipulation of the electric signals, i.e., pairing the segments and either adding or subtracting the signals of the paired segments, can provide the relationship between the signals and the magnitude and direction of the acoustic wave. Feasibility of this structure is confirmed through 3-D simulations of the receiving properties of the transducer with the finite element method.

11:30

5aEA14. Investigation of continuous scanning laser Doppler vibrometry for non-contact measurement of linear and angular surface deflections. Muhammad Salman and Karim Sabra (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

Laser Doppler vibrometry (LDV) is a non-contact technique for sensing surface vibrations. Traditionally LDV uses one or several fixed beams to measure vibrational velocity of specific points and orientations. Measurement of angular velocity requires at least two beams. Instead, we develop a technique, continuous scanning laser Doppler vibrometry (CSLDV), using a single laser beam continuously sweeping the area of interest using a scanning mirror. Linear scans allow the measurement of normal and angular velocities while circular scans allow the measurement of normal and two angular velocities. We validated the CSLDV technique for measuring low frequency (less than 50 Hz) broadband vibrations of gel samples which mimics natural vibration of human body. Such system could potentially be used to monitor multiple DOF of the skin surface for haptic or remote control applications. There is a disadvantage of using CSLDV which is speckle noise that is generated when coherent light source is reflected back from an optically rough surface. We will discuss the effects of scan lengths, scanning frequency, target to sensor distance, and the excitation amplitude on the performance of CSLDV.

11:45

5aEA15. An aeroacoustic microelectromechanical systems microphone phased array. Drew Wetzel, Chris Bahr, Matthew Williams, Jessica Meloy, Mark Sheplak, and Louis Cattafesta (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, P.O. Box 116250 Gainesville, FL 32611-6250)

Phased microphone arrays are useful tools for noise source localization using a process known as beam-forming. In scale-model wind tunnel experiments, the frequency range of interest can extend as high as 90 kHz. In both open and closed wall wind tunnels, microphones with high dynamic range are required to sense large turbulent pressure fluctuations from the open jet shear layer and the tunnel wall boundary layer. Microphones that meet the frequency and dynamic range demands of such experiments are readily available but expensive. When considering the high-sensor counts typically needed for phased array measurements, total sensor cost can be a limiting factor. The presentation will discuss a proof-of-concept phased array consisting of 25 piezoelectric microelectromechanical systems (MEMS) microphones arranged in a log-spiral pattern on a single printed circuit board. The microphones were designed in-house and have a dynamic range from 40–160-dB SPL and possess a resonant frequency greater than 100 kHz. A proven MEMS-based array that leverages the benefits of batch fabrication could cost significantly less than a traditional equivalent. The MEMS array will be characterized in the UF Aeroacoustic Flow Facility and compared to a conventional array of the same pattern comprised of 0.25-in. high-frequency microphones.

Session 5aED

Education in Acoustics and Physical Acoustics: Tools for Teaching Advanced Acoustics

Kent L. Gee, Cochair
Brigham Young Univ., Dept. of Physics and Astronomy, Provo, UT 84602

Scott D. Sommerfeldt, Cochair
Brigham Young Univ., Dept. of Physics and Astronomy, Provo, UT 84602

Chair's Introduction—8:00

Invited Papers

8:05

5aED1. An information-rich learning environment for instruction in acoustics. Robert D. Celmer (Dept. of Mech. Eng., Acoust. Program and Lab., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117)

The static written word has always had its limits when it comes to learning about sound. Much of the subject matter is dynamic, multifaceted, and of course, aural. This presentation will describe multimedia materials developed for in-class presentation and self-paced review exercises for acoustics instruction at the University of Hartford. Some of the materials were developed using certain authoring applications, draw and animation programs, sound manipulation software, as well as 3-D-CAD and spectral analysis applets. Audio equipment used in class, as well as acoustic treatments of the classroom/listening environment, will be described. This approach to acoustic pedagogy will be discussed in the context of a student-centered learning environment. Demonstrations of the materials for the instruction of acoustical concepts, as well as case studies, will be presented.

8:20

5aED2. Using interactive simulations to visualize Fourier analysis. Wendy K. Adams (Dept. of Phys., Univ. of Northern Colorado, CB 127, Greeley, CO 80639, wendy.adams@colorado.edu)

Fourier analysis is an elegant and powerful method for expressing general functions as a sum of simpler trigonometric functions. In this presentation the PhET sim "Fourier: Making Waves" http://PhET.colorado.edu/en/simulation/fourier, will be presented including the research behind the simulation, how students react to the sim, and ideas for use in class. Students typically learn the math needed to do Fourier transforms and learn how to express a function in time or space and in terms of wavelength, wave number, or mode. However, many of these relationships are only memorized for the short term (exam) and are not retained. This simulation is designed to help students visualize how a combination of simple sines and cosines can create a more complicated function and listen to the sounds produced by each harmonic. They can explore each of the symbols lambda, T, k, omega, and n to learn what each represents on the graph and their relationships with one another. There is also a game tab with ten different levels that challenges students to choose the correct harmonics to match more and more complicated functions. Finally there is a tab to help students visualize moving from a discrete to a continuous series.

8:35

5aED3. Advanced acoustic demonstration videos for higher education: Longitudinal wave motion. Robert Astrom (Astrom Acoust., 8045 Clark's Chapel, Athens, OH 45701, bob@astromacoustics.com)

In education every subject requires a unique instructional approach. Unlike English or mathematics, science based classes such as chemistry and physics cannot be taught effectively with lesson plans based solely on reading and homework assignments. These subjects require active learning through demonstrations and/or hands on experiments. That said some valuable experiments are too dangerous or costly to be feasible in all programs. The intent of this project is to advance the knowledge of and interest in acoustics by providing educators with interesting and educational videos of science based acoustic demonstrations and experiments which fit these situations. One of the most difficult concepts for educators to teach and students to understand is longitudinal wave motion. The videos presented are designed to illustrate longitudinal wave propagation using what many would consider dangerous or costly experiments. In keeping with the spirit of cost control the series of videos are to be provided to educators free of charge.

8:50

5aED4. Teaching room modes and diffraction using COMSOL MULTIPHYSICS. Ralph T. Muehleisen (Civil, Architectural and Environ. Eng., Illinois Inst. of Technol., Chicago, IL 60616, muehleisen@iit.edu)

The concepts of room modes and diffraction are fairly easy to explain in a qualitative way but the math involved to develop the physics of both can be difficult. When explaining the concept of room modes in more detail, many educators start with the idea of a rectangular room with rigid boundaries to get analytic solutions. However, convincing students that room modes exist in rooms of other

shapes is not always easy since analytic solutions cannot be found. Similarly, diffraction is easy to explain as a concept but the functions involved in mathematically explaining diffraction are advanced and difficult to understand. In order to get meaningful solutions to both room mode and diffraction, some researchers and educators turn to finite element software. In this paper, the use of Comsol Multiphysics, a general purpose finite element program, in teaching room modes and diffraction, is shown. COMSOL can be used to draw rooms of an arbitrary shape and generate resonance frequencies and room modes in a matter of minutes. COMSOL can also be used to generate animations of sound waves diffracting around objects. Examples of both uses will be shown.

9:05

5aED5. Time domain visualization of uniform spherical waves in a cavity using the method of wave images. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338, jerry.ginsberg@me.gatech.edu)

The method of wave images uses the principle of superposition in conjunction with d'Alembert's solution of the wave equation to construct time domain solutions. The formulation has been applied to describe the response in 1-D rectilinear waveguides for nonzero initial conditions [J. H. Ginsberg, J. Acoust. Soc. Am. **119**, 1954–1960 (2006)] and forced motion at a boundary [J. H. Ginsberg, POMA **6**, 025002 (2009)]. The present work extends the concept to describe the interplay of point-symmetric converging and diverging spherical waves in a cavity. It is shown that the responses to nonzero initial conditions, a source at the center, and motion at the boundary may be constructed from wave images for five elementary solutions that combine converging and diverging waves. The general solutions that are derived provide useful benchmarks that upper-level students may employ to verify frequency domain modal solutions. Furthermore, the ability to visualize an acoustical signal within a cavity as the time domain superposition of propagating waves provides a different, and in some respects more physically intuitive, perspective in comparison to frequency domain solutions.

9:20

5aED6. Broadband versus narrowband experimental methods in acoustics. Joseph Gladden (Dept. of Phys., Univ. of Mississippi, University, MS 38677, jgladden@olemiss.edu)

Active acoustic measurements (those requiring the input of acoustic energy) can be broadly classified as either broadband or narrowband. In a broadband technique, the acoustic energy introduced into the system is highly localized in time (i.e., a sharp pulse). When transformed into the frequency domain using Fourier analysis, it becomes clear that this energy is spread over a wide frequency range. In a narrowband technique, the acoustic energy is introduced over a longer time period and occupies a much more localized region in the frequency domain. The limit being a single tone excitation signal. Benefits and drawbacks for each of these methods will be discussed as they are applied to several experimental systems including resonance and pulse-echo measurements.

9:35

5aED7. A first laboratory course for teaching advanced acoustics. Steven L. Garrett (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

The overwhelming majority of "structured" instructional activity in graduate schools offering advanced degrees in acoustics is spent in classes that require solution of problems sets and exams and might also include some relevant literature searches and reports. Most potential employers of these advanced degree recipients assume that their education included exposure to some hands-on measurement of sounds and vibrations followed by analysis and presentation of those measurements in an appropriate (typically graphical) format. At the very least, measurements using sound level meters, hydrophones, and accelerometers should be included along with their calibration as well as experience with transducers that includes comparison of measurements with theory. Nine laboratory exercises will be described that are required for an MS degree. These currently include characterization of electrodynamic loudspeaker parameters, Helmholtz resonators, coupled acousto-mechanical systems, sound propagation in a water-filled waveguide, shock wave development and wave-wave interactions in one dimension, determination of elastic moduli by measurement of bar resonances, reciprocity calibration, and accelerometer calibration. Instrumentation utilized in the course includes multi-channel spectrum analysis (swept-sine and FFT), lock-in amplifiers (phase-sensitive detection), function generators, and waveform capture using a digital storage oscilloscope. The importance of completing a "preliminary exercise" prior to each laboratory will be stressed.

9:50

5aED8. The song of the singing rod. Brian E. Anderson and Wayne D. Peterson (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu)

This paper discusses basic and advanced aspects of the sound radiated by the singing rod demonstration commonly used in physics courses to depict an example of longitudinal waves. Analysis of the sound radiated by various rods with small-signal and large-signal excitations is presented for four different rods. The small-signal sound radiation consists of a fundamental frequency and odd harmonics (each corresponding to a longitudinal mode) when the rod is held at its midpoint. Large-signal sound radiation is highly dependent on the rod's geometry. The large-signal sound can possess strong even harmonics and/or beating tones resulting from modal coupling of transverse bending modes and either subharmonic longitudinal modes or torsional modes. A detailed analysis of the sound radiation from a singing rod can provide excellent laboratory exercises or classroom demonstrations for advanced undergraduate or graduate level acoustics courses whose scope includes resonances of a bar.

10:05—10:15 Break

10:15

5aED9. Acoustic pulses in the atmosphere: An outdoor teaching laboratory. Thomas B. Gabrielson (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, MS 6120D, State College, PA 16804)

For the past two years, the Penn State Acoustic Data Measurements and Analysis course has incorporated several outdoor-sound exercises. One of these exercises is particularly amenable to illustrating concepts in acoustic propagation, signal design, and signal detection. Using relatively inexpensive hardware, student-designed pulse sequences are transmitted, received, and recorded in various outdoor environments. The advantages and disadvantages of large-bandwidth pulses are readily apparent. The students can perform replica correlation or matched filter processing and compare the achieved results to theoretical performance. The effects of the transmitting transducer on pulse shape are seen easily. Variability in the atmospheric propagation path, multipath, and upwind/downwind effects can be demonstrated as well as the impact of ambient noise on pulse detection. In addition, pulses can be reflected from moving objects (buses, for example) either for analyzing Doppler shift or for experimenting with Doppler-tolerant detection. Waveform types range from simple sinusoidal pulses to frequency-modulated pulses and frequency- or phase-shift coded maximum-length sequences to reproductions of animal vocalizations. In the fall of 2010, 40 students participated in this exercise: 20 resident students and 20 distance-education students.

10:30

5aED10. A pedagogical demonstration of weak-shock propagation from a gas-filled balloon explosion. Michael B. Muhlestein, Kent L. Gee (Dept. of Phys. and Astronomy, BYU, N283 ESC, Provo, UT 84602), and Jeff H. Macedone (Brigham Young Univ., Provo, UT 84602)

A classroom demonstration of weak-shock propagation theory has been developed and studied. A balloon filled with a stoichiometric mix of acetylene and oxygen is ignited, causing a near-spherical explosion. Because peak sound pressure levels at 1 m exceed 180 dB, nonlinear theory is clearly required for a proper analysis of the acoustic propagation. Weak-shock theory is first reviewed and then applied to the propagation of a simplified explosion model consisting of a shock with an exponentially decaying tail. A comparison between the model and an experiment using 3.18 mm condenser microphones reveals good agreement between theory and measurement for the nonlinear spatiotemporal wave evolution. Potential pedagogical uses for this demonstration are discussed.

Contributed Papers

10:45

5aED11. Graduate and undergraduate laboratory courses in acoustics and vibration. Aldo A. J. Glean, John A. Judge, Joseph F. Vignola, Patrick F. O'Malley, and Teresa J. Woods (Dept. of Mech. Eng., Catholic Univ., 620 Michigan Ave., Washington, DC 20064, 10glean@cardinalmail.cua.edu)

Two laboratory courses covering acoustics and vibration measurements have recently been developed in the Mechanical Engineering Department at the Catholic University of America. The first course, a junior-level dynamics laboratory, is the first mechanical engineering laboratory course taken by undergraduates. We use acoustics and vibration experiments to illustrate measurement principles broadly applicable to a variety of engineering disciplines. The second course, aimed at first year graduate students, gives a more in-depth coverage of similar topics with an emphasis on automated collection and processing of large data sets. The latter course is intended for students interested in pursuing graduate research in vibration and acoustics, but neither course assumes previous acoustics and vibration background. We describe several of the experiments conducted in these courses and relate them to specific learning objectives.

11:00

5aED12. The Lagrangian method and generalized coordinates used in graduate teaching of electromechanics and electroacoustics. David A. Brown (Elec. Eng. and Adv. Tech. and Manuf. Cntr., 151 Martine St., Fall River, MA, dbrown@umassd.edu)

While many problems of mechanics may be treated by the Newtonian method involving the analysis of the laws of motion in response to applied forces in inertial reference coordinates, it is helpful to realize that some problems may be treated more directly and with more physical insight using Hamiltonian or Lagrangian (energy based) methods and the application of generalized coordinates. This is particularly useful for electromechanics and electroacoustics where the conversion of electrical, mechanical, and acoustical energy is paramount. The number of coordinates needed is equal to the number of degrees of freedom of the mechanical system (transducer). The Lagrange equations may then be transformed from one system to another (electrical to mechanical to acoustical or vice versa) and each part of the problem may be solved separately and then synthesis in the form a multicontour equivalent electrical network. The Lagrangian based approach [J. Acoust. Soc. Am. **118**(2), 2005)] is taught with examples of piezoceramic spheres, cylinders, bars, plates, and disks in a graduate course on Electroacoustics at the University of Massachusetts Dartmouth.

11:15

5aED13. The Verasonics ultrasound system as a pedagogic tool in teaching wave propagation, scattering, beamforming, and signal processing concepts in physics and engineering. Peter J. Kaczkowski (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pjk@uw.edu) and Ronald E. Daigle (Verasonics, WA 98053)

The Verasonics ultrasound system is a highly programmable data acquisition and processing platform designed to facilitate development of new medical ultrasound imaging methods. In contrast to conventional commercial ultrasound systems, individual element digitized rf data are available to the developer. All beamforming and postprocessing are done in software, and both the hardware data acquisition sequence and the host computer processing flow are programmable by the user using a MATLAB interface. Because the system is designed to be highly flexible, it can also be useful as a practical tool in teaching acoustic wave physics, transducer and array design, and data processing concepts, using benchtop scale homemade acoustic and elastic media, including flow models. For script evaluation and testing, the Verasonics system includes a hardware simulator that uses a simple point scatterer numerical model to compute rf backscatter data. rf data can also be recorded during a hardware acquisition, and then reprocessed using different user-developed algorithms for comparative study. Because the system is easy to learn, many fundamental concepts can be explored in a labo-

ratory setting, using scattering media or custom transducers fabricated as part of the student experimental plan. The system enables sophisticated hands-on experience with acoustics beyond the numerical world.

11:30

5aED14. Energy flux streamlines versus the alternatives for the visualization of energy flow in acoustical scattering problems: Simple examples. Cleon E. Dean (Dept. of Phys., Georgia Southern Univ., P.O. Box 8031, Statesboro, GA 30460-8031, cdean@georgiasouthern.edu) and James P. Braselton (Georgia Southern Univ., Statesboro, GA 30460-8093)

Energy flux streamlines yield certain advantages for the visualization of acoustical scattering processes. They give information about scattering angles, interaction with surfaces and scatterers, and can even show the relative intensity of the sound at a given location. However, the use of energy flux streamlines presents certain difficulties as well. Beyond the difficulty in calculating them, the use of the energy flux field presumes a complete solution of the sound scattering problem. Thus energy flux streamlines are usually descriptive rather than predictive. And while the set of streamlines can be chosen so as to show the relative intensities of the sound field, what works in one geometry will not necessarily work in another. Examples will be adduced illustrating these points and more.

11:45

5aED15. Experiments with heat- and vortex-driven acoustic instabilities in tube resonators. Konstantin Matveev (MME School, WSU, Pullman, WA 99164-2920, matveev@wsu.edu) and Rafael Hernandez (MME School, WSU, Pullman, WA 99164-2920)

Self-excited tonal sound often appears in engineering systems with mean flow and heat release. To assist students in learning about acoustic instabilities in resonators, a modular low-cost system was constructed and a series of experiments was developed. The experimental setup consists of pipe sections with baffles, damping chamber, air blower, nichrome-wire heater, and piezoelements. The system can operate as a Rijke tube, thermoacoustic engine, vortex-excited resonator, and harvester of acoustic power. Tests carried out with this system include (i) identification of instability domains in the controllable parametric space of the system geometry, mean flow rates, and supplied heat; (ii) investigation of transient phenomena, such as growth and attenuation of sound amplitude; (iii) demonstration of nonlinear effects, such as hysteresis in the system behavior and frequency locking; and (iv) harvesting of energy of self-excited sound using piezoelements. The observed phenomena are interpreted with help of simplified theoretical models. [Work supported by the NSF Grant No. 0853171.]

FRIDAY MORNING, 27 MAY 2011 — WILLOW B, 8:00 A.M. TO 12:20 P.M.

Session 5aNS

Noise, Committee on Standards, and Psychological and Physiological Acoustics: Occupational Noise Exposure: Assessment to Intervention

William J. Murphy, Cochair
NIOSH, 4676 Columbia Pkwy., Cincinatti, OH 45226-1998

Charles S. Hayden, Cochair
NIOSH, 4676 Columbia Pkwy., Cincinatti, OH 45226-1998

Chair's Introduction—8:00

Invited Papers

8:05

5aNS1. National occupational research agenda for noise: What we know and what we still need to know. Alice H. Suter (Alice Suter & Assoc., 1106 NE Tillamook St., Portland, OR 97212) and Christa Themann (The DeSales Group, Cincinnati, OH 45233)

The white paper generated by the original hearing loss prevention team of the National Occupational Research Agenda (NORA) has been revised and updated. In the process, important recent research results have come to light; some of them opening up new questions. Unfortunately some of the old questions remain unanswered. This paper will summarize some of the unanswered questions and concentrate on various new findings as well as research needs in areas such as impulse noise assessment, the interaction of noise and aging, the effects of noise on communication and safety, tinnitus, extra-auditory effects, and the evaluation of hearing conservation programs. There will always be areas about which knowledge is insufficient, but there are also salient reasons why action must be taken to prevent unwanted consequences.

8:25

5aNS2. Selling a quiet workplace through "buy quiet" programs. Charles S. Hayden, II and Heidi Hudson (CDC/NIOSH, 4676 Columbia Pkwy. C27, Cincinnati, OH 45226)

By implementing within the procurement process a program coined as "buy quiet," an employer can most effectively reduce hazardous levels of noise at their worksites. The process also shifts some of the responsibility for "quiet" onto the groups most capable of reducing noise emission at its source, the manufacturers of the machinery, and equipment being purchased. Controlling noise at its source is best accomplished by those manufacturers, as they are the technical experts on the operating parameters and therefore best suited to make the necessary design changes to reduce noise emissions without adversely affecting the quality or effectiveness of the operation. The process requires a purchaser to compare published noise emission levels of differing models of equipment being pur-

chased and, whenever possible, purchasing the quieter model. Within the process are tradeoff analysis worksheets to weigh the cost of reduced noise emission with other standard purchasing requirements. Buy quiet provides an easy and effective method for an employer to demonstrate a commitment to the use of best available technology to reduce the number of worker's suffering occupational noise induced hearing loss.

8:45

5aNS3. Improving the accuracy of noise exposure estimates for workers with highly variable exposures. Richard L Neitzel, William E Daniell, Lianne Sheppard (Univ. of Washington DEOHS, Seattle, WA 98195-7234), Hugh W. Davies (Univ. of British Columbia, Vancouver, BC V6T 1Z3, Canada), and Noah S Seixas (Univ. of Washington DEOHS, Seattle, WA 98195-7234)

Noise exposure assessment is difficult, and particularly so for workers with variable noise levels. We evaluated exposure estimates created using three different techniques: trade-mean, task-based, and subjective rating. We created trade-mean, task-based, and subjective rating estimates for a group of 68 construction workers using information collected on three workshifts over 4 months. This information included their trade, the tasks performed on each workshift, their subjective ratings of their noise exposures, as well as a full-shift exposure measurement on each of the three workshifts. We then created hybrid exposure assessment techniques using various combinations of the trade-mean, task-based, and subjective rating estimates, and compared these hybrid estimates to subjects' measured exposures and to estimates from the single techniques. Hybrid techniques generally resulted in substantial improvements in accuracy compared to the single techniques. A linear regression-based hybrid approach had the best performance, but two much simpler hybrid techniques did nearly as well. Adding trade-mean information did not improve the accuracy of hybrid estimates. These results suggest that a hybrid regression technique combining task-based and subjective rating estimates may produce the most accurate estimates of exposure for workers with highly variable noise exposures.

9:05

5aNS4. Evaluating the effectiveness of interventions to control noise and work-related hearing loss. Thais C. Morata (Div. of Appl. Res. and Technol., Natl. Inst. for Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, OH 45226, tmorata@cdc.gov)

The objective of this presentation is to discuss the literature and recommendations on the evaluation of the effectiveness of interventions to control noise and prevent hearing loss. The American Recovery and Reinvestment Act of 2009 included a provision for federal funding to investigate how different interventions stack up against each other. The Act called on the Institute of Medicine to recommend a list of priority topics to be the initial focus of a new national investment in comparative effectiveness research. The need for research on hearing loss was placed in the highest priority group. Two recent Cochrane Reviews addressed the effects of interventions for the prevention of work-related hearing loss. Those concluded that some interventions improve the mean use of hearing protection devices compared to non-intervention; that there is low quality evidence that legislation can reduce noise levels in workplaces and contradictory evidence that prevention programs are effective in the long-term. There is consensus that most interventions focus on the use of hearing protection devices, and effectiveness depends on the quality of the implementation. Even though case studies show that substantial noise control can be achieved in the workplace, there is no evidence of this practice in the scientific literature.

9:25

5aNS5. They are your ears: Personal protection and personal responsibility. William J. Murphy (CDC/NIOSH Hearing Loss Prevention Team, 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226, wjm4@cdc.gov)

Hearing protection devices are a primary means of protecting workers from occupational noise-induced hearing loss. However, the Achilles' heel of this approach is the correct and consistent use of personal protective equipment. When workers fail to wear or fit protection correctly, they risk compromising their hearing. Currently five companies have developed commercial fit-testing solutions for hearing protection devices that can be used in a range of acoustical environments. In some cases the fit-test method requires a quiet test space because hearing thresholds are measured for unoccluded and occluded conditions under headphones. In other cases, the measurements are conducted above threshold and can be performed in acoustical environments with more background noise. These solutions will be presented along with evidence of how training in protector use can affect self-efficacy and attitudes toward using protection. Pending changes in the US Environmental Protection Agency regulations for labeling a variety of hearing protection will be discussed. Example cases for estimating occupational noise exposure with the new labels will be presented.

9:45

5aNS6. Statistical assessment behind a standard on hearing protector field attenuation measurement devices. Jérémie Voix (École de Technologie Superieure, Montréal, PQ H3C 1K3, Canada, jeremie.voix@etsmtl.ca) and William J. Murphy (Natl. Inst. for Occupational Safety and Health, Cincinnati, OH 45226-1998)

New measurement systems for assessing individual hearing protection device (HPD) performance in the field have been developed over the past several years to address the question of what amount of protection is a given individual getting from an HPD. Although these systems, referred to as field attenuation measurement systems (FAMSs), have the same purpose and produce attenuation values that are presented in similar ways, the underlying technology used to produce a personal attenuation rating (PAR) can be drastically different, ranging from psychophysical tests to objective microphone measurements and involving single or multiple frequency measurements. In an effort to ensure that FAMS provide an attenuation rating that is a both a scientifically valid number and a meaningful reading for the end-user, the members of the American National Standard Institute Working Group 11 have recently been starting to work on a proposed standard to specify the minimum performance criteria for a FAMS to assure it provides data with defined accuracy and precision. The current paper describes the underlying statistical assessments that are considered in the standard. Specifically, the computational details of the number of subjects required for various assessments made within the standard for the determination of the maximum permissible background noise, measurement uncertainty, and HPD fit uncertainty will be presented. The paper will also detail the calculation of the repeatability and reproducibility.

10:05—10:20 Break

Contributed Papers

10:20

5aNS7. Acoustic test fixture for insertion loss measurements of hearing protector devices. Mikkel Bo Bergholt Nilsson (G.R.A.S. Sound Vib., Skovlytoften 33, DK-2840 Denmark, mn@gras.dk) and Jacob Sondergaard (G.R.A.S. NA, 2285 East Enterprise Parkway, Twinsburg, OH 44087)

According to ANSI standard S12.42 new aspects have to be considered when measuring the Insertion Loss of Hearing Protectors at an Acoustic Test Fixture. First, the ear simulator has to be able to measure up to 170 dB SPL, so a 1/4 in. microphone is needed. The ear canal extension is also longer than the standard extension known from KEMAR and is fitted with flesh simulation. The circumaural pinna has a larger diameter to ensure a perfect seal for all kinds of hearing protection devices (HPD). Furthermore, the total system consisting of ear simulator, ear canal extension, and pinna is heated to body temperature. All the effects of these new aspects have been measured to assess their influence compared to previous ATF types like the ISO 4869-3.

10:35

5aNS8. Measurement of impulse peak insertion loss for five hearing protectors. William J. Murphy (CDC/NIOSH Hearing Loss Prevention Team, 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226, wjm4@cdc.gov), Gregory A. Flamme (Mich. Univ., Kalamazoo MI 49008), Amir Khan, Joseph Echt (CDC/NIOSH Hearing Loss Prevention Team, 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226), and Belinda C. Johnson (CDC/NIOSH Biomonitoring Hazard Assessment Branch, 4676 Columbia Parkway, MS C-26, Cincinnati, OH 45226)

In 2009, the US Environmental Protection Agency proposed an impulse noise reduction rating (NRR) for hearing protection devices. The impulse NRR is based the American National Standard, ANSI S12.42-2010, and requires measurements with an acoustic test fixture for three ranges of impulse noises: 130–134, 148–152, and 166–170 dB peak SPL. Five protectors of each of five models (The Combat Arms Linear, Combat Arms Nonlinear, EAR Pod Express, Etymotic EB1, and Bilsom 707 Impact II, all in passive mode) were evaluated per the levels specified in the ANSI standard. Impulses were generated by an acoustic shock tube in the laboratory and by a 0.223 caliber rifle in the field. At each peak impulse level, protector samples were fitted on the test fixture five times and for each insertion, at least three impulses were measured. The impulse NRR increased with peak pressure and ranged between 20 and 38 dB. For some protectors, significant differences were observed across protector examples of the same model and across insertions. Relationships between the continuous noise NRR, the impulse NRR, and the increase in allowable impulse exposures due to the protector will also be presented.

10:50

5aNS9. Hearing conservation in high school students: A model using significant threshold shifts. Dave Martens, Aericka Dunn, Katherine Freeman, Shanna White, and Al Yonovitz (Dept. of Comm Sci. and Disord., Univ. of Montana, dave.martens@umconnect.umt.edu)

Hearing loss affects 19.5% of adolescents between 15 and 19 years [Shargorodsky *et al.* (2010)]. Although often subtle, this hearing deficit impairs perception of speech and warning signals and may increase the risk of depression, accidents, and social isolation. It is now commonly known that even slight hearing loss (15–24 dB) can create a need for speech and language therapy and auditory training with specialized equipment. In adults, the leading preventable cause of acquired sensorineural hearing loss is exposure to excessive levels of noise, however, concern has now reached a pinnacle that children and young adults are developing noise induced hearing loss as a result of overexposure to amplified music, especially through the use of personal music players. Three high schools ($N>3000$) are now part of a study that (1) provides threshold audiograms on each student, (2) determines a threshold shift between a baseline and annual audiogram, and (3) provides intensity measures of personal music devices and education to students. This program and its implementation will be discussed.

11:05

5aNS10. Acoustical characterization of exploding hydrogen-oxygen balloons. Julia A. Vernon, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602), and Jeffrey H. Macedone (Brigham Young Univ., Provo, UT 84602)

Exploding balloons are popular demonstrations in introductory chemistry and physical science classes and as part of outreach programs. Anechoic measurements of various hydrogen and hydrogen-oxygen balloons were made using 3.18 and 6.35 mm microphones placed at various angles and distances from the balloon. Time waveform data from each explosion were collected at a sampling frequency of 192 kHz. Initial research, presented previously, was conducted to determine potential auditory hazard as a result of exposure to these balloons. Further analysis has been conducted on the explosions' time waveforms and calculated spectra to characterize hydrogen-oxygen balloons as an impulsive noise source. Consideration is given to level, waveform A duration, rise time, angular variation, and repeatability. Comparisons with other impulsive noise sources are also presented.

11:20

5aNS11. The experimental measurement of motorcycle noise. Michael Carley, John Kennedy (Dept. of Mech. Eng., Univ. of Bath, Bath BA2 7AY United Kingdom), Ian Walker (Univ. of Bath, Bath BA2 7AY United Kingdom), and Nigel Holt (School of Sci. Society and Management, Bath Spa Univ., Newton St Loe, Bath, BA2 9BN, United Kingdom)

The noise source mechanisms involved in motorcycling include various aerodynamic sources and engine noise. The problem of noise source identification requires extensive data acquisition of a type and level that have not previously been applied. Data acquisition on-track and on-road are problematic due to rider safety constraints and the portability of appropriate instrumentation. One way to address this problem is the use data from wind tunnel tests. The validity of these measurements for noise source identification must first be demonstrated. In order to achieve this extensive wind tunnel, tests have been conducted and compared with the results from on-track measurements. Sound pressure levels as a function of speed were compared between on-track and wind tunnel tests and were found to be comparable. Spectral conditioning techniques were applied to separate engine and wind-tunnel noise from aerodynamic noise and showed that the aerodynamic components were equivalent in both cases. The spectral conditioning of on-track data showed that the contribution of engine noise to the overall noise is function of speed and is more significant than had previously been thought. These procedures form a basis for accurate experimental measurements of motorcycle noise.

11:35

5aNS12. The sources and effects of noise exposure in motorcyling. Michael Carley, John Kennedy (Mech. Eng., Univ. of Bath, Bath BA2 7AY, United Kingdom), Nigel Holt (Bath Spa Univ., Bath BA2 9BN, United Kingdom), and Ian Walker (Univ. of Bath, Bath BA2 7AY, United Kingdom)

We report on the Bath Motorcycle Collaboration, an interdisciplinary collaborative research effort involving the Departments of Mechanical Engineering and Psychology at the University of Bath and Bath Spa University in the United Kingdom. The group has taken a broad approach to the problem of noise in motorcycling, examining its sources, transmission, and effects. Noise-induced hearing loss is a problem which can affect professional riders and racers as well as leisure riders and commuters. To study the problem, extensive wind tunnel tests have been conducted to provide detailed aerodynamic measurements and flow visualization around the helmet. These results have then been compared with and validated using on-track data covering realistic riding conditions. Insertion loss measurements combined with loudness matching tasks on groups of volunteers have been used to investigate the process of noise transmission through the head/helmet system. Hearing threshold shift measurements have been conducted to quan-

tify the effects of this type of noise exposure on riders. This comprehensive approach has yielded valuable information for rider safety and has helped identify the research questions which will lead to a proper understanding of this important health and safety issue.

11:50

5aNS13. The effects of windscreen flow on noise in motorcycle helmets. John Kennedy, Michael Carley (Dept. of Mech. Eng., Univ. of Bath, Bath BA2 7AY, United Kingdom), Nigel Holt (Bath Spa Univ., Newton St Loe, Bath BA2 9BN, United Kingdom), and Ian Walker (Dept. of Psych., Univ. of Bath, Bath BA2 7AY, United Kingdom)

Vortex shedding from a motorcycle windscreen results in three flow regions into which the helmet of the rider may be immersed. First, the helmet may be fully in the free stream. Second, the helmet may be directly in the path of vortex shedding from the windscreen. Third, the helmet may be beneath the vortex shedding and shielded from the free stream by the windscreen. On-track tests were conducted and show a difference in sound pressure level of over 10 dB and a change in spectral content for different riding positions and helmet angle. Similar tests were then conducted in a wind tunnel, where simultaneous microphone and flow visualisation measurements allowed the identification and investigation of each flow region under controlled conditions. The contribution of vortex shedding to the noise was assessed using a combination of wavelet analysis and conditional averaging to identify intermittent structures.

12:05

5aNS14. Spectral filtering characteristics of a motorcycle helmet. Ian Walker (Dept. of Psych., Univ. of Bath, Bath BA2 7AY, United Kingdom), Nigel Holt (Bath Spa Univ., Newton St. Loe, Bath BA2 9BN, United Kingdom), John Kennedy, and Michael Carley (Univ. of Bath, Bath BA2 7AY, United Kingdom)

Noise transmission characteristics of a motorcycle helmet have been analyzed using a combination of insertion loss measurements and loudness matching in a behavioral study. Results demonstrate the action of the motorcycle helmet as a spectral filter. The insertion loss measurements confirm previously published data showing attenuation in the frequency range above 500 Hz. A further feature, the significance of which is addressed and highlighted here for the first time, is an amplification of noise below 500 Hz. In short, the helmet acts as a frequency dependent filter on the input to the human auditory system. Data from the matching task were used to generate equiloudness curves which show the effect of the helmet on riders' perceptions of loudness. The generated curves were compared to the international standards (ISO226). The character of the equal loudness curves was strongly influenced by the helmet. This difference is discussed in the framework of the filtering characteristics of the helmet.

FRIDAY MORNING, 27 MAY 2011 — WILLOW A, 8:00 TO 10:25 A.M.

Session 5aPA

Physical Acoustics, Acoustical Oceanography, and Underwater Acoustics: Acoustics of Gas Hydrates

Preston S. Wilson, Chair

Univ. of Texas at Austin, Dept. of Mechanical Engineering, 1 University Station, Austin, TX 78712-02923

Chair's Introduction—8:00

Invited Papers

8:05

5aPA1. Acoustic properties of natural gas hydrates and the geophysical assessment of the subsurface distribution of hydrates in the Gulf of Mexico and Atlantic. William Shedd (BOEMRE, 1201 Elmwood Park Blvd., New Orleans, LA 70123, william.shedd@boemre.gov), Matt Frye (BOEMRE, 381 Eldon St., Herndon, VA 20170)), Paul Godfriaux, and Kody Kramer (1201 Elmwood Park Blvd., New Orleans, LA 70123)

Natural gas hydrates are a solid form of natural gas found in the deep water marine margins of continents and under permafrost in Arctic regions worldwide. They have been recognized as a very significant potential energy source in the future. They form under high pressure and low temperature. Hydrate saturated sediments are acoustically faster and slightly less dense than water saturated sediments, but much faster and denser than gas saturated sediments. These properties allow for the identification of marine hydrate saturated sediments that are underlain by gas saturated sediments. The resulting geophysical reflector, referred to as a bottom simulating reflector, or BSR, often mimics the seafloor in areas where geothermal gradient is laterally consistent. The Bureau of Ocean Energy Management, Regulation, and Enforcement has used three-dimensional seismic data in the Gulf of Mexico and two-dimensional seismic data in the Atlantic to (1) map the distribution of BSRs, (2) drill six wells in the GOM with moderate to high hydrate saturations in sand reservoirs, and (3) assess the resource potential of hydrates.

8:35

5aPA2. Acoustic studies of submarine gas hydrates on the Cascadia Margin. Mikhail M. Zykov and N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada)

Submarine gas hydrates were detected over large areas in the Cascadia margin off the west coast of British Columbia by the presence of bottom simulating reflectors in seismic surveys. Initial estimates of gas quantity based on these data and results from an ocean drilling program leg were highly optimistic, and subsequently many different experiments were carried out to quantify the estimates. This paper presents results from an air gun survey with ocean bottom seismometers deployed around a hydrated cold vent to determine the hydrate concentration in the vicinity of the vent. Reflection travel time data from the survey were used in a linearized inversion to determine the velocity structure within the hydrate stability zone. Reflection amplitude data were analyzed to generate maps of the

reflectivity at the sea floor and at the base of the stability zone. The travel time inversion revealed that the sediment velocities were very low, indicating low concentration of hydrate (less than 2% of sediment volume) in the stability zone. The sea floor reflection amplitude data provided a map showing an elliptical extent of the sea floor vent. Inversion of amplitude data from the BSR indicated a small decrease in velocity, less than 7%.

8:55

5aPA3. Modeling for remote acoustic characterization of gas hydrates. Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu)

Hydrates have a strong effect on elastic properties of the seafloor and their spatial (vertical and lateral) variability. The contrast in velocity created by the hydrate-cemented zone produces a strong seismic and acoustic reflection, the "bottom simulating reflection," which is commonly used to locate gas hydrate deposits. Gas-hydrate-cemented strata also act as seals for trapped free gas. Hydrates may form complicated three-dimensional (3-D) structures and inclusions, such as nodes, veins, and flakes, resulting in significant volume heterogeneity of the sediment. This may cause a strong impact on the seabed scattering properties. Currently existing models of scattering from heterogeneous sea beds can be used for predicting this impact, and also serve as a base for development of new algorithms for remote acoustic characterization of gas hydrates and monitoring their stability. This possibility is discussed, and examples of such modeling are presented.

9:15

5aPA4. Active multibeam sonar-derived bubble plume fluxes and dynamics. Ira Leifer (Marine Sci. Inst., Univ. of California, Santa Barbara, CA 93106-5080, ira.leifer@bubbleology.com), Bruce P. Luyendyk, and Dan Culling (Univ. of California, Santa Barbara, CA 93106-9630)

By their nature, seeps are spatially and temporally variable and episodic; thus, effective emissions' quantification presents significant challenges because local measurements likely are unrepresentative. A multibeam sonar system was developed to observe ebullition spatially and used to study water column bubble plume processes, as well as subsurface migration processes for seepage from the Coal Oil Point seep field and the East Siberian Arctic Sea is presented. The system is designed for deep-sea application related to hydrates as part of a benthic observatory.

9:35

5aPA5. Laboratory measurements on gas hydrates and bubbly liquids using active and passive low-frequency acoustic techniques. Chad A. Greene, Preston S. Wilson (Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292, chad@chadagreene.com), and Richard B. Coffin (Naval Res. Lab., Washington, DC)

The unique molecular structures of gas hydrates result in curious acoustic properties, which have yet to be adequately described. Understanding the acoustic behavior of stable and dissociating gas hydrates in liquids is vital for their localization and quantification using seismic or echosounding techniques. Further, with improved characterization of the acoustic properties of gas hydrates and bubbly liquids containing methane gas, acoustic methods may become an invaluable tool for monitoring hydrate dissociation and determining the magnitude of its effects on climate change. Acoustic properties of gassy substances are known to have a strong dependence on excitation frequency; however, tabulated values of hydrate material properties are most often measured in the frequency range of hundreds of kilohertz, while natural hydrate deposits and gas seeps are typically surveyed at seismic frequencies several orders of magnitude below laboratory measurement frequencies. This presentation details laboratory experiments in which a low-frequency (10 Hz–10 kHz) acoustic resonator apparatus was used to measure (a) sound speeds of bubbly liquids containing ideal and real gases and (b) bulk moduli and dissociation pressures of natural structure I and structure II gas hydrate samples. [Work supported by the Office of Naval Research.]

Contributed Papers

9:55

5aPA6. Elastic constants of palladium hydride at elevated temperature and pressure. Rasheed Adebisi and Joseph Gladden (Dept. of Phys. and Astronomy, Univ. of Mississippi, Lewis Hall, University, MS 38677-1848)

Hydrogen atoms occupy the octahedral interstitial sites provided by metals with fcc lattices (such as palladium). Palladium hydride (PdH_x) exists in two phases. At room temperature, the low concentration ($PdH_{x=0.02}$) called α-phase has slightly larger lattice parameter compared to that of pure palladium. Above this concentration and up to ($PdH_{x=0.6}$) the high concentration β-phase appears resulting in ($\alpha+\beta$)-phase, a mixed phase region, and the lattice parameter becomes substantially greater than that of pure palladium. When $x>0.6$, α-phase disappears and the system becomes purely β-phase. The three phases coexist at the tri-critical point (temperature, pressure, and concentration). Resonant ultrasound spectroscopy (RUS) has been used to investigate the elastic constants of palladium hydride crystal and the equilibrium dynamics of the system around the tri-critical point temperature and pressure. A large decrease in shear modulus is noticed near this tri-critical point. The desire to have fundamental understanding of metal-hydrogen systems and to explore this for possible technological applications has been the motivation for research of these systems. This investigation includes the design and construction of high temperature and high pressure RUS cell.

10:10

5aPA7. Acoustic resonators, quantum mechanics, and gas metrology. Michael R. Moldover (Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8360, michael.moldover@nist.gov) and James B. Mehl (P.O. Box 307, Orcas, WA 98280)

Gas-filled, quasi-spherical cavity resonators were invented to accurately measure the thermodynamic temperature. They have been used with success from 7 to 550 K. Within a few years, the Boltzmann constant k_B will become a defined constant. Prior to the definition, groups in several countries plan to re-determine k_B using quasi-spherical resonators. In doing so, they expect to measure the speed of sound in argon and helium with sub-part-per-million uncertainties. This work has stimulated several branches of physics and gas metrology. For example, quantum-mechanical calculations of the properties of helium (second and third virial coefficients, viscosity, and thermal conductivity) are more accurate than these properties can be measured. Thus the calculated properties have become standards. Other examples include calculating of microwave resonance frequencies (including coupling effects) of quasi-spherical cavities at the sub-part-per-million level, improved understanding of ducts to conduct gas and sound into and out of cavities, and improved understanding of the interactions of gas modes with the shell surrounding the cavity.

Session 5aPP

Psychological and Physiological Acoustics: Assessment and Consequences of Hearing Loss

Frederick J. Gallun, Chair
Portland VA Medical Center, National Center for Rehabilitation, 3710 S.W. U.S. Veterans Hospital Rd., Portland, OR 97239

Contributed Papers

8:00

5aPP1. Motorcycle helmets and the frequency dependence of temporary hearing threshold shift. Nigel Holt (School of Sci. Society and Management, Bath Spa Univ., Newton St. Loe, Bath BA2 9BN, United Kingdom.), Ian Walker, John Kennedy, and Michael Carley (Univ. of Bath, Bath BA2 7AY, United Kingdom.)

Temporary hearing threshold shifts (THTSs) as a result of exposure to noise vary as a function of the noise's spectral content. However, to date THTS has been measured and predicted in a way that does not take account of frequency variation—most notably in standards such as British Standard 5330. We therefore carried out pure-tone audiometry on participants before and after exposure to white noise in order to quantify the frequency dependence of the THTS. Moreover, as this research group has previously shown that motorcycle helmets act as spectral filters, attenuating noise in the region above 500 Hz and amplifying noise in the regions below 500 Hz; this was done both with and without a motorcycle helmet. As our previous findings would suggest, the pattern of threshold shift is a function of the filter characteristics of the helmet, including an increased sensitivity at higher frequencies. There was also greater than expected reduction in sensitivity at frequencies where the helmet amplifies incident noise. The results indicate an acoustic effect of helmets which has not previously been reported.

8:15

5aPP2. Development of an advanced hearing protection evaluation system. Kevin Shank, Josiah Oliver, Fred Lalande, Mehmet Bicak, and Kenji Homma (Adaptive Technologies, 2020 Kraft Dr., Ste. 3040, Blacksburg, VA 24060)

Acoustic test fixtures (ATFs) are practical and often necessary tools for testing hearing protection devices (HPDs) especially with extremely loud impulsive and/or continuous noise, for which the use of live subjects might not be advisable. Although there have been various standardized and laboratory ATFs from past research, there still exists large uncertainty in the correlation between the attenuation results obtained from ATFs and those obtained from actual human subject tests, particularly for intraaural HPDs. It is suspected that one of the main factors contributing to the discrepancy may be insufficient fidelity in the circumaural/intraaural flesh system, whose underlying dynamics is not yet clear. Therefore the first goal of this research is to better understand this biomechanical system and implement it into a new ATF prototype which emulates circumaural/intraaural HPD attenuation performance. This prototype ATF will be validated against human subjects. This presentation discusses the research methodologies and design strategies for developing this advanced hearing protection evaluation system.

8:30

5aPP3. Heartbeat-synchronous audiometry. William M. Hartmann and Yun Jin Cho (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824)

The goal of heartbeat-synchronous audiometry (HSA) is to measure the effect of a listener's heartbeat on pure-tone detection thresholds. Our experiment used 100-ms tones (20-ms rise/fall) with onsets occurring at selected phases of the heartbeat cycle. Cardiac waves were acquired with three chest electrodes at a sample rate of 2400 sps. Successive peaks of the QRS complex (ventricular depolarization), sharpened with a software Schmitt trigger, served to identify heartbeat cycles. The arrival time for the next heartbeat was predicted using a weighted moving-average online computation with special compensation for early beats. Software optimization led to measured rms prediction errors averaging 0.04 (sd=0.01) cycles. The HSA technique was applied to one-interval (Békésy tracking) and two-interval (up-down staircase) audiometric methods. Reproducible 4-kHz threshold maxima (cardiac-phase=0.3 cycles) and minima (cardiac-phase=0.7 cycles) differing by more than 6 dB were seen for a listener with pulsatile tinnitus. For normal listeners, interest centers on the frequencies of emissions or deep valleys in the audiogram. [Work supported by the NIDCD Grant No. 00181.]

8:45

5aPP4. Utility of categorical loudness scaling to estimate the outer- and inner-hair-cell-loss-related components of hearing impairment. Jxfc;rgens Tim, Ewert Stephan, Brand Thomas, and Kollmeier Birger (Medizinische Physik, Carl-von-Ossietzky Universitt Oldenburg)

Sensorineural hearing loss is mainly accompanied by a reduction or dysfunction of outer hair cells (OHCs) and inner hair cells (IHCs) in the cochlea. A dysfunction of OHCs is often assumed to correspond to an expansive component and a dysfunction of IHCs to an attenuating component of hearing loss. This study presents a method for assessing OHC and IHC losses from categorical loudness curves using adaptive categorical loudness scaling (ACALOS). A loudness model was used to fit modeled loudness curves to measured loudness curves by adjusting a model parameter HL_{OHC} that quantifies the amount of hearing loss related to the dysfunction of OHCs relative to total hearing loss. This parameter HL_{OHC} was compared to corresponding parameters extracted using temporal masking curves (TMCs) in 16 listeners with different audiometric thresholds. The results showed that both ACALOS and TMCs estimated HL_{OHC} consistently, whereas ACALOS offers much more time-efficiency and about the same accuracy as TMCs. Since ACALOS is an easy and natural task for inexperienced listeners and because of the high time-efficiency, ACALOS appears to be a good candidate for the assessment of this part of the suprathreshold processing of sensorineural hearing-impaired listeners in clinical practice. [Work supported by the DFG (Grant No. SFB TRR31).]

9:00

5aPP5. Filter characteristics derived from auditory-nerve fiber responses following noise-induced hearing loss. Sushrut S. Kale (Weldon School of Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., West Lafayette, IN 47907) and Michael G. Heinz (Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907)

Sensorineural hearing loss (SNHL) produces a loss of frequency selectivity and elevated thresholds in the auditory periphery. Broadened cochlear tuning significantly affects magnitude and phase responses of auditory filters and thus can change cochlear-response timing. The present study compared auditory-filter characteristics in normal-hearing and noise-exposed anesthetized chinchillas. Impulse responses of auditory filters were derived from responses of auditory-nerve fibers to broadband noise using spike-triggered averaging techniques. Frequency modulations (or frequency glides) in impulse responses were quantified by computing instantaneous frequencies from impulse-response zero crossings. Group delays for individual auditory filters were computed from phase-frequency responses. Cochlear traveling-wave delays were estimated from shuffled cross-correlograms. Preliminary

results suggest that auditory-filter best frequency shifts apically following SNHL. Latencies of impulse responses and traveling-wave delays were much shorter following noise exposure. For normal-hearing fibers, frequency glides in impulse responses increased as a function of time for high characteristic-frequency (CF) fibers and decreased for low-CF fibers, consistent with previous studies. SNHL significantly affected frequency glides of noise-exposed fibers in a manner consistent with the loss of tonotopicity. These characterizations of impaired auditory filters have important implications for SNHL effects on temporal and spatiotemporal coding of complex sounds, such as speech. [Work supported by NIH Grant No. R01DC009838.]

9:15

5aPP6. Effects of repeated measures on the late auditory evoked response. Edward L. Goshorn, Charles G. Marx, and Alaina Simmons (Dept. of Speech and Hearing Sci., Univ. of Southern Mississippi, Hattiesburg, MS 39406, edward.goshorn@usm.edu)

A concern in clinical applications of the late auditory evoked response is the variability of amplitude and latency. While there is agreement that less variability is seen within than across subjects, the effects of stimulus and acquisition factors are not fully understood. This project examined the effects of repeated measures and stimulus type [1 kHz tone at 40 and 80 ms and speech (da) at 40 ms] on the amplitude and latency of P1. Latency was measured conventionally while amplitude was measured in two ways: peak-to-baseline (P-B) and peak-to-trough (P-T). Two amplitude measures were obtained because some commercially available instruments provide a P-B measure as the default amplitude. Stimuli were presented to 30 normal hearing young adults at 60 dBnHL. P1 latency increased significantly ($p=0.024$) from replicate 1 (35.9 ms) to replicate 3 (41.1 ms) for the 40 ms tone but not for 80 ms nor for speech. P-B amplitudes were significantly ($p<0.01$) smaller than P-T. Also, P-T amplitudes diminished significantly ($p<0.01$) from replicate 1 to replicate 3 for each stimulus type but P-B amplitudes did not. These findings suggest that P-B and P-T are not identical measures of P1 physiological activity.

9:30

5aPP7. Pure-tone glide detection and simulated formant transition discrimination by listeners with hearing loss. Peggy B. Nelson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu), Magdalena Wojtczak, Yingjiu Nie, Elizabeth Anderson, and Evelyn Davies-Venn (Univ. of Minnesota, Minneapolis, MN 55455)

Listeners with hearing loss (HI) have poorer pure-tone glide detection thresholds than normal-hearing (NH) listeners [Nelson *et al.*, ASA Baltimore (2009)]. Many HI listeners also experience reduced masking release for glide detection in gated noise compared to their performance in steady noise. Pure tones, however, are not good representations of formant transitions in speech. In this study, we compare listeners' thresholds for detecting pure-tone glides and for detecting changes in the spectral profile of a harmonic complex that mimics formant transitions. Formantlike transitions were simulated by adding intensity increments to harmonic components in the range 1–1.5 kHz. Increments were progressively delayed with increasing component frequency. Pure-tone glides and formantlike transitions were presented in quiet, steady noise, and gated noise to NH and HI listeners. Some but not all HI listeners showed poorer spectral profile change discrimination when compared to NH listeners, despite having normal increment-detection thresholds. A comparison of these results with pure-tone glide detection and masking release for speech will be presented. [Work supported by NIDCD Grant No. R01-DC008306.]

9:45

5aPP8. The effect of hearing loss on the resolution of partials and fundamental frequency discrimination. Brian C. Moore and Brian R. Glasberg (Dept. of Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, United Kingdom)

The relationship between the ability to hear out partials in complex tones, discrimination of the fundamental frequency (F0) of complex tones, and frequency selectivity was examined for subjects with mild-to-moderate cochlear hearing loss. The ability to hear out partials was measured using a two-interval task. Each interval included a sinusoid followed by a complex tone; one complex contained a partial with the same frequency as the sinusoid, while in the other complex that partial was missing. Subjects had to indicate the interval in which the partial was present in the complex. The components in the complex were uniformly spaced on the ERBN-number scale. Performance was generally good for the two "edge" partials, but poorer for the inner partials. Performance for the latter improved with increasing spacing. F0 discrimination was measured for a bandpass-filtered complex tone containing low harmonics. The equivalent rectangular bandwidth (ERB) of the auditory filter was estimated using the notched-noise method for center frequencies of 0.5, 1, and 2 kHz. Significant correlations were found between the ability to hear out inner partials, F0 discrimination, and the ERB. The results support the idea that F0 discrimination of tones with low harmonics depends on the ability to resolve the harmonics.

10:00—10:15 Break

10:15

5aPP9. Relationship between amplitude modulation in psychophysical tasks and speech in listeners with normal and impaired hearing. Eric Hoover, Pamela Souza (Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208), and Frederick Gallun (Natl. Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239)

Previous work suggests that listeners with hearing loss may be unable to discriminate high-rate amplitude modulations, even when the modulation depth is above the modulation detection threshold [Grant *et al.*, J. Acoust. Soc. Am. (1998)] **104**, 1051–60. For listeners with normal hearing, high-rate amplitude modulations contribute to speech understanding when spectral information is degraded [Xu *et al.*, **117**, 3255–3267 (2005)]. In listeners with hearing loss, the ability to use amplitude modulation in speech may be limited to frequencies at which modulation frequency discrimination is possible. To test this, amplitude modulation detection and discrimination thresholds were measured in listeners with normal hearing and mild-to-moderate sensorineural loss. The signal was a broadband noise carrier sinusoidally amplitude modulated at rates of 10–640 Hz. Sentence and nonsense syllable recognition was measured for the same listeners. To restrict spectral information, speech was vocoded with a sinusoidal carrier modulated with three or six channels of envelope information. Envelope filter cutoff frequency was varied from 10–640 Hz. Compared to listeners with normal hearing, the listeners with hearing loss had more variable modulation frequency discrimination and that discrimination was related to their ability to use high-rate modulation information in speech recognition. [Work supported by NIH.]

10:30

5aPP10. Effects of frequency compression on the intelligibility and quality of speech in noise. Pamela Souza (Northwestern Univ., Commun. Sci. Disorders, Evanston, IL 60208), Kathryn H. Arehart (Univ. of Colorado), James M. Kates (GN ReSound Corp.), Ramesh Kumar Muralimanohar, Naomi Croghan (Univ. of Colorado), and Eric Hoover (Northwestern Univ.)

Frequency-lowering technologies such as frequency compression have become common in high-end digital hearing aids. However, there are few data on the consequences of adjusting frequency compression ratio (FCR) and frequency compression cutoff (FCC). In this study, sentences at signal-to-noise ratios (SNRs) ranging from -10 to $+10$ dB were frequency compressed at a variety of FCRs (1.5–3) and FCCs (1–2 kHz). Frequency compression was implemented using sinusoidal modeling: the incoming signal was segmented, the spectral peaks found for each segment, and an output sinusoid at a shifted frequency was generated for each peak. Subjects were adult listeners with normal hearing or mild-moderate sensorineural loss. Intelligibility results illustrate a complex interaction between SNR and frequency compression parameters. At favorable SNRs, frequency compression had minimal effects. At less favorable SNRs, reducing FCC degraded intelligibility more than reducing FCR. At the poorest SNRs any amount of frequency compression degraded intelligibility but the specific FCC/FCR had little effect. Quality results show less interaction whereby any frequency compression was perceived as reducing speech quality. Compared to data from normal-hearing listeners, results from listeners with hearing loss were more variable and were interpretable in the context of audiogram configuration. [Work supported by NIH and GN ReSound.]

10:45

5aPP11. Analysis of hearing aid acoustical feedback by means of fast multipole method-boundary element method and the transfer matrix method. Julio A. Cordioli, Douglas C. Araujo, Arcanjo Lenzi, and Zargos Masson (Dept. Mech. Eng., Federal Univ. of Santa Catarina, 88040-970, Brazil, cordioli@emc.ufsc.br)

A numerical procedure is investigated for the analysis of hearing aid feedback. The receiver, connecting tubes, ear channel, and vent are considered by means of the standard transfer matrix method. The receiver transfer matrix is obtained experimentally through the two load method, while the transfer matrices representing the other components are obtained by assuming plane waves and taking in account viscous-thermal effects. The radiation impedance of the vent and the transfer function between the vent outlet and the microphone are calculated using the boundary element method and the fast multipole method (FMM-BEM). The use of FMM-BEM solver allows the solution of considerably larger models than standard BEM solvers, which means that finer details of the pinna, vent outlet, and hearing aid microphone may be included in the analysis. Numerical results are compared with experimental data obtained using a hearing aid mounted on the pinna of a manikin, and good agreement is observed.

11:00

5aPP12. Cochlear implant channels with high thresholds degrade medial vowel perception. Julie Arenberg Bierer (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98103, jbierer@u.washington.edu), Steven M. Bierer, Erin S. Maloff, and Ann Lin (Univ. of Washington, Seattle, WA 98105)

There is considerable evidence that the functional interface between cochlear implant electrodes and the auditory nerve varies from channel to channel within individual patients. Recently a method was proposed to gauge the status of a channel's interface on the basis of its threshold to focused stimulation using tripolar electrode configuration. This approach showed that channels with high tripolar thresholds have broader psychophysical tuning curves. In this study, the relationship between the number and distribution of high threshold channels with medial vowel identification in a bvt paradigm was examined. Tripolar thresholds were measured for all channels in seven listeners. Vowel identification was tested in the sound field with six repetitions. On the basis of each listener's electrodogram (the sound processor output of current amplitude as a function of time and channel), the similarity of the vowels using a principle component analysis was examined. We then related the vowel confusions based on the electrodogram of different subsets of channels, based on basal to apical location or tripolar threshold. Preliminary results suggest that the channels with high tripolar thresholds, or in regions of high variability in tripolar threshold, can account for a significant amount of the variance in subject performance. [Work supported by NIH Grant No. DC8883.]

11:15

5aPP13. Using spectral constancy to encode temporal pitch and improve cochlear-implant melody perception. Duo Zhang, Fan-Gang Zeng (Dept. of Otolaryngol.-Head and Neck Surgery and Biomedical Eng., Univ. of California, 364 Med Surge II, Irvine, CA 92697-1275, fzeng@uci.edu), and Bishnu S. Atal (Univ. of Washington, Seattle, WA 98195-2500)

Despite good speech performance in quiet, melody perception remains a challenge to cochlear-implant (CI) users. Current CI pitch-encoding schemes significantly alter original signals, producing potentially detrimental effects on speech perception. The present study proposes to take advantage of spectral constancy to encode temporal pitch while minimizing signal distortion. Spectral constancy refers to unaltered timbre perception when the overall level of a signal is roved. For example, the vowel /a/ sounds the same independent of its overall level. The idea is to extract and encode pitch by increasing the gain of all channels proportionally and synchronously according to the pitch period. This scheme alters the temporal envelope of voiced sounds but preserves their spectral envelope in each time frame. It does not alter temporal envelope of unvoiced sounds, making it a good comprise between encoding pitch and preserving spectral and temporal envelopes. The new scheme has been implemented in a real-time speech processor and evaluated in nine nucleus CI users who identified ten melodies with minimal training and in a test protocol that randomized the stimulus presentation order. Compared with the standard continuous interleaved sampling strategy, the new scheme significantly improved CI melody perception by 10–20%.

11:30

5aPP14. Relationship between channel interaction and spectral-ripple discrimination in cochlear implant users. Gary L. Jones, Ward R. Drennan, and Jay T. Rubinstein (Otolaryngol.-HNS, Univ. of Washington, 1959 Pacific St., Seattle, WA 98105)

Many cochlear implant (CI) users achieve remarkable success in understanding speech, but a key limitation affecting CI development is poor understanding of factors that contribute to individual performance. The spectral-ripple discrimination test offers a time-efficient, nonlinguistic measure that is correlated with perception of speech and music by CI users. What makes this test time-efficient, and thus clinically relevant, is that it is a "one-point" measure: Only the ripple density parameter is varied. However, there is controversy within the CI field about what this one-point test actually measures. The current work examines and models the relationship between thresholds in the spectral-ripple test, in which stimuli are presented acoustically, and interaction indices measured under the controlled conditions afforded by a research processor. Preliminary results show that (1) within individual subjects, there can be large variations in the interaction indexp; (2) interaction indices generally decrease with increasing electrode separation; (3) ripple discrimination thresholds increase with decreasing mean interaction index; and (4) trends in ripple thresholds predicted by a phenomenological model are consistent with psychophysical results. Results are consistent with the use of cross-channel cues by CI users in spectral-ripple discrimination. [Research supported by NIH-NIDCD Grant Nos. F32-DC011431, R01-DC007525, P30-DC04661, and L30-DC008490 and the Advanced Bionics Corporation.]

11:45

5aPP15. Speech and music recognition with acoustic simulations of a harmonic single sideband encoding strategy for cochlear implants. Xing Li, Kaibao Nie, Nikita Imennov, Jong Ho Won, Les Atlas, and Jay Rubinstein (Univ. of Washington, Seattle, WA 98195, xingli@u.washington .edu)

Cochlear implant users experience difficulties in speech perception in noise and music perception. Previous studies suggested that these difficulties are partially due to the lack of temporal fine structure (TFS) coding in cochlear implants [Moore, JARO (2008)]. TFSs are generally high-frequency signals; therefore, delivering perceivable TFS cues in electrical hearing is a challenge because the temporal sensitivity in cochlear implants is typically restricted to 300 Hz and below. A recently-proposed strategy, harmonic single sideband encoder (HSSE), could potentially use the harmonic structure of sounds to coherently demodulate TFS information [Li *et al.*, ICASSP (2010)]. To evaluate these potential benefits to speech and music perception, this study compared the performance of five normal hearing listeners using four- and eight-channel sinusoidal vocoder simulations of HSSE and the continuous interleaved stimulation (CIS) strategy. Scores on the following tasks were measured: sentence recognition in noise and single-talker maskers, melody and timbre recognition, and Mandarin tone discrimination. In all of the tasks, HSSE listeners obtained a significant performance improvement over the CIS vocoder. Furthermore, simulation results from an electrically-stimulated model of the auditory nerve fibers demonstrated that HSSE is capable of encoding temporal pitch cues better than CIS. [This study was supported by NIH R01-DC007525, P30-DC004661, and T32-DC005361; AFOSR Grant FA9550061019; CGF 657635; ITHS 620491; Advanced Bionics Graduate Fellowship; and NSF TG-IBN090004.]

Session 5aSC

Speech Communication: Adaptation, Priming, and Learning (Poster Session)

Grant McGuire, Chair
Univ. of California, Santa Cruz, Stevenson Faculty Services, Santa Cruz, CA 95064

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 a.m.

5aSC1. Speech intonation and perception: A study of frequency scales for Brazilian Portuguese. Marcus Vinícius Moreira Martins (Dept. de Letras Clássicas e Vernáculas, Univ. de São Paulo, Luciano Gualberto, 403—São Paulo, SP 05508-900, Brazil marcusvmmartins@gmail.com) and Waldemar Ferreira Netto (Univ. de São Paulo, Luciano Gualberto, 403—São Paulo, SP 05508-900, Brazil)

This work aims to verify the appropriation of the semitones scale for data analysis in studies involving intonational aspects of Brazilian Portuguese (henceforth BP), regarding mainly speakers perception and intuition at the time of production. The semitone scale was selected because it operates with frequencies ranges and with a central frequency to which the neighboring frequencies, in a given range, would be attracted covering the perception issue. The test was performed with 13 BP native speakers (seven women and six men) and consisted primarily in an exposure for a target-utterance. In next step, they were required to reproduce what they had heard: imitating just the intonation despite of the voice qualities and segments realization. The data (f0 values) obtained were plotted in mels, barks, semitones, and ERB-rates and all compared by (i) correlation (r^2), (ii) variation from target-utterance values of F0, and (iii) graphics comparison. The results so far obtained and analyzed indicates that the semitones scales can be used with no loses for frequencies under 600 Hz if compared with Hz scale and better fits for heights comparison. In relation to other scales, any difference was noted in absolute values at this moment. [Work funded by CNPq, Process no. 152293/2010-8.]

5aSC2. Perceived imitation of regional dialects. Sara C. Phillips and Cynthia G. Clopper (′Dept. of Linguist., Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, phillips@ling.osu.edu)

This study examined the effect of regional dialect on perceived spontaneous phonetic imitation. Participants shadowed CVC target words produced by talkers from two American English dialect regions, the North and the Midland. Imitation was measured acoustically in terms of vowel quality, vowel duration, midpoint f0, f0 trajectory, and onset and coda consonant duration. The productions and targets from the shadowing task were then used as stimulus materials in an AXB discrimination task to measure perceived imitation. Listeners selected whether the shadowed production of each word was more similar to the target (X) than a baseline recording of that same word. The AXB experiment included three conditions to distinguish between imitation of talker-specific and dialect-specific properties, in which X was the original target from the shadowing experiment, X was produced by another talker from the same region, or X was produced by a talker from the other region, respectively. Preliminary results suggest that listeners used talker-specific, rather than dialect-specific, properties when judging imitation. Among the acoustic measures, perceived imitation was correlated with vowel duration. These results suggest that talkers in a shadowing task imitate talker-specific variability more than general properties of regional dialect.

5aSC3. Gender effects on unconscious phonetic imitation. Alexis K. Black (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, akblack2g@gmail.com)

Studies on unconscious phonetic imitation have examined whether participant and/or model talker gender plays a role in degree and direction of the behavior. Findings, however, have alternately demonstrated greater imitation by men, greater imitation by women, more imitation toward a male model talker, and more imitation toward opposite sex model talkers. Other studies have reported no sex-based differences in imitation. Three experiments using a blocked-shadowing paradigm assessed the role of sex on degree and direction of imitation. In the first two experiments, base rates of imitation to a single model talker of a particular sex were compared across three acoustic features: vowel quality, word durations, and voice onset time (VOT). In the third experiment participants were exposed to both male and female models. One model talker exhibited modified VOT, enabling examination of sex-based differences in imitation. Preliminary results suggest that an imitator is more likely to mimic a model talker of the opposite gender and that acoustic features are imitated to different degrees. It is suggested that different acoustic features are associated with different social categories. These differences may explain the diverse and conflicting findings on sex-based effects that have been reported in the imitation literature.

5aSC4. The effect of divided attention on phonetic imitation. Jennifer Abel, Molly Babel, and Alexis Black (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada)

Research suggests that phonetic imitation is an automatic and subconscious process, but it is clearly a behavior that is variable across participants and conditions. The purpose of this experiment is to explore how a participant's level of attention to the speech signal moderates their degree of imitation. To this end, six conditions were prepared using a blocked exposure design: no redirection of attention (participants were instructed to simply listen to the model talker producing words), attention redirected through a math task, attention redirected through a picture-drawing task, attention focused through a word-memorization task, attention focused through a talker-description task, and attention focused through an explicit imitation task. A seventh condition using an immediate shadowing paradigm to compare to the blocked exposure design was run as well. Seventy native speakers of English (ten per condition) were recruited as participants. In all conditions participants' baseline productions of the stimuli wordlist were recorded prior to exposure to the model talker (a female speaker of North American English). Recordings are currently being analyzed for imitation using various acoustic parameters including whole word duration, vowel spectra, and *f*0.

5aSC5. Implicit and explicit phonetic imitations in single-word shadowing. Molly Babel, Meaghan Delaney, and Soraya Savji (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada)

Talkers spontaneously accommodate to a model talker in single word shadowing tasks. This behavior has been termed implicit phonetic imitation, as talkers have no awareness of having modified their speech. In this study,

we compare implicit phonetic imitation to explicit phonetic imitation across two groups of participants. All participants completed an auditory naming task, which included 50 monosyllabic words with the vowels /i ae a o u/ produced by a male model talker. With the exception of task instructions, the procedure was identical for both groups; one group, however, was given the instruction to explicitly imitate the model talker, while the other group was simply instructed to repeat the words naturally. We compare imitation across the two groups both acoustically by measuring vowel spectra and final stop patterns and auditorily by having listeners judge perceptual similarity. The results indicate (1) explicit and implicit imitation target different aspects of vowels; (2) the release of final stops are imitated in both conditions, but to a greater extent in the explicit condition; and (3) listeners perceive imitation in both conditions, but, again, to a greater extent in the explicit condition.

5aSC6. Phonetic convergence after perceptual exposure toward native and nonnative speakers. Midam Kim (1575 Oak Ave., #61, Evanston, IL 60201)

This study explores phonetic convergence by native English speakers after perceptual exposure to speech by either a native or a non-native English speaker. Three native and two Korean non-native English speakers read two sets of English words and sentences as the model speakers. A separate group of native English speakers read the two material sets, and was then exposed to one of the material sets either through auditory inputs read by a model speaker (experimental groups) or visual inputs (control groups), and read all materials again. Half of the participants were exposed to a native model speaker, and the other half to a non-native model speaker. Participants also conducted an implicit association test, which measured their attitudes toward native and foreign speakers of English. The first and second recordings of the test talkers were acoustically compared in terms of the Euclidian distances along acoustic-phonetic dimensions between participants' postexposure and the model productions, and between their pre-exposure and the model productions. Preliminary results illustrate that, at least for some acoustic measures, (1) participants with relatively large distances between their pre-exposure recordings and the model's recordings phonetically converged toward their models; and (2) participants generalized their convergence patterns to auditorily unexposed materials.

5aSC7. Prosodic and segmental conversion between speakers of different dialects. Jelena Krivokapic (Dept. of Linguist., Yale Univ., 370 Temple St., 302, New Haven, CT 06511, jelena.krivokapic@yale.edu)

Previous work has found that speakers converge to each other's production, as evaluated for acoustic properties of segments and in perceptual tasks [S. Fowler, J. Phonetics **25**, 421–426 (1995); Pardo, J. Acoust. Soc. Am. **119**, 2382–2393 (2006); Tobin, J. Acoust. Soc. Am. **125**, 2757 (2009)]. Little is known about convergence on prosodic properties [L. Schweitzer, J. Acoust. Soc. Am. **128**, 2458 (2010)]. An experiment is presented that examines how interaction between native speakers of British English and American English affects segmental and prosodic properties. The synchronous speech paradigm (Cummins, ARLO, **3**, 7–11 (2002); Z. Cummins Proceedings of Eurospeech 2003 (2003), pp. 777–780), where two speakers read sentences at the same time, is used. Dyads consisting of one British and one American speaker read a short story that contained four words where the two dialects differ in stress pattern and ten target words where the dialects differ in vowels. The following variables are examined: vowel formant structure, vowel duration, VOT, *F*0 contours, stress pattern, phrasing, and pitch accent. The goal is to investigate how speakers converge to each other's speech, and whether segmental and prosodic properties converge to the same degree. Data for dyads consisting of speakers of American English and Indian English are currently being collected.

5aSC8. Final stop accommodation in married couples. Sara E. Miller Newman (Dept. of English, North Carolina State Univ., Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, semnewman@yahoo.com) and Hayley E. Heaton (North Carolina State Univ., Raleigh, NC 27695-8105)

This study investigates convergence between married couples to determine if there are patterns of convergence and if that convergence can be attributed to social indexing or speaker accommodation. Preliminary work done with sociolinguistic interviews of residents of Harkers Island, NC demonstrated that the release of the final stop consonant /t/ has social meaning: Those residents who most identified with island life deleted or failed to release their final /t/ in casual speech, and female speakers were much more likely to release final /t/ than male speakers in their cohort. The two married couples examined in the Harkers Island data revealed two separate patterns of accommodation. One couple exhibited convergence that mirrored the male, island identifying spectrum where his percentage of stop release was 5.4% and hers was 0%. The other couple exhibited convergence that mirrored the female /t/ release spectrum with her percentage of stop release of 29.6% and his of 30%. These preliminary results suggest that not only is there phonetic accommodation between spouses, but also that this accommodation is not mutual convergence. It is hypothesized that patterns of phonetic convergence will emerge, and that these patterns will fall along the continuum of social indexing in the larger community.

5aSC9. Accommodation to sinewave speech is an online effect. James M. Hillenbrand and Michael J. Clark (Dept. of Speech Pathol. and Audiol., Western Michigan Univ., 1903 W. Michigan Ave., Kalamazoo MI 49008)

In earlier work [Hillenbrand *et al.*, J. Acoust. Soc. Am., 124, 2435 (2008)], we reported poor intelligibility (53.5%) for sinewave replicas of vowels in /hVd/ syllables. Tests with a separate group of listeners showed substantially higher intelligibility (73.1%) when the same syllables were preceded by a brief sinewave carrier phrase (CP). Additional tests showed the following: (1) a CP enhancement effect was seen even when the talkers used to generate the CP and the /hVd/ syllables did not match, showing that listeners are accommodating to the sinewave speech patterns and not solely to the talkers, and (2) a natural speech CP produced a *decrease* in intelligibility, showing again that listeners are accommodating in some way to the unusual characteristics of sinewave speech. In the present study, listeners identified sinewave /hVd/ syllables either alone or preceded by the sinewave CP in *alternating blocks* of trials (i.e., sinewave /hVd/ syllables alone and /hVd/ syllables preceded by a sinewave CP). Results using this alternating-block format showed that enhancement due to the CP comes and goes as the CP comes and goes. This finding suggests that the effect is an online or real-time effect. [Work supported by NIH.]

5aSC10. Variables influencing the size of carrier-phrase dependent effects on speech perception. Antonia D. Vitela and Andrew J. Lotto (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721)

Previous work has provided evidence that the presence/absence of carrier phrase effects on target speech sound categorization can be well predicted by comparing the long-term average spectrum (LTAS) of the carrier phrase with the spectrum of the target. In this series of studies, we move beyond asking whether there is a significant effect or not to examining the relative size of these effects. Target series of minimal-pair words varying in vowel were preceded by carrier phrases synthesized to simulate a talker with a large or short vocal tract. Shifts in vowel categorization percentage as a function of change in talker served as the variable of interest. Results indicate that manipulations of the perception of the carrier as speech (by presenting forward or backwards), the LTAS of the carrier (by gating the duration of the carrier presented), and the local information in the carrier (by preserving or not the part of the carrier immediately preceding the target) affected the size of the resulting shifts. Of particular interest, the LTAS has an affect above and beyond the spectrum nearest to the target and the carriers perceived as speech resulted in larger shifts even when LTAS was held constant. [Work Supported by NIH-NIDCD].

5aSC11. Efference copy and context effects. Mark Scott and Bryan Gick (Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada)

One hypothesized component of speech production is "efference copy"; a signal carrying the predicted sensory-consequences of the motor-system's actions [Wolpert and Ghahramani (2000)]. While brain-imaging studies [e.g., Aliu *et al.* (2009); Numminen *et al.* (1999)] have shown dampened auditory-cortex response to self-generated sounds (the predicted effect of efference copy), there are few behavioral demonstrations. This experiment will examine whether a context-effect (a common behavioral measure) is also dampened by efference copy. A context effect is a shift in categorization caused by surrounding sounds. For example, a syllable ambiguous between /da/ and /ga/ is perceived as more /da/-like when preceded by /ar/, but more /ga/-like when preceded by /al/ [Mann (1980)]. In this experiment, participants will silently mouth /ar/ or /al/ in time to a recording (it is assumed that mouthing engages efference copy). In one condition the recording will match what participants are mouthing; in another condition it will mismatch; in a third condition participants will hear the sounds without mouthing. After each mouthing, they will categorize a target syllable as /da/ or /ga/. The prediction is that the context effect will be dampened in the matching condition (due to efference copy), but not in the mismatching condition.

5aSC12. Effects of naturally-occurring segmental ambiguity on cross-modal priming of minimal pairs. Cynthia G. Clopper (Dept. of Linguist., Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Facilitation is observed in cross-modal lexical decision tasks when the auditory prime and visual target are the same word relative to when the prime and target are unrelated words. In addition, segmentally ambiguous primes can facilitate lexical decision for both members of a minimal pair. The goal of the current study was to explore the effects of naturally-occurring ambiguity on cross-modal priming in a lexical decision task. The targets consisted of minimal pair words containing the vowels /h, \æ/. The auditory primes were either the target word, its minimal pair, or an unrelated word, produced by talkers from the Northern and Midland dialects of American English. The results confirmed facilitation when the prime and target matched relative to when they were unrelated. In addition, inhibition was observed when the prime was the minimal pair of the target relative to when they were unrelated, but only for the Midland dialect. In the Northern dialect, these vowels are undergoing a change that makes them more perceptually confusable. Thus, these findings suggest that the less ambiguous Midland primes led to competition between the minimal pairs, whereas the more ambiguous Northern primes neither facilitated nor inhibited lexical access of the target word.

5aSC13. Cue weighting for speech categorization changes based on regularities in short-term speech input. Ran Liu, Howard Soh, and Lori L. Holt (Dept. of Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, ranliu@andrew.cmu.edu)

The ability to flexibly adapt long-term speech category representations to informative regularities in short-term input is critical for on-line speech perception. The present experiment investigates how short-term changes in the variability of two distinct acoustic cues affect the relative weighting of the cues for speech categorization. Native English adults distinguish the vowel categories /ae/ and /E/ using both spectral and duration cues. A spectral continuum from /ae/ to /E/ was crossed with a duration continuum between the same vowels to synthesize a 2-D grid of words ranging between "set" and "sat." Baseline categorization data were collected from trials sampling the full grid of words; for each trial, listeners selected whether they heard set or sat. Then, they received short-term exposure to trials drawn only from subsets of stimuli for which one cue was held constant while the other cue exhibited the full range of variability. Results reveal that listeners shift their relative cue weights, compared to baseline, to rely more on the highly variable cue for categorization. This suggests that listeners track the variability present across multiple acoustic cues and dynamically adjust speech categorization to reflect this short-term statistical regularity. [Work supported by NIH and NSF.]

5aSC14. Adaptive learning response to auditory perturbation of voice quality feedback. Kari Urberg-Carlson, Benjamin Munson, and Peter Watson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., Minneapolis, MN 55455, urbe0001@umn.edu)

Adaptive learning of speech behavior has been demonstrated in the areas of vowel height and backness [Houde (1998); Guenther (2006)] and pitch [Larson (1998)]. In each of these studies, the acoustic feedback presented to a speaker was perturbed, and most speakers modified their speech behavior to compensate. Adaptive learning is distinguished from feedback control if the compensatory behavior persists briefly once the perturbation is removed. The aim of the current study is to examine compensatory vocal behavior when voice quality is artificially perturbed. The common voice disorder, muscle tension dysphonia (MTD), in which a patient is hoarse in the absence of any physiological impairments, may be triggered by this type of compensatory behavior such as adaptation during an upper-respiratory infection. In this experiment, the perceived breathiness of speakers without vocal pathology will be artificially increased by mixing speech-shaped noise into the vocal feedback presented to the speaker. In pilot data with one speaker, when noise was added, spectral slant (H2-H1) increased. The increase in spectral slant indicates an increase in the closed phase of phonation, compensating for the perceived breathiness. Implications of these data will be discussed relative to the onset and maintenance of MTD.

5aSC15. Accent effects in auditory processing by native and non-native speakers. Kristin Franco and Margarita Kaushanskaya (Univ. of Wisconsin-Madison, Madison WI 53705)

The current study examined whether non-native listeners' perception is facilitated by a match between their L1 and the accent with which L2 words are pronounced. An auditory lexical decision task was used, where English words and nonwords were pronounced by a native speaker of English or a native speaker of Mexican Spanish. Half the words were cognates (overlapping in phonetic form and meaning across languages) while half were noncognates. Spanish-English bilinguals from Mexico and English-speaking monolinguals listened to accented and nonaccented stimuli and decided as quickly as possible whether they were English words. Results indicate an effect of lexical status and accent in both listener groups, with faster reaction times (RTs) for words than nonwords, and slower RTs on accented than nonaccented stimuli. However, English-speaking monolinguals were slower than Spanish-English bilinguals in recognizing accented English words. Moreover, Spanish-English bilinguals demonstrated faster RTs for cognates than for noncognates, but only when these were spoken with a Spanish accent. Results indicate that non-native accent influences auditory processing even in listeners who are familiar with the particular accent. Further, cognate effects indicate that lexical-phonological variables interact with acoustical factors in determining the efficiency with which the bilingual lexical system processes auditory information.

5aSC16. Prosody transfer in second language acquisition: Mandarin Chinese tonal alignment in English pitch accent. Miran Kim and Yu-an Lu (Dept. Linguist., Stony Brook Univ., Stony Brook, NY 11794, mi.kim@stonybrook.edu)

Tonal alignment patterns of Mandarin Chinese (L1) in production of English pitch accent are investigated in this study. Previous studies have observed that the F0 of lexical tone in Mandarin is closely aligned with the carrier syllable. This study examines whether this stability in the lexical tonal alignment is transferred to L2 prosody production, such as pitch accent in English. The F0 contours of native speakers of Chinese were analyzed for the English listing contour (L^*+H) and a sequence of lexical tones (low-rise) at varying speech rates. Regardless of the distinct prosodic structures between Chinese lexical tone and English pitch accent, our results indicate that the native tonal alignment pattern can influence L2 production, in which the listing contours produced by Chinese speakers are systematically associated with syllable edges. This tonal alignment pattern was consistent across variable speech rates; each tonal target was associated with a syllable edge. On the other hand, the tonal alignment in the English speakers' listing contour was not as stable as the Chinese pattern. The results suggest that Chinese speakers seem to reinterpret the pitch accent (L^*+H) similar to lexical tone sequences (e.g., L-R), and transfer their native pattern to the production of L2 prosody.

5aSC17. Long term average spectrum predicts accent normalization. Jingyuan Huang, Lori Holt (Dept. of Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213), and Andrew Lotto (Univ. of Arizona, Tucson, AZ 85721-0071)

Several studies have manipulated the accent of a spoken passage, observing shifts in subsequent speech perception. Such "accent normalization" may be related to remapping phonetic space or, alternatively, to tuning auditory representations via context. In the current experiments, artificial "accents" with precisely specified acoustic vowel distributions isolated the influence of these two potential mechanisms. Participants were exposed to a spoken passage drawn from Dr. Seuss' The Foot Book with /i/ shifted to /I/ (higher-$F1$), or /U/ shifted to /u/ (lower-$F1$), or with both manipulations and then categorized /i/-/I/ or /U/-/u/ series. The third context had shifted phonetic categories, but little change in the context's long term average spectrum (LTAS), known to influence subsequent speech categorization. As expected, higher-$F1$ contexts led to more low-$F1$ (/i/, /u/) target responses whereas lower-$F1$ contexts predicted more high-$F1$ targets (/I/, /U/) responses. However, we did not observe shifted vowel categorization after the context with little LTAS change (/i/ and /U/ shifted in opposing directions). Thus, although lexical information suggested shifted phonetic categories in this condition, there was no accent normalization. General auditory mechanisms sensitive to LTAS may play an important role in what has been thought to be accent-dependent remapping of phonetic space.

5aSC18. Musical training affects the representation of speech. McNeel G. Jantzen, Dane Aamodt, and Zachary Stratton (Dept. of Psych., Western Washington Univ., 516 High St., Bellingham, WA 98225, mcneel.jantzen @wwu.edu)

Musicians are more sensitive to acoustic features such as onset timing and frequency [Levitin (2006)]. Musical training may enhance the processing of acoustic information for speech sounds. The following questions were addressed to determine how musical training enhanced ability to perceive and represent a single speech sound in the presence of multiple sounds: (1) Does musical training improve performance on a dichotic listening test? (2) How does musical training affect the ability to attend to auditory streams? (3) How does the neural representation of the speech signal compare to behavioral results for musicians and nonmusicians? (4) Do musicians show more bilateral and/or right hemispheric activity when perceiving speech sounds. 30 subjects, 15 musicians, and 15 nonmusicians were presented with a voiced unaspirated stop consonant in one ear and a voiceless unaspirated consonant in the other ear such that all combinations were presented. Through five tasks, subjects identified a dominant speech sound, indicated location for a specified speech sound, and indicated speech sound for a specified location. Behavioral results indicate that musical training improved subjects' ability to perceive specified speech sounds. Musical training effects and organization of acoustic features were reflected in the EEG as observed by location and amplitude of the ERPs.

5aSC19. Perceptual and production training of Spanish intervocalic /r, d/ and tap. Wendy Herd (Dept. of Ling., Univ. of Kansas, 1451 Lilac Ln., Lawrence, KS 66049, wenherd@ku.edu)

This study investigates the effectiveness of three different high variability training paradigms in training 42 speakers of American English to correctly perceive and produce Spanish intervocalic /r, d/ and tap. English speakers have difficulty producing /r/, because it is a novel phoneme. Although the tap exists as an allophone in American English, learners have difficulty recategorizing it as a phoneme. Similarly, learners experience difficulty acquiring the spirantization of /d/. Since Spanish spirantization and English flapping both affect /d/ intervocalically, the acquisition of the /d/-tap contrast proves difficult. Past research reported that high-variability perceptual training improves both perception and production and that production training improves both as well. However, trainees were able to listen to stimuli during production training, making it unclear whether production training transfers to perception. This study systematically controls both training modalities so they can be directly compared and introduces another training methodology that includes both perception and production. All three training paradigms improved English learners' perception or production. While production trainees did not improve in their overall perception and declined in their perception of one contrast, perception trainees improved in their production and overall perception, indicating that perception transfers more effectively than production. [Research supported by the NSF.]

5aSC20. Auditory training: Stimulus exposure, task execution, and response feedback affect the neural detection of sound. Kelly Tremblay and Katrina McClannahan (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105)

The P1-N1-P2 cortical auditory evoked potential (AEP) has been used to study the effects of training-related plasticity in the human central auditory system. One consistent finding is that enhanced P2 amplitudes coincide with improved perception. A typical interpretation of this result is that auditory training alters the physiological representation of the cue being trained. However, it is also possible that changes in P2 amplitude reflect other processes that are associated with, but not specific to, the improved representation of the trained cue itself. Here we isolated different components of auditory training by examining the effects of stimulus exposure, behavioral task execution, as well as task- plus-feedback on the AEPs of 30 normal-hearing young adults. For each group, AEP responses were passively obtained in response to two speech syllables differing in voice-onset-time, on three separate days. The first two sessions were on consecutive days and the third session occurred 1 week later. Results suggest that mere stimulus exposure alters the way sound is represented in the human auditory system, but the magnitude of P2 amplitude change is significantly greater when participants interact with the stimuli while executing a task. This point is especially true with the addition of feedback. [Work supported by NIH NIDCD Grant No. R01DC007705.]

5aSC21. Training listeners to report fundamental frequency and formant range information independently. Santiago Barreda and Terrance M. Nearey (Dept. of Linguist., Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, sbarreda@ualberta.ca)

The vowels of speakers of different sizes vary in terms of their average f0 and formant frequencies. In general, larger speakers produce vowels with lower formant frequencies and lower f0s. Listeners have demonstrated the ability to estimate the approximate size of a speaker and previous experiments have shown that these judgments are based on the joint consideration of f0 and formant range information. Thus both lower f0s and lower formant frequency ranges are associated with larger speakers by listeners. Studies which have asked listeners to evaluate voices have focused on the extraction of apparent speaker characteristics (which are informed by f0 and formant frequencies) rather than asking speakers to report f0 and formant range information directly. The current study consists of a training procedure by which participants will learn to report the f0 and formant range of voices independently. The training consists of a voice matching game in which participants hear a pair of vowels produced by a voice and are asked to indicate which of the candidate voices they just heard. Results will be analyzed in light of current theories of vowel perception and normalization.

5aSC22. Constraints on the learning of foreign accents. Alison M. Trude and Sarah Brown-Schmidt (Dept. of Psych., Univ. of Illinois at Urbana-Champaign, 603 E. Daniel St., Champaign, IL 61820, trude1@illinois.edu)

Understanding foreign-accented speech presents challenges for listeners. Three experiments tested learning of a foreign-accented vowel shift and its application during on-line speech processing. Native Quebec French and English speeches were compared using a visual world eye-tracking task. The French talker pronounced /i/ as /ɪ/ before all consonants except voiced fricatives. On critical trials, participants viewed pictures of an unaccented /i/ word (e.g., bees) and accented /i/ word (e.g., beet, pronounced [bt] by the French talker), then heard one talker say the target word. On French-talker trials, listeners should rule out the competitor because it does not share a vowel with the target, reducing competition. However, fixation measures showed comparable competition on "bees" trials and increased competition for the French talker on "beet" trials, indicating difficulty in applying knowledge of the accent. Performance on bees trials improved in Exp2-3, where stimulus variability was reduced by presenting only critical words with /t/ codas. On beet trials, performance improved most on Exp3, in which the unaccented counterpart to the accented word (e.g., bit, for beet) never appeared. The results suggest reducing linguistic variability is helpful in learning the accent, but lexical competition must also be reduced for successful processing of accented words.

5aSC23. Underlying sources of individual differences: Cognitive abilities in second language phonological development. Hanyong Park (Dept. of Linguist., Univ. of Wisconsin-Milwaukee, Curtin Hall 523, P.O. Box 413, Milwaukee, WI 53201, park27@uwm.edu), Isabelle Darcy, and Chung-Lin Yang (Indiana Univ., Bloomington, IN 47405)

Research on individual differences has identified factors constraining L2 acquisition in terms of a global performance; yet little progress has been made in identifying specific predictors of phonological acquisition. To explore such potential predictors in L2-learners, working memory, selective attention, processing speed, vocabulary size, and naming speed in L1 and/or L2 were assessed. These scores were then compared to individual L2-phonological acquisition index scores, which were obtained by combining the scores from three tasks: an ABX task (testing acquisition of English phonetic categories), a sequence repetition task (testing acquisition of English stress), and a speeded lexical decision task (testing the encoding of English syllable structure). The results were based on data from 20 Korean learners of English with varying lengths of US-residence (LOR) and 10 English native speakers as controls. In all the tasks, the native speakers performed better than the longer-LOR learners, who, in turn, did better than the shorter-LOR learners. There were large individual differences within each group. Individual working memory measures were significantly correlated with the individual L2-phonological acquisition index scores. Further, specific task scores correlated independently with specific cognitive measures, suggesting that a balanced mix of cognitive abilities can be the key to better phonological acquisition. [NIH-NIDCD Training Grant No. T32-DC00012.]

5aSC24. Intra- versus inter-talker effects in multi-talker perceptual training. Richard Wright (Dept. of Linguist., Univ. of Washington, Box 354340, Seattle, WA 98195-4340, rawright@u.washington.edu)

Learners of non-native speech sounds benefit from intensive multi-talker training [e.g., Logan *et al.* (1991), Yamada (1993), and Bradlow *et al.* (1999)]. This training is thought to produce highly generalized perceptual learning and result in long-term modifications of the perceptual system. However, it is unclear whether the multi-talker benefit is due to inter-talker variation, to intra-talker variation, or to some combination thereof. In this experiment intra-talker (style) variation was fully crossed with inter-talker variation. 48 subjects were trained in one of four set stimuli [(1) single-talker single-style, (2) single-talker dual-style, (3) dual-talker single-style, and (4) dual-talker, dual-style]. Stimuli were nonsense words composed of either all native sounds or containing non-native speech sounds ([y] or [x]) presented with randomly paired nonsense shapes (fribbles). Subjects were probed at each quartile of the training to monitor learning of fribble names (correct versus incorrect fribble-name pairing). Subjects were then tested in a generalization task with new talkers. Results of this experiment indicate an important role for intra-talker variation in the ability of learners to generalize to novel exemplars of trained words. This result holds for both types of stimuli: nonsense native and nonsense non-native, indicating that multitalker training benefits extend to native word learning.

5aSC25. Benefits of voice gender and perceptual learning on the perception of masked speech processed through cochlear implant simulations. Jessica R. Sullivan (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 N.E. 42nd St., Seattle, WA 98105-6246, sulli10@uw.edu), Peter Assmann, and Shaikat Hossain (Univ. of Texas at Dallas, Richardson, TX 75083)

Cochlear implant simulations were used to investigate the potential benefits of auditory perceptual training for cochlear implant users. Listeners with normal hearing attended to speech masked by a competing talker, processed through a simulation of an eight-channel cochlear implant. The aim was to determine whether perceptual training would enable listeners to benefit from differences in voice gender between target and masker. Thirty adults with normal hearing were randomly assigned to one of three training groups: matched-gender word recognition, mismatched-gender word recognition, or gender recognition. Participants completed 2 h of training within a 1-week period then returned 1 week later for a late-post testing session. Training sessions consisted of ten blocks during which the SNR (-6, 0, 6 dB) was adapted based on performance. Significant speech intelligibility improvements were observed from pre- to post-sessions for all three training groups, and these improvements were maintained at the late-post session for all three groups. Mismatched-gender word recognition training produced a larger improvement in word recognition accuracy than matched training; however, gender recognition training also yielded substantial improvement, suggesting that voice gender cues contribute to speech recognition in competing talker situations for cochlear implant listeners.

5aSC26. Learning acoustically complex word-like units within a video-game training paradigm. Sung-joo Lim, Lori L. Holt, and Francisco Lacerda (Dept. of Psych., Carnegie Mellon Univ., 5000 Forbes Ave., PA 15213, sungjol@andrew.cmu.edu)

Over the course of language development, infants learn native speech categories and word boundaries from speech input. Although speech category learning and word segmentation learning occur in parallel, most investigations have focused on one, assuming somewhat mature develop of the other. To investigate the extent to which listeners can simultaneously solve the categorization and segmentation learning challenges, we created an artificial, non-linguistic stimulus space that modeled the acoustic complexities of natural speech by recording a single talker's multiple utterances of a set of sentences containing four keywords. There was acoustic variability across utterances, presenting a categorization challenge. The keywords were embedded in continuous speech, presenting a segmentation challenge. Sentences were spectrally rotated, rendering them wholly unintelligible, and presented within a video-game training paradigm that does not rely upon explicit feedback and yet is effective in training non-speech and non-native speech categorization [Wade & Holt (2005); Lim & Holt (submitted)]. With just 2 h of play, adult listeners could reliably extract word-length sound categories from continuous sound streams and generalized learning to novel tokens. The amount of "sentence" variability within training did not influence learning. [Research supported by NIH, NSF, and Riksbanken].

5aSC27. The influence of amplitude modulation frequency on perceived roughness of vowels. Rahul Shrivastav and David A. Eddins (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017 Tampa, FL 33620, deddins@usf.edu)

The goal of the current experiment is to determine how perceived roughness of vowels is influenced by the frequency of inherent amplitude modulation. A set of 10 vowel stimuli was synthesized using a Klatt synthesizer and were modeled after ten talkers selected from the Satloff/Heman-Ackah disordered voice database. These vowels were selected using stratified sampling procedures to represent a wide range of roughness. The vowel f0 and the first three formants were determined and used to generate a synthetic copy of these vowels. The aspiration noise level was adjusted to subjectively match the overall quality of the talkers. To each of the ten prototypical vowels, sinusoidal amplitude modulation was superimposed with a modulation depth of -6 dB and modulation frequencies ranging from 10 to 70 Hz. Listeners judged roughness in a matching task where the modulation depth of a comparison sound was adjusted to match the perceived roughness of each of the 70 speech stimuli (10 talkers $\times$ 7 modulation frequencies). The dependence of perceived roughness on amplitude modulation frequency will be compared to the perceived roughness of sinusoidal carriers as reported by Fastl and Zwicker (Springer, New York, 1997).

5aSC28. Intelligibility of dual narrow passbands with mismatched bandwidths: Anomalous findings and their rules. Richard M. Warren, James A. Bashford, Jr., and Peter W. Lenz (Psych. Dept., Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201)

Unlike paired 1-octave bands, intelligibility of paired 1/3-octave speech bands can exhibit extreme hyperadditivity when heard together. Thus, for everyday sentences, the intelligibilies of 1/3-octave (4 semitones) rectangular speech bands centered at 1 and 3 kHz were 14% and 6%, respectively; when heard together intelligibility was 75%. In experiment 1, each of these bands in turn was chosen as the pedestal and remained fixed at 4 semitones and was paired with bandwidths of the second band at 3, 2, 1, or 0.5 semitones. Intelligibilities for each pairing were the same regardless of which band served as the pedestal despite the difference in the constituent intelligibilities. For example, the intelligibility of paired bands consisting of a 4 semitone pedestal and a 2 semitone variable (total bandwidth of 6 semitones) was 60% regardless of which band served as pedestal. In experiment 2, the same total bandwidth of 6 semitones was produced by setting the bandwidths of each of the bands at 3 semitones: Once again the intelligibility was 60%. Additional experiments have supported these novel findings concerning the intelligibility of paired speech bands. Implications will be discussed including the relevance for cochlear implant design. [Work supported by NIH.]

5aSC29. Effects of low-pass filtering and accent in a single-talker interference task. Miriam O. Krause, Peggy B. Nelson, and Mary R. T. Kennedy (Speech-Lang.-Hearing Sci. Dept., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, krau0067@umn.edu)

Non-native-accented speech can be less intelligible than native-accented speech [Lang Learning **49**, 285–310]. Furthermore, low-pass filtering can reproduce the reduced audibility of high-frequency hearing loss [JSLHR **21**, 5–36]. The current study combines these two lines of inquiry using a sentence-repetition task in which monolingual listeners repeated sentences spoken by native- and non-native-accented speakers of English. The dependent measure was percent of keywords repeated accurately, and the study

used a crossed design. One group repeated unfiltered speech and the other repeated speech low-pass filtered at 1400 Hz. For each group, listeners repeated sentences from each talker separately in the "alone" condition, while in the "attend" condition they heard both speakers simultaneously and repeated only one at a time. As expected, the filtered group performed significantly less accurate than the unfiltered group for both the alone and attending conditions. However, while the control group was able to maintain performance in the attending condition regardless of accent, listeners to filtered speech were roughly half as accurate for the non-native-accented speaker compared to the native-accented speaker in both conditions. Results suggest that peripheral hearing effects overwhelmed attentional effects in determining ability to repeat sentences accurately for these listeners. [Research supported by University of Minnesota Graduate School.]

5aSC30. Effects of spectral degradation on contextually driven shifts in phonetic categorization. Ariane E. Rhone (Dept. of Linguist., Univ. of Maryland, 1401 Marie Mount Hall, College Park, MD 20742) and Matthew B. Winn (Univ. of Maryland, College Park, MD 20742, mwinn@hesp.umd.edu)

Under conditions of spectral degradation, listeners have been shown to experience deficits in perception of some consonant and vowel contrasts that are driven by spectral cues. In everyday listening situations, cues to phonetic contrasts are dynamic, owing to varying listening contexts both across- and within-talkers. This adds an added layer of spectral information not typically captured in assessment of cochlear implant users or normal-hearing listeners in simulated conditions. A series of experiments was designed to assess the ability of these listeners to use spectral cues to normalize and adjust phonetic boundaries across talkers and across various phonetic contexts. These tasks involved the adjustment of the alveolar/palatal fricative boundary, which has previously been shown to be sensitive to talker gender as well as vowel context. Preliminary results suggest that when spectral resolution is degraded, listeners show an impaired ability to adjust to these contexts. It is also suggested that visual information can substantially mitigate these deficits, at least for the phonetic contrasts tested here. These results help to explain some of the difficulties experienced by cochlear implant users in common conversational settings, and reinforce the importance of compensatory strategies (such as audio-visual integration) to aid in perception.

5aSC31. Divergent patterns of voicing perception in various challenging listening conditions. Matthew B. Winn and Monita Chatterjee (Dept. of Hearing and Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20742, mwinn@hesp.umd.edu)

Perception of consonant voicing is especially robust, even under conditions of harsh spectral degradation, hearing loss, and background noise. Considering the limitations and variability of the acoustic input, it is unlikely that the same perceptual strategy is used in all situations. A sequence of experiments was designed to assess the weighting of multiple acoustic cues to the voicing contrast word-initially and word-finally, and to reveal patterns unavailable in conventional confusion matrices and information transfer analysis. In preliminary results, the combination of background noise and low-pass filtering brought about a dramatic shift in weighting of the VOT and F0 cues for initial consonants; these degradations in isolation did not produce such a shift. Under conditions of spectral degradation (noise-band vocoding), listeners significantly increased their use of vowel duration, and significantly decreased their use of vowel formant transition, which is dominant in the optimal condition. These results suggest that in challenging conditions, listeners are apt to use secondary acoustic cues that are potentially less reliable or less perceptually salient than primary cues used in optimal conditions. It is concluded that comparison of performances by listeners in various conditions should be guarded, even when recognition scores appear to be equivalent on the surface.

5aSC32. The intelligibility of Lombard speech produced and perceived in different noise conditions. Abby J. Walker (The Ohio State Univ., Columbus, OH 43215, ajw129@gmail.com)

Lombard speech is typically found to be more intelligible than speech produced in quiet when played in noise. In this study nonwords produced by 6 speakers in 4 different noise conditions (6-talker babble, and unfiltered, low-pass filtered, and high-pass filtered white Gaussian noise) were played to 100 participants in the presence of matching or mismatching noise at -5 dB signal-to-noise ratio (SNR). There were two sets of nonwords, which differed minimally in either consonant (aCa words) or vowel (shVb words). Participants were asked to identify which nonword they heard. While there was no significant effect of matching or mismatching noise conditions on intelligibility, there was a difference between the intelligibility of speech produced in harder (babble and unfiltered white noise) or easier (low and high pass noise) conditions: If speech was produced in a more difficult noise condition, that speech was more intelligible overall. Additionally, this study found that in many noise conditions, quiet speech was equally or more intelligible than Lombard speech, a finding contrary to many studies. This result is discussed in relation to methodological issues involving the relative ease of the task, and the effect of the Lombard effect on vowels versus consonants.

5aSC33. Speech intelligibility in noise for adult and children users of cochlear implants. Rajka Smiljanic (Linguist., Univ. of Texas–Austin, 1 University Station B5100, Austin, TX 78712-0198) and Douglas P. Sladen (Univ. of Texas–Austin, Austin, TX 78712)

Even though high levels of speech understanding in quiet can be developed postimplantation, speech recognition in noise is an especially challenging task for cochlear implant (CI) users. This study examined whether adult and children CI users benefit from signal clarity and contextual-semantic information to improve their speech recognition in noise. Ten adult and ten children CI users, and ten adults and ten children with normal hearing participated in sentence-in-noise listening tests. They heard sentences in which final word varied in predictability, i.e., high versus low semantic context, produced by one female and one male talker in conversational or clear speaking styles. The goal was to assess the effect of age and the length of experience with using cochlear implants on the ability to utilize the compensatory information at higher levels of speech processing independently and in combination with the acoustic-phonetic enhancements in speech perception. This research will allow us to explore the interaction between lower-level sensory and higher-level cognitive factors that affect speech processing in noise for adults and children with cochlear implants. The results of this study add to our current understanding of the development of speech recognition for individuals with cochlear implants and with normal hearing.

5aSC34. Does spatial release from masking interact with working memory capacity in speech perception? Alexander L. Francis, Katie Connell, and Leigh Anderson (Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN 47907, francisa@purdue.edu)

Increasing the spatial separation of target and masking speech improves speech recognition but this benefit tends to decline with age. Previous research suggests that perceived spatial separation of targets and maskers facilitates selective attention, a mechanism that depends in part on the availability of working memory capacity. Thus, age-related loss of benefit from spatial separation may increase demands on working memory capacity for speech perception. The present study investigates the interaction between spatial separation of target and masking speech and the availability of working memory capacity, reporting results from an experiment in which listeners repeated sentences in the presence of masking speech under conditions of low- and high-working memory demands imposed via a simultaneous memory load task. Stimuli were presented from a speaker located in front of the listener with two-talker masking speech presented from one of two speakers located either 10 deg or 90 deg to one side. Preliminary results from younger adults suggest no effect of load on sentence recognition, but poorer memory task performance with 10 deg as opposed to 90 deg separation, supporting the hypothesis that spatial separation reduces working memory demand for recognizing speech in competing speech. Results from older listeners will also be discussed.

5aSC35. Talker variability in lexical access: Evidence from semantic priming. Yu Zhang and Chao-Yang Lee (Div. of Commun. Sci. & Disord., Grover Ctr. W239, Ohio Univ., Athens, OH 45701, yz137808@ohio.edu)

It is usually assumed that lexical representations are abstract and devoid of detailed information about various sources of acoustic variability. This assumption is evaluated in this study by examining the effect of talker variability on the access to word meaning in a short-term semantic priming experiment. Prime-target pairs that were semantically related (e.g., king-queen) or unrelated (e.g., bell-queen) were produced by the same talker or different talkers. Two interstimulus intervals (50 and 250 ms) were used between the prime and target to explore the time course of semantic priming. The auditory stimuli were presented to 60 listeners, whose task was to judge whether the target was a word or nonword item in English. It was hypothesized that a change in talker between the prime and target would influence the magnitude of semantic facilitation. Analysis of response accuracy and reaction time showed that the magnitude of semantic priming was attenuated in the different-talker condition, although the effect was obtained only for targets produced by the female speaker. There were no effects involving different interstimulus intervals. These results provided partial evidence that talker variability is encoded in lexical representations and affects lexical access.

FRIDAY MORNING, 27 MAY 2011 METROPOLITAN A, 8:45 TO 11:40 A.M.

Session 5aSP

Signal Processing in Acoustics, Engineering Acoustics, Underwater Acoustics, and Animal Bioacoustics: Detection and Classification of Buried and Proud Targets I

Jason E. Summers, Cochair
Applied Research in Acoustics, 1222 4th St., S.W., Washington, DC 20024

Patrick J. Loughlin, Cochair
Univ. of Pittsburgh, Dept. of Bioengineering, Pittsburgh, PA 15261

Chair's Introduction—8:45

Invited Papers

8:50

5aSP1. Application of low-frequency methods for estimating object size. Jack McLaughlin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jackm@apl.washington.edu), Brandon Hamschin (Univ. of Pittsburgh, Pittsburgh, PA 15261), and Greg Okopal (Univ. of Washington, Seattle, WA 98105)

Classification of submerged objects has traditionally been performed using high frequency sonars and imaging techniques. While this permits fine matching of target templates to images acquired in the field, HF methods are necessarily limited in range due to absorption of sound by the water. LF sonars, while offering increased detection range, come with some significant challenges related to the limited bandwidth available. Nonetheless, we show that it is feasible to estimate object size using nonimaging techniques. There are a number of low-frequency phenomena that can be exploited to this end. Among these are edge diffraction in which sharply angled facets of objects ("edges") act like independent, radiating point sources, and helical waves, which can be set up in cylindrical objects. We show that with appropriate postprocessing of these returns, object edges can be localized thus allowing object extent to be assessed. In this paper, we describe our processing system, and then give results when this system is applied to over 40 sequences of returns from a rail system. In each sequence, a single solid, proud cylinder is insonified, and our system reports an estimate of cylinder length and radius. Histograms of these estimates cluster roughly around the true values.

9:10

5aSP2. Bluefin12 buried mine identification system. Richard Holtzapple (NSWC Panama City Div. Code X11, 110 Vernon Ave., Panama City, FL 32405, richard.holtzapple@navy.mil)

The Bluefin12 buried mine identification system, developed under the sponsorship of the Office of Naval Research (ONR) as a close range target reacquire and identify system, is based on the 12.75-in. unmanned underwater vehicle (UUV) developed by Bluefin Robotics. This system is integrated with a multi-sensor package that consists of the buried object scanning sonar (BOSS) system, a passive multi-axis real-time tracking gradiometer and an electro-optic sensor. The BOSS system is a downward looking sonar system that uses a broad-band, low-frequency omni-directional acoustic projector, and 20-element hydrophone receive arrays embedded in each of two 1-m length wings. The BOSS provides detection and classification of not only proud but also partially and fully buried targets. Beam forming and synthetic aperture processing generates multi-aspect three-dimensional imagery consisting of *X-Y* (top), *X-Z* (front), and *Y-Z* (side) perspective views providing target dimension and orientation and where the *X-Z* and *Y-Z* views provide target burial depth information. This paper presents a brief history of the development and integration of the BOSS system on the Bluefin12 UUV and specific results obtained during field surveys and system demonstration tests held in Panama City, FL and other sites.

9:30

5aSP3. *In situ* learning techniques for underwater object classification. Patrick Rabenold, Hui Li (Signal Innovations Group, 1009 Slater Rd., Ste. 200, Durham, NC 27703, prabenold@siginnovations.com), and Lawrence Carin (Duke Univ., Durham, NC, 27708)

A Bayesian *in situ* learning framework is presented that integrates several techniques to dynamically learn and adapt to variable sensing environments and the resulting manifestations of objects of interest. Sensing data are often collected from an environment for which little or no *a prior* knowledge is available. The presented framework exploits the contextual information of unlabeled data by learning a semi-supervised classifier on the complete data manifold. An active learning component augments limited labeled data through the adaptive selection of unlabeled samples that, given acquired labels, minimize uncertainty in the classifier. Both myopic and nonmyopic active learning methods are presented, where nonmyopic selection of the most-informative subset of samples is an extension of the myopic approach and leverages properties of submodular set functions. Additionally, the framework addresses the problem of

imperfect acquired labels by accounting for uncertainty in a label within the classifier design. The benefits of the *in situ* learning techniques are demonstrated through the application to underwater object recognition. [This work was supported by the Office of Naval Research.]

9:50

5aSP4. Direction of arrival estimation of near-field sources using sparse representation framework. Yinghui Zhao (Ctr. of Earth Observation and Digital Earth, Chines Acad. of Sci., Beijing 100094, China, yhzhao@ceode.ac.cn), Mahmood R. Azimi-Sadjadi, and Amanda Dinstel (Colorado State Univ., Fort Collins, CO 80523)

This paper uses the sparse representation framework to investigate localization of near-field sources (e.g., underwater bottom or buried targets) from the data captured using two uniform linear sensor (hydrophone) subarrays. The connection between the two array steering transformation matrices of the near-field sources corresponding to the two subarrays is first analyzed using the second Taylor expansion. This connection allows the construction of a new equivalent far-field steering matrix for each near-field source, hence converting the near-field source localization problem to a more convenient far-field one. The relationship between the signals observed by the two subarrays and the new constructed far-field directional matrix is investigated indicating that the DoA estimates of the sources can be cast into finding a solution to a sparse representation problem. The resolution of the sparse equation will also be discussed. Finally, simulation results are presented to demonstrate the effects of different noise levels on the accuracy of the DoA estimation for the cases of single and multiple sources.

10:10—10:25 Break

Contributed Papers

10:25

5aSP5. Comparison of time-frequency distributions for target classification. Rodolfo Arrieta (Naval Surface Warfare Ctr., PCD, Code X12, 110 Vernon Ave., Panama City, FL 32407, rodolfo.arrieta@navy.mil), Richard Holtzapple, and Raymond Lim (PCD, Panama City, FL 32407)

Time frequency representations characterize signals in the time-frequency plane and can aid in the physical interpretation of backscattered chirps. However, these 2-D representations force trade-offs in resolution, computational efficiency, noise reduction, and cross-term generation. We are mainly concerned with the ability of the transform to reject noise, but we are also concerned with the effect of the cross-term artifacts that do not represent real phenomena on the ability to classify targets using machine learning approaches. First, strong backscatter returns available for a particular target were identified. In the cluttered field under study, this was accomplished from a synthetic aperture image of the field. The processing reported herein used these returns from each studied target to produce a partition of the time frequency plane that contained disjoint sets of time-frequency regions. Each target was represented by a subset of these regions. To test the separability of the targets based on these subsets, random pings were selected from each target's high backscatter region. These pings were then compared against the disjoint sets. This procedure was carried out with several different time-frequency methods in order to determine the importance of the method's resolution, noise reduction quality, and cross-term artifacts, in producing robust time-frequency plane partitions that may be used to classify targets. [This work was sponsored by the SERDP.]

10:40

5aSP6. Bistatic detection and imaging of proud targets in shallow water using coherent space-time-frequency processing. Shaun D. Anderson and Karim G. Sabra (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405)

An ongoing challenge for underwater sonar systems is to discriminate a man made target (typically having an elastic shell) from surrounding clutter returns especially in the presence of multipath. Network of autonomous systems of unmanned vehicles provides a practical means for bistatic measurements and thus potentially bistatic enhancement for target detection. One popular technique for automatic target detection is synthetic aperture imaging. An aspect that complicates this technique is the evolutional, time dependent aspect of the bistatic echo spectrum from elastic shells especially for the acoustic guided waves echo components at low to midfrequency. To this end, time-frequency analysis, and, in particular, Wigner-Ville analysis have been shown to provide a robust processing tool for interpreting the evolutional time dependent aspect of the scattered acoustic wave field especially around the midfrequency enhancement. Time-frequency methods will be used to enhance the sonar images created from monostatic or bistatic low-frequency measurements. The proposed approach will be tested using numerical simulations and experimental at-sea data. [Work supported by the ONR.]

10:55

5aSP7. Improving classification of underwater objects by optimal signal design. Brandon Hamschin (Dept. of Elec. & Comput. Eng., Univ. of Pittsburgh, 348 Benedum Hall, Pittsburgh, PA 15261, bmh161@gmail.com) and Patrick Loughlin (Univ. of Pittsburgh, 749 Benedum Hall, Pittsburgh, PA, 15261)

Detection, classification, and localization of underwater objects is a primary function of active sonar systems. Detection involves making a decision on whether or not an object of interest is present. Once a positive detection has been made, further information may be needed to classify the object as one among a set of possible objects of interest. Previous efforts have been directed at designing transmit sonar waveforms to maximize detection performance. In this work, we extend the optimal sonar design approach to enhance classification after detection. In particular, we present an optimal signal design approach that is aimed at maximizing the probability of correctly classifying the true target from among a set of assumed candidates. The approach is evaluated theoretically and via simulations, by which it is shown that waveform design can yield improvements in classification performance. [Work supported by ONR.]

11:10

5aSP8. Sound scattering by a small sphere. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Earth System Res. Lab./Physical Sci. Div. Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

Acoustic Green's functions for a homogeneous medium with an embedded spherical obstacle arise in analyzes of scattering by objects on or near the seafloor, radiation by finite sources, sound attenuation in and scattering from clouds of suspended particles, etc. An exact solution of the problem of diffraction of a spherical sound wave on a sphere was obtained by Lord Rayleigh in 1872 and is given by an infinite series involving products of Bessel functions and Legendre polynomials. In this paper, a simple, closed-form solution is obtained for scattering by a sphere with a radius small compared to the wavelength. The solution is valid for arbitrary positions of the source and receiver relative to the scatterer. Low-frequency scattering is shown to be rather sensitive to boundary conditions on the surface of the obstacle. Relation is discussed of the asymptotic low-frequency solution to the exact solution due to Lord Rayleigh and to Lord Kelvin's exact solution (1845) of a corresponding problem in electrostatics.

11:25

5aSP9. A method of image fusion based on bidimensional empirical mode decomposition. Wan Jian, Zhao Chunhui, and Ren Longtao (Information and Commun. Eng. College, Harbin Eng. Univ., 150001, China)

Based on the principle of bidimensional empirical mode decomposition (BEMD), a new method of image fusion is presented. The processing procedure can be divided into three steps: (1) Every image to be fused is decomposed to several intrinsic mode functions (IFMs) and a residual component by BEMD; (2) according to the characteristic of BEMD, the IFMs in different frequency bandwidths are fused separately into new components, using linear weighted rule, whose coefficients can be required by the calculation of the mean-square-error or the information-entropy of the IFMs; (3) these new components are fused finally, obtaining the fused image. The simulation results show that the image fusion based on BEMD using the character of frequency has more advantages than that using the local energy of the pixel, and is also superior to the wavelet image fusion on pixel level.

FRIDAY MORNING, 27 MAY 2011 GRAND BALLROOM D, 7:55 A.M. TO 12:30 P.M.

Session 5aUW

Underwater Acoustics: Acoustic Communications

Daniel Rouseff, Chair

Univ. of Washington, Applied Physics Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698

Chair's Introduction—7:55

Contributed Papers

8:00

5aUW1. Investigation of the environmental factors affecting a horizontal underwater acoustic communication channel. Grant M. Pusey and Alec J. Duncan (Ctr. for Marine Sci. and Technol., Curtin Univ., GPO Box U1987, Perth, Western Australia 6845, Australia)

This study sought to characterize the long-term performance of a horizontal underwater acoustic communication system using experimentation and channel simulation. A long-term investigation was carried out off the coast of Western Australia in a depth of 100 m, simultaneously collecting environmental and modem communication performance data for over 16 days. Additionally, an underwater acoustic communication simulator based on the Bellhop propagation model was developed to provide performance predictions for the deployed modems. The signal-to-noise ratio was measured to be relatively high for the entire duration of the trial which helped classify environmental influences as being related to other parameters. Using multiple linear regression, it was determined that the sound speed profile, sea surface roughness, and signal-to-noise ratio contributed to the increased performance of underwater acoustic communication. This result was confirmed by implementing a simulation over the area of interest in a modeled environment and by updating the environmental parameters for each time step. By progressively adding more information to the simulator including ambient noise, wave height, and the sound speed profile, simulations provided effective predictions of the performance measured during the trials.

8:15

5aUW2. A fast algorithm for computing doppler introduced by sea surface gravity waves. John C. Peterson (Heat, Light & Sound Res., Inc., 3366 N. Torrey Pines Court, Ste. 310, La Jolla, CA 92037)

The technical advances in phase coherent underwater acoustic communications and the increasing availability of relatively inexpensive commercial modems have created a renewed interest in utilizing sound channel models for predicting modem performance. While accurate channel models already exist, their usefulness for modeling extensive "what-if" scenarios is limited by the computational resources required. Acoustic field models based on raytracing methods are a fast and efficient approach to sound channel modeling. They readily lend themselves to estimating the impulse response or time arrival structure of a hypothetical sound channel. The computed impulse response can then be convolved with the waveform transmitted by the modem to obtain an estimate of the timeseries observed at a hypothetical receiver. The existing models for addressing the Doppler effects introduced by sea surface gravity waves into the receiver timeseries are accurate, but relatively resource hungry. A fast algorithm for modeling Doppler based on simple postprocessing of the eigenrays computed by the raytracing based field model will be presented.

8:30

5aUW3. Simplified formulations for sea-surface scattering for use in modeling equalizer performance in underwater communications. Daniel Rouseff and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Reflection off the rough sea surface typically introduces time spread in an acoustic signal. The details of channel response and hence the time spread will change as the sea surface evolves. Communications signals that are spread by the rough surface may still be recompressed and demodulated successfully by an equalizer providing that the channel response does not change too rapidly. To aid in designing an equalizer, it would be useful to know which acoustic paths should be treated as useful signal and which must be treated as noise because they change too rapidly. Viewed as a rough surface scattering problem, the classic Kirchhoff approximation should be appropriate for modeling reflected communications signals. Textbook descriptions of the Kirchhoff approximation are not promising, however, as they imply that the calculations depend on the details of the surface wave spectrum. In the present work, simplified expressions for the surface reflected communications signals are derived. The simplified results are tested in two ways: Predictions for the mutual coherence function are compared to numerically intense calculations, and predictions for communications performance are compared to experimental results. [Work supported by the ONR.]

8:45

5aUW4. Differential frequency hopping performance in Doppler-inducing underwater acoustic communication channels. Luca Cazzanti, Arindam K. Das (Appl. Phys. Lab, Univ. Washington, 1013 NE 40th St., Seattle, WA 98105, luca@apl.washington.edu), Dianne E. Egnor, and Geoffrey S. Edelson (BAE Systems, MER15-2651, P.O. Box 868, Nashua, NH 03061-0868)

The relationships between the parameters of Doppler spread-inducing underwater environments and the bit-error performance of differential frequency hopping (DFH) modulation in the underwater acoustic channel are characterized. Wind speed determines the nature of the effect that the water surface imposes on the acoustic DFH wavefoms propagating underwater. Low wind speeds result in an essentially flat, low-absorption sea surface. In this regime, strong surface reflections and little frequency spreading make intersymbol interference (ISI) the dominant effect on the received waveforms. At high wind speeds, the higher density of air bubbles in the surface layer absorbs almost all energy incident on the surface, resulting in no surface reflections reaching the receiver. In this regime, the surface has little effect on the received signal, either in the form of ISI or frequency spreading. The intermediate ranges of wind speed, with a mix of ISI and surface-induced Doppler spread, pose the most challenging conditions. Simulations and at-sea experiments show that recent algorithmic improvements to the receiver make DFH robust to a variety of environmental conditions and that DFH modulation parameters can be easily adapted to a variety of operationally relevant scenarios based on environmental information.

9:00

5aUW5. Subspace dimension estimation for equalization of underwater acoustic channels. Ballard J. Blair and James C. Preisig (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, bblair@whoi.edu)

The use of an array of receiving elements has several advantages such as increased signal to noise ratio due to array gain and the ability form beams, which reject interference and enhance a source. One common way to integrate an array receiver into a communication system is through the use of a multichannel decision feedback equalizer (DFE), which adaptively combines the signals from the receive elements and mitigates channel effects. However, the computational complexity of a multichannel DFE is too high for implementation into real-time systems. One solution is to use a beamformer to reduce the number of channels the DFE needs to combine and thus reducing the overall complexity. While there has been some research showing the optimal number of beams if the channel is known, it is still an open question how to determine this information from a received signal. This research examines approaches for determining the number of beams to use so that the performance of a reduced complexity beamformer-DFE system is at least as good as a system, which uses multichannel DFE with all of the sensors as inputs. Results will be presented from experimental data comparing the performance of the systems in different ocean environments.

9:15

5aUW6. Multi-carrier synthetic aperture communication in shallow water. Taehyuk Kang, Hee Chun Song, and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238)

Orthogonal frequency division multiplexing (OFDM) communications in the presence of motion is investigated using data collected from the Kauai Acomms MURI 2008 (KAM08) experiment, conducted off the western side of Kauai, HI, in June-July 2008. The experiment involved a vertical array moored in 106-m-deep shallow water and a source towed at a speed of 3 kn at ranges between 600 m and 6 km. In order to attain reliable communications with only a single receive element, a synthetic aperture approach is applied. After combining multiple transmissions, an error-free reception is achieved with a low-density parity check code, confirming the feasibility of coherent synthetic aperture communications using OFDM.

9:30

5aUW7. Evaluation of single- and multicarrier methods for underwater communication using data-driven simulations. Menglu Xia, Daniel Rouseff (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, mengluxia@apl.washington.edu), James A. Ritcey, and Xiang Zou (Univ. of Washington, Seattle, WA 98105)

Unlike in radio communications, there is no standard statistical model for the channel impulse response in underwater acoustic communications. To compare the performance of different communications strategies requires data collected under common environmental conditions. In the present paper, a data-driven simulator is developed for comparing single- and multicarrier communications algorithms. Models for the time-varying channel impulse response over a particular bandwidth are first constructed using archival experimental data. The models are then driven with novel communications sequences of lesser bandwidth. The output is synthetic received data that can then be demodulated to evaluate communications performance. By applying reciprocity to the single-input, multiple-output archival data, both multiple receive arrays and multiple transmit arrays can be simulated. Communications performance for the synthetic data is compared to actual experimental results. [Work partially supported by the ONR.]

9:45

5aUW8. Adaptive interference suppression and cancellation for underwater acoustic communications. Steve E. Cho (Dept. of Elect. and Comp. Eng., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, scho@ucsd.edu), Heechun Song, and William S. Hodgkiss (Univ. of California, San Diego, La Jolla, CA 92093)

An adaptive, time-reversal multichannel combiner is embedded within an iterative, successive interference cancellation framework to create a reduced-complexity, multiuser receiver with multiple-access interference suppression. The combined receiver achieves the goal of separating simultaneous, full-band transmissions from physically separated users transmitting to a base station in an underwater uplink setting. Although both receiver components could feasibly act alone, the combined receiver is shown to be more robust than its individuals, achieving both temporal and spatial interference mitigations in the presence of strong multiuser interference. Furthermore, with the addition of matching pursuit, a popular sparse channel estimation technique, and a block-by-block channel adaptation, the overall receiver is designed to be applicable to time-varying underwater acoustic environments. Analysis of experimental data collected during FAF-06 illustrates that for a two-user multiple-access system, multiuser separation can be achieved even in low signal-to-noise ratio (SNR) environments.

10:00—10:15 Break

10:15

5aUW9. Long-range acoustic communication in deep water with a shallow source and horizontal towed array. Hee-Chun Song, Steve Cho, Taehyuk Kang, William S. Hodgkiss (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238), and John Preston (The Penn State Univ., State College, PA 16803)

In September 2010 a long-range acoustic communication (LRAC10) experiment was carried out in deep water off the Southern California Coast. The experiment involved two mobile components: (1) a source towed slowly at a speed of 2–3 knots at 75-m depth and (2) a horizontal line array towed at 3.5 knots at a depth of 200-m. Phase-coherent communication sequences were transmitted in the 200–300 Hz frequency band at various ranges. Initial analysis of the LRAC10 data demonstrates that an information rate of 50 bits/s can be achieved over a 550-km range using QPSK modulation and error-correction coding combined with beamforming.

10:30

5aUW10. Sparse acoustic response function estimation via Gaussian-mixture model with application to M-ary spread spectrum communications. Paul J. Gendron (53560 Hull St., SSC-Pacific, Maritime Surveillance Div. Bayside Campus, San Diego, CA 92152)

An important feature of underwater acoustic response functions is their sparsity in delay-Doppler. Models of the response must have the complexity to exploit this sparsity and the flexibility to adapt to the diverse environmental conditions that impart it. The Gaussian-mixture assignment over delay-Doppler slots can provide such a framework and leads to adaptive shrinkage operators that effectively attenuate small noise-like delay-Doppler paths and leaves larger occupied paths unaffected [Gendron, IEEE Trans. SP **53** 2005]. Accurate treatment of the latent parameters of the model aids performance across environments and extends its usefulness in underwater acoustic applications. This model is applied to M-ary orthogonal spread spectrum signaling where necessary SNR/bit at low received SNR is obtained with array, processing, and coding gain. The approach allows for bulk path dilation estimation and compensation that extends the effective channel coher-

ence duration to that associated with the natural Doppler variations between the various acoustic paths. Symbol decisions are iterative with channel estimates and bulk time varying dilation compensation. These receiver structures are tested on very shallow water acoustic transmissions in Buzzard's Bay MA. Probability of bit error is reported under 10^{-5} with two elements combining at -16 dB received SNR.

10:45

5aUW11. Underwater image transmission performance in multi-path flat fading channel. Jong R. Yoon and Jongwook Kim (Dept. of Information and Commun. Engr., Pukyong Natl. Univ., 599-1 Daeyeon-3Dong, Busan 608-737, Korea)

Time-variant sea surface scattering, Doppler spreading due to source /receiver motion, and grazing angle dependent bottom reflection in the multi-path underwater acoustic channel cause the signal fading in amplitude and phase fluctuation. Consequently, the performance of underwater acoustic communication systems is degraded, and high-speed digital communication is disrupted. In this study, two different multi-path channels such as a water tank and a river are considered and the experiments are conducted under flat fading condition. Rayleigh and Rice fading channel models are adopted and their distributions are compared with the measured in experiments. Finally, the standard Lenna image using binary frequency shift keying modulation is transmitted and the quality of the received image is analyzed based on the channel models. It is shown that the quality of the received image and the channel model is consistent with each other.

11:00

5aUW12. Computation of bit error rate for underwater communication with and without low-salinity water in the western sea of Jeju. Tae-Hoon Bok, Juho Kim, Dong-Guk Paeng, Chong Hyun Lee, Jinho Bae (Dept. of Ocean System Eng., Jeju Natl. Univ., 66 Jejudaehakno, Jeju 690-756, Rep. Korea), Ig-Chan Pang (Jeju Natl. Univ., Jeju 690-756, Rep. Korea), and Seongil Kim (Agency for Defense Development, Gyeongsangnam-do 645-016, Rep. Korea)

Salinity is one of variables in the empirical formula of the sound speed (SS) of seawater. Its effects on the SS are generally neglected because the average salinity is 34 psu and varies within a few psu seasonally and spatially in the ocean. Recently, low-salinity water around 25 psu flows into the western sea of Jeju Island in Korea due to the Yangtze River flood in China during summer. In this paper, it was analyzed how the low-salinity water affected the SS profile (SSP) and the underwater communication. The SSP was calculated with and without the low-salinity layer, and the communication channel was estimated by the simulated acoustic eigen-rays. The bit error rate (BER) was computed using binary phase-shift keying modulation, and the effects of the low-salinity water on the BER were investigated. The SSP was changed to a positive slope by the low-salinity layer at the subsurface up to 20 m of depth, forming the acoustic wave guide which mostly resulted in the decrease of the BER. Consequently, this paper suggests that it is important to consider the low-salinity water near Jeju for the underwater acoustic modeling and communication. [Work supported by UD100014DD.]

11:15

5aUW13. Communication performance analysis according to seasons in Yellow Sea. Juho Kim, Tae-Hoon Bok, Chong-Hyuen Lee, Dong-Guk Paeng, Jin-Ho Bae (Dept. of Ocean System Eng., Jeju Natl. Univ., Ara 1-dong, 102 Jejudaehakno, Jeju-si, Jeju Special Self-Governing Province, 690-756 Republic of Korea), and Seongil Kim (Agency for Defense Development, Gyeongsangnamdo, 645-016 Rep. Korea)

Communication environments vary with time in the context of underwater channel because multipaths are affected by the sound speed profile (SSP) in the ocean. In this paper, the average SSPs for 10 years (2000–2009) were calculated in the Yellow Sea so that the eigen-ray paths with the channel impulse responses were determined. The computer simulation was conducted to compare the performance of underwater communication near the surface according to the variation of season. The performance of underwater communication was analyzed using the BPSK modulation and time reversal method. The significant differences of bit error rate (BER) were shown according to the change of season. In many cases, the received signals were distorted by the multi-path channel whose features are frequency selected fading. The time reversal method was applied to compare the BER without frequency selected effect. It turns out that the good performance was shown in the autumn because of the direct path through the mixed layer channel which made the multi-path effects relatively weaker. [Work supported by UD100014DD.]

11:30

5aUW14. A new signal processing method for multipath compensation in shallow underwater acoustic communication. Da Zhi Gao, Ning Wang, and Hao Zhong Wang (College of Information Sci. and Technol., Ocean Univ. of China, 238 Song Ling Rd., Qingdao, Shandong, 266100, China, dzgao@ouc.edu.cn)

The most difficult problem in shallow underwater acoustic communications is considered to be the the time spread inherent to acoustic propagation in guide. The standard approach is to design an equalizer that attempts to correct for these propagation effects. In the presentation, a novel approach to multipath compensation in shallow water acoustic communication is proposed. The method is based on a dedispersion transformation technique, which was proposed in our previous paper [N. Wang, 9th Western Pacific Acoustic Conference, Beijing (2009)]. After dedispersion transform to a long range acoustic propagation signal, the field sampling data of a single hydrophone are transformed to several normal modes with the same waveforms. The dispersion of each modes, especially higher modes, can be removed after the transform. Then the computation cost of equalization can be decreased dramatically. The proposed approach requires less prior environmental information except an approximate beta-value of waveguide invariant. The validity of the present approach is verified in simulation.

11:45

5aUW15. The simulation research on the underwater communication system based on the vector hydrophone. Guangpu Zhang (Underwater Acoust. College, Ha'r bin Eng. Univ., Rm. No. 903, Underwater Acoust. building, No. 145 Nan tong St., guangpu_zhang@163.com)

In this paperC the vector hydrophone is applied to the OFDM underwater acoustic communication system. The directivity of vector hydrophone is used to shield the strong interference. The method has obvious effect on enhancing the anti-isotropic-interference, improving SNR, and reducing bit error rate. The result of simulation indicates that the OFDM communication system based on the vector hydrophone is the effective solution scheme for achieving the high speed underwater acoustic communication.

12:00

5aUW16. An analysis on fractional Fourier transform in Doppler and multipath mobile underwater acoustic channel. Jing Wei Yin, Sen Yang, and Xiao Zhang (Dept. of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nan Tong St., Harbin, China)

Fractional Fourier transform is a kind of time-frequency analysis method, which combines the information of signals in time domain and frequency domain. FRFT has chirp-based decomposition characteristics, for a given chirp signal, there is a fraction which gathers the energy of linear frequency-modulated signals to a maximum, which provides a basis for the detection of chirp signal with Doppler frequency offset and can also be used in the synchronization detection of mobile communication in underwater acoustic and the channel parameter estimation. This paper describes the theoretical basis of FRFT and its characteristics analysis, then by the computer simulation, performance analysis of FRFT and correlation detection in the conditions of Doppler and multipath are worked out, and the facts are verified that fractional Fourier transform is more suitable than copy correlator for Doppler frequency offset and multipath underwater acoustic channel.

12:15

5aUW17. Anti-crosstalk technology based on the adaptive estimator of instantaneous frequency. Fu Jin, Liang Guolong, Zhang Guangpu, and Wang Yan (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 150001, China)

The bandwidth of underwater acoustic channel is limited. Anti-crosstalk technology is used in many acoustic fields to increase the utilization ratio of bandwidth. Based on the principle of adaptive estimator of instantaneous frequency, a new anti-crosstalk method is presented, which uses the matched filter to resist the noise and estimate the time-delay and uses the mean square error of instantaneous frequency as detection statistics to pick out the crosstalk interference from the same or nearby acoustic channel. The simulation results show that this method is not only useful in the detection of signals in low SNR but also is effective in the suppression of channel crosstalk, and the method can adapt to Doppler mismatch very well.

FRIDAY AFTERNOON, 27 MAY 2011 GRAND BALLROOM B, 1:00 TO 2:20 P.M.

Session 5pAAa

Architectural Acoustics: Acoustics of Healthcare Spaces II

Kenneth P. Roy, Cochair
Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

Erica E. Ryherd, Cochair
Georgia Inst. of Technology, Mechanical Engineering, 771 Ferst Dr., Atlanta, GA 30332-0405

Chair's Introduction—1:00

Contributed Papers

1:05

5pAAa1. Room acoustics simulation of hospital patient care-unit. Sentagi S. Utami and Mojtaba Navvab (Dept. of Architecture, Univ. of Michigan, 2000 Bonisteel Boulevard, Ann Arbor, MI 48105, sentagi@umich.edu)

Acoustical design challenges in hospitals are providing better communication to reduce medical errors and speech privacy that has also become a legal issue according to the Health Information Portability and Accountability Act (HIPAA). Some industrial products performance criteria are being exaggerated in the claims as solutions for better room acoustics of healthcare spaces. This study demonstrates an acoustical quality improvement of a patient care-unit environment. Building regulation codes, health service activities, space openness, and confined space layout are used as parametric studies for effective room acoustics design solutions. Current techniques in computer simulation are utilized along with field measurements to localize sources and identify critical elements that create the most impact. Acoustical treatments with characteristics adopting the industrial products are applied within the architectural elements including adjustable hung curtains, furniture, and art displays on wall surfaces. The improvement results are subjectively evaluated within VR environment of selected scenes to obtain real time spatial experience given the auralization capabilities. These results show the detailed comparison of each design variable with respect to their ability to fulfill the speech communication requirements. The study shows the limits and possibilities for acoustic design improvement of patient care-units at various stages of design.

1:20

5pAAa2. The acoustic environment of inpatient hospital wards. N. J. Shiers, B. M. Shield, and R. C. Glanville (Faculty of Eng., Sci. and Built Environment, London South Bank Univ., London SE1 0AA, United Kingdom, shiersn@lsbu.ac.uk)

There is an increasing body of research into the acoustic environment of hospitals, which provides evidence of the detrimental effects of noise on the well being and comfort of patients and on staff, and of a significant rise in hospital noise levels in recent years. Much of this evidence has focused on specific areas of healthcare such as critical care and operating theatres, with comparatively few studies carried out within general inpatient wards and in UK hospitals. A project is currently being undertaken to investigate, through objective and subjective surveys, the noise climate and acoustic design within general inpatient facilities in the United Kingdom, and their influence on the acoustic comfort of patients and staff. Noise and acoustic surveys have been carried out in a range of inpatient wards, with corresponding questionnaire surveys of staff and patients. Three major UK hospitals have been involved in the study, and a substantial amount of data has been collected. Reported results will include comparisons of noise levels measured in a number of ward locations including multibed and singlebed rooms, identification of the sources of high level noise, and some analysis of patient and staff attitudes to noise.

1:35

5pAAa3. Application of constrained-layer damping to improve speech privacy in healthcare facilities. Benjamin M. Shafer and Brandon Tinianov (Serious Mater., 1250 Elko Dr., Sunnyvale, CA 94085, bshafer@seriousmaterials.com)

Contemporary wall assemblies for demising partitions in commercial buildings, such as hospitals, are commonly designed using heavy-gauge (versus light-gauge) steel framing components and gypsum drywall. Recent studies indicate that these types of partitions do not provide adequate speech privacy between rooms [A. Bétit, Sound & Vibration, 14–16 (2010)]. Former testing has proven that constrained-layer damping technology may be used to achieve much higher transmission loss values in the speech range than gypsum drywall. Measurements were made to explore the use of constrained-layer damping technology as a single-layer solution on heavy-gauge steel framing versus multiple layers of gypsum drywall, specifically for speech privacy. Theoretical results are also presented and discussed.

1:50

5pAAa4. Tools to predict binaural speech intelligibility in complex listening environments for normal and hearing-impaired listeners. Birger Kollmeier (Medizinische Physik, Universitt Oldenburg and Fraunhofer IDMT Hearing, Speech and Audio Technol., D-26111 Oldenburg, Germany, birger.kollmeier@uni-oldenburg.de), Jan Rennies (Fraunhofer IDMT Hearing, Speech and Audio Technol., D-26129 Oldenburg, Germany), and Thomas Brand (Universitt Oldenburg, D-26111 Oldenburg Germany)

Appropriate models for speech intelligibility in acoustically "difficult" situations (i.e., background noise from several interfering sound sources and reverberation) are an important tool not only for research, but also to characterize the intelligibility achieved in special-purpose rooms such as healthcare spaces. This contribution presents an overview on existing and new models that have mostly been implemented in the "speech intelligibility prediction toolbox" in order to allow for a comparison of model performance and to assess the expected speech intelligibility for a predefined situation as a function of individual hearing loss. The extention of a binaural speech intelligibility model with room acoustical parameters is introduced, which quantitatively predicts the combined influence of reverberation and different positions of target speaker and interferers both in quiet and in noise. The accuracy in prediction for individuals can be increased by including the respective individual audiogram. Taken together, a comparatively exact prediction of speech intelligibility can be done in comparatively "easy" acoustical situations for listeners with a mild-to-moderate hearing loss. The current models still show room for improvement as soon as more complex listening situations, hearing loss types, and more detailed properties of the speech reception process are considered.

2:05

5pAAa5. On the acoustics of operating rooms. Seth M. Harrison and Nicholas R. Sacco (Acoust. and Vib. Group, KJWW Eng. Consultants, 623 26th Ave, Rock Island, IL 61201, harrisonsm@kjww.com)

Designing modern operating rooms to me*et al.*l design criteria can be a challenge. Creating a sterile, well-lit, temperature and humidity appropriate environment for medical operations typically takes precedence over the acoustics of the operating room. The sterile, washable operating room surfaces are commonly hard and reflective, offering little sound absorption. These surface finishes result in strong discrete reflections, increased background noise, and reduced speech intelligibility. Code mandated energy efficiency and ventilation requirements coupled with the owner-driven desire for economical mechanical systems can result in excessive HVAC noise, adding to background noise levels. Yet, speech intelligibility and speech privacy levels are expected to be superb to minimize errors due to miscommunication and to prevent the unintentional spread of confidential patient information. The acoustical challenge is to design operating rooms to control noise and to enhance room acoustics without affecting the performance of other systems. Speech intelligibility and impulse response measurements from modern hospital operating rooms and design strategies will be discussed.

FRIDAY AFTERNOON, 27 MAY 2011 GRAND BALLROOM B, 2:30 TO 5:10 P.M.

Session 5pAAb

Architectural Acoustics and Noise: Speech Privacy

Eric L. Reuter, Chair
Reuter Associates, LLC, P.O. Box 4623, Portsmouth, NH 03802-4623

Chair's Introduction—2:30

Invited Papers

2:35

5pAAb1. Harmonizing national & international speech privacy guidelines. David M. Sykes (Remington Partners, 31 Baker Farms Rd., Lincoln, MA 01773, david.sykes@remington-partners.com)

Absent leadership from the U.S. federal government, the process of developing (and continuously improving) consistent speech privacy guidelines for healthcare facilities, is like herding cats. Nevertheless, progress continues. 1 year after the release of the FGI 2010 Guidelines (and 15 years after Congress passed HIPAA) the first code-level U.S. guidelines for speech privacy, noise, and vibration in healthcare facilities were released, and several leading groups have agreed to cite the 2010 Guidelines as their sole "reference standard." These groups are as follows: the Department of Health and Human Services (HIPAA), the Veterans Administration, the Centers for Disease Control, 38 regional authorities having jurisdiction (AHJ's), the U.S. Green Building Council's LEED for Health Care, the Green Guide for Health Care, 48 state code authorities, and most recently, the International Green Code Council. What is left? the Joint Commission, the Department of Defense, the World Health Organization, and others. Meantime, work on the 2014 edition of the Guidelines is already underway with important updates to the speech privacy section. Since the FGI Guidelines are actively used in 16 countries, it is imperative to involve the global acoustical community in the research and development needed to strengthen them. What is the best way forward?

2:55

5pAAb2. Acoustical modeling software as a tool to evaluate speech privacy in open plan office environments. Jessica E. Newton and Justin E. Meyer (Thorburn Assoc., Inc., 2500 Gateway Ctr., Blvd Ste. 800, Morrisville, NC 27560)

A productive work environment requires enough privacy to conduct confidential business on the telephone while maintaining background noise levels that allow others to concentrate on their work. However, many businesses also rely on the collaborative work environment that can be created in an open plan office. This study concentrates on using EASE computer models as a tool to demonstrate to clients and end-users what type of acoustic environment to expect when selecting different types of workstations. The degree of speech privacy afforded was tested by determining the opposite effect: the level of speech intelligibility between workstations. A typical open office layout was modeled using EASE acoustical modeling software. The speech intelligibility between workstations was modeled in several potential configurations to demonstrate the effects of the changes in speech privacy within the office. The configurations tested include varying the heights of workstation partitions and using either reflective or absorptive partition finishes.

3:15

5pAAb3. Methods of flanking path control in party walls. John LoVerde and Wayland Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

It is generally appreciated in the acoustical community that flanking paths can control the overall noise isolation, especially for high-STC wall designs [Nightingale, *et al.*, NRC-IRC, Flanking Transmission in Framed Buildings, Phases I, II, and III (1997–2010)]. A recent condominium retrofit into an existing structure produced complaints of very low privacy between some units, in spite of a double-stud wall design that would typically provide acceptable isolation. Airborne noise reduction testing revealed a variety of unique flanking paths due to the constraints of the existing structure and the intersection of disparate materials (the construction includes wood, concrete, steel, and masonry components). A number of methods were implemented, and additional testing performed to quantify the flanking paths and the effectiveness of the mitigation. The methods and results are presented.

3:35

5pAAb4. Influence of partition height and sound masking on speech privacy and occupant acceptance in open plan offices: Results of surveys and field tests. Steven D. Pettyjohn (The Acoust. & Vib. Group, Inc., 5700 Broadway, Sacramento, CA 95820, spettyjohn@acousticsandvibration.com)

This paper combines the results of two studies done to provide speech privacy and occupant acceptance of open plan office furniture and sound masking. In the first study, the occupants of open plan offices in six buildings were surveyed to learn what they thought of acoustical conditions. Acoustics was not the only area addressed as temperature was another topic. Statistical evaluation of the results showed that acceptance of their cubicle design was directly proportional to the height of the panels. F-tests did not show a correlation with any other factor such as the presence of sound masking systems. In the second project, the influence of a sound masking system on speech privacy was evaluated. A decision was made to install low height cubicle panels to permit maximum interaction in a call center. Sound measurements were made in spaces with and without sound masking systems and informal surveys done to understand occupant satisfaction and the influence of sound masking on speech privacy and the ability of occupants to concentrate on their work. A summary of the survey and analysis is provided along with the results from sound tests in spaces with and without sound masking and with various cubicle panel heights.

3:55

5pAAb5. Case study: Innovative techniques for improving speech privacy in open plan offices. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Sq., Vernon, CT 06066, bbrooks@brooksacoustics.com)

An acoustical engineering study was conducted to develop renovations to improve the speech privacy in an open plan office. The existing open plan configuration was to be maintained in this renovation, constraining the design effort. Therefore, several innovative designs were added to more traditional techniques. Articulation index (AI) test data, measured both before and after the renovation installation, were used to assess the improvements in speech privacy between several areas in the subject office space. The renovations included the installation of absorbent panels on the ceiling and walls, partition toppers for the conference rooms, aerogel banners, door seals, and a sound masking system. Measured speech privacy was significantly increased in the renovated office areas.

4:15

5pAAb6. Case study: Patient admissions/waiting area speech privacy. Gregory C. Tocci and John T. Foulkes (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, gtocci@cavtocci.com)

Patients forming a queue at a transaction counter are being overheard by persons next in line in the queue. The compromised speech privacy, besides being a potential violation of HIPAA and HITECH regulations, may lead to patients not fully disclosing conditions they may have, fearing they might be overheard by others in the queue. This might led to important details of medical conditions not being addressed during subsequent medical treatment. This case history summarizes the applicable regulations, design objectives for speech privacy, and the methods for improving speech privacy including cordoning the queue away from the transaction desk, sound absorption, barriers, and the use of electronic sound masking. The presentation describes an existing medical reception area. Speech privacy improvements are currently being installed. Privacy indices were computed using EASE. This paper will discuss the benefits of the speech privacy improvements studied, particularly the use of electronic sound masking which was both required to achieve the design objective and the technique that provided the greatest improvement in speech privacy.

4:35

5pAAb7. Case study: Medical office building designed without regard for acoustical privacy. Eric L. Reuter (Reuter Assoc., LLC, P.O. Box 4623, Portsmouth, NH 03802, ereuter@reuterassociates.com)

A newly constructed medical office building was designed (without the aid of an acoustical consultant) such that registration, waiting, and check-out areas are closely spaced and lack architectural separation, making private verbal exchange of information impractical. A survey of patients conducted by the owner shortly after the building opened indicated that lack of oral privacy is the leading criticism of the facility. Measurements of existing privacy Index between several talker/listener position pairs demonstrate extreme deficiency relative to criteria provided by the Facilities Guidelines Institute, Green Guide for Health Care, and others. Recon-

figuration would be prohibitively expensive, leaving electronic sound masking as the only practical option for improving privacy. However, the masking level that would be required to achieve adequate privacy would exceed that recommended for occupant comfort. The owner is faced with balancing the risk of litigation for breach of privacy law with the cost associated with upgrading the facility for compliance. Measured data, calculations, and an examination of the risks and benefits of various upgrade options will be presented.

Contributed Paper

4:55

5pAAb8. Case study: Improving speech privacy in a cathedral ceiling open office. Timothy Foulkes (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA) and William Elliott (Cavanaugh Tocci Assoc., Sudbury MA)

A large indoor food market with a wood cathedral ceiling was adapted for use as an office space. A mezzanine level was added as part of the office fitup. The space was entirely open plan including the conference rooms, which had glass screen walls but no ceilings. The acoustic problems included excess loudness, lack of privacy, difficulty with telephone communications, and poor room acoustics in the conference room. This was a Green project, which put an emphasis on natural lighting and ventilation, and organic materials. Careful consideration of available products on the market led to some creative recommendations for acoustic upgrades.

FRIDAY AFTERNOON, 27 MAY 2011 — ISSAQUAH, 1:00 TO 2:30 P.M.

Session 5pAB

Animal Bioacoustics: General Topics in Passive Acoustic Monitoring of Animals II

Manuel Castellote, Chair
NOAA, Alaska Fisheries Science Center, Seattle, WA 98115

Contributed Papers

1:00

5pAB1. Whistle variation is associated with surface behavior in the Guyana dolphin (Sotalia guanensis). L. J. May-Collado (Dept. of Biology, Univ. of Puerto Rico, San Juan, PR 00931, lmaycollado@gmail.com)

Guyana dolphins emit whistles with the widest frequency range ever reported today (1.34 to 48.4 kHz). Distance between populations has been suggested as an important factor promoting whistle variation in this dolphin species. However, other factors known to promote intra-specific whistle variation among delphinids such as behavior have not been explored yet in Guyana dolphins. Behavior has been shown as important factor promoting whistle variation in many whistling species, and thus it should be taken into consideration in comparative studies of dolphin communication. Here, I study the relationship between whistle structure and surface behavioral states (travel, foraging, and social behaviors) of Guyana dolphins from a small resident population in Costa Rica. Behavior had a significant effect on whistle duration and whistle minimum, delta, ending, and start frequencies variables. Moreover, more than 47% of the whistles were correctly classified to their corresponding surface behavior. These results indicate that not only behavior is an important factor contributing to intra-specific whistle variation in Guyana dolphins but that behavior is an important variable to describe any acoustic study which goal is to compare populations.

1:15

5pAB2. Factors determining whistle emission rate in bottlenose dolphins of Bocas del Toro, Panama. S. G. Quiñones-Lebrón and L. J. May-Collado (Dept. of Biology, Univ. of Puerto Rico, San Juan, PR 00931, lmaycollado@gmail.com)

Dolphins of Bocas del Toro are known to change their communication signals when interacting with dolphin-watching boats. In this study we examine the role of engine noise, calf presence, and behavior on the emission rate of whistles. Whistle emission rate was measured in groups where calves were and were not present and while interacting and not interacting with boats. A general higher whistle emission rate was shown when only the research boat was present and during social events, although whistle contours were not significantly different. Whistle rate was also slightly higher in groups with calves, particularly during social events. While in the presence of other boats, whistle emissions were higher during traveling events. A low whistle emission in the presence of other boats than the research boat (largely dolphin-watching boats) may be the result of masking or a response to the engine noise during these encounters. In addition, groups with calves may be avoiding intrusive dolphin-watching boats. Most groups with calves, for which whistle emission was higher during social events, were observed during encounters with the research boat (engine off). These results provide insights on some of the environmental and biological factors that may be influencing dolphin whistle emission.

1:30

5pAB3. Training diving ducks for behavioral audiograms. Sara C. Therrien, Catherine E. Carr, Robert J. Dooling, Arthur N. Popper (Dept. of Biology, Univ. of Maryland, College Park, MD 20742, therrien@umd.edu), Ronald E. Therrien, and Alicia M. Wells-Berlin (USGS Patuxent Wildlife Res. Ctr., Laurel, MD 20708)

Lesser scaup (*Aythya affinis*) are a species of diving duck that dive to depths of greater than 20 m to forage on crustaceans, mollusks, and fish. Currently, there are no measures of underwater hearing of any diving bird because of the inherent difficulties of training a bird to respond to sound underwater. Lesser scaup in a captive colony at USGS Patuxent Wildlife Research Center in Laurel, MD are being trained to participate in in-air and underwater behavioral audiograms. Ducklings were hand-reared to respond to trainers, auditory signals, and mealworm rewards. The ducks were then trained on a go/no-go task to respond to varying frequencies and intensity levels by pecking an LED-lit target. All targets and acquisition devices were designed to follow similar procedures in-air and underwater. Previous auditory brainstem response (ABR) tests demonstrated an in-air maximum sensitivity at 2–3 kHz. These behavioral audiograms will provide a measure to compare ABR and psychoacoustic thresholds as well as a measure of underwater thresholds, which would be difficult to implement using only the ABR.

1:45

5pAB4. Equine vocalization: A comparison of Arabian and Morgan horse whinnies. David G. Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter D. Herstein (Charlestown, RI 02813)

Horses tend to be more vocal when in a barn in order to compensate for reduced visibility, restricted mobility, and close access by intruders. This can express itself in such ways as a vocal greeting to the arrival of a feed cart. Under these conditions it has established that the spectral content of whinnies can vary between specific behavioral situations, although we are still a long way from determining what this implies. Here we compare whinnies

for two different horse breeds (at two different locations), Arabians (Michigan State University) and Morgans (University of Connecticut) under similar situations. Arabians appear to have a wider range of frequency variation at times even approaching the rapid sweep of their fellow artiodactyls, the tapirs. On the other hand, Morgans can exhibit a stronger tremolo, which many associate with whinnies. In general there does appear to be a significant difference between the two horse breeds in their vocal responses.

2:00

5pAB5. Some reflections on chorusing. Michael S. Stocker (Ocean Conservation Res., P.O. Box 559 Lagunitas, CA 94938, mstocker@ocr.org)

Sound and song production in many animals is sexually dimorphic inasmuch as the males of most species either exclusively produce sounds, or produce the more complex sounds of the species. As a consequence, chorusing is commonly framed under the rubric of competitive strategies of breeding fitness advertisement for individual animals while ambiguating the individual sources of sound from predators within the sound of the group. Chorusing can be either synchronous (common in many stridulating insects) or asynchronous (common in anurans and fish). Other strategies may also be present in chorusing, including collective annunciation of the group fitness and physical extents, and identification of external threats to the chorusing "acoustic community." This paper explores some other aspects of "chorusing" that includes other group-acoustic behaviors such as fish schooling and synchronous bird flocking behaviors.

2:15

5pAB6. Audibility of the cricket, acheta domesticus, obtained from auditory evoked potentials. Bailee Guisti, Shanna White, and Al Yonovitz (Dept. of Comm Sci. and Disord., Univ. of Montana, bailee.guisti@umconnect.umt.edu)

For crickets, auditory and vibratory communication is important in reproductive behavior, agonistic interactions, detection of predators, and acoustic orientation in the environment. In the cricket, the forelegs house the specialized tibial organs that comprise the tympanal structures. Much of the information regarding the auditory sensitivity of the receptors of the foreleg is the result of study of individual receptor cells. The whole receptor population of the tympanal organ covers the frequency range from at least 2000 k–70 kHz. Little information is available on the neuronal and physiological properties of the brain to acoustic signals. The purpose of this study was to record auditory evoked potentials on the surface of the cricket brain, Acheta domesticus. Electrodes were placed onto the brain of crickets and signals were amplified and averaged over the range 100 ms. Sinusoidal stimuli covered a range between 4 and 48 kHz. Waveform morphology and input-output curves for the cricket will be shown.

FRIDAY AFTERNOON, 27 MAY 2011 GRAND BALLROOM A, 1:00 TO 3:10 P.M.

Session 5pBA

Biomedical Acoustics and Physical Acoustics: Photons and Phonons: Diagnostic and Therapeutic Applications II

Parag V. Chitnis, Cochair
Riverside Research Inst., F. L. Lizzi Center for Biomedical Engineering, 156 William St., New York, NY 10038

Ronald Silverman, Cochair
Riverside Research Inst., F. L. Lizzi Center for Biomedical Engineering, 156 William St., New York, NY 10038

Invited Papers

1:00

5pBA1. Nanoparticle-mediated ultrasound-guided photoacoustics in imaging and therapy of cancer. Stanislav Emelianov (Dept. of Biomedical Eng., The Univ. of Texas at Austin, 1 University Station C0800, Austin, TX 78712, emelian@mail.utexas.edu)

A morphological, functional, and molecular imaging technique capable of visualizing biochemical, pharmacological, and other processes *in vivo* and repetitively during various stages of tumor progression and cancer treatment is desired for many biomedical, preclinical, and clinical applications. Over the past several years, we have been developing an integrated imaging approach—ultrasound-guided photoacoustic imaging. Ultrasound is used to visualize anatomical structures and photoacoustics is used to provide functional information about tissue using spectral differences between absorption of oxygenated and deoxygenated hemoglobin and cellular activity using molecularly targeted plasmonic nanosensors. In this paper, the ability of ultrasound and nanoparticle-mediated photoacoustic imaging to simultaneously obtain the anatomical and molecular map of tumor *in vivo* will be demonstrated. Furthermore, the role of ultrasound and photoacoustic imaging in therapy of cancer will be discussed. Finally, the development and future directions in ultrasound-guided photoacoustics and contrast agents will be discussed.

1:20

5pBA2. Gold-nanoparticles for optoacoustic imaging and therapy. M. Frenz, M. Kitz, S. Preisser, M. Jaeger (Inst. of Appl. Phys., Univ. of Bern, Sidlerstr. 5, CH-3012 Bern, Switzerland, frenz@iap.unibe.ch), A. Wetterwald, and G. N. Thalmann (Univ. of Bern, CH-3010 Bern, Switzerland)

Gold nanoparticles have shown promise as contrast agent in optoacoustic imaging and as selective absorbers in phototherapy because of the strong surface plasmon resonance absorption at specific wavelengths. Using pressure transient detection, flash photography, two-photon luminescence microscopy, and TEM imaging, vapor bubbles generated upon irradiation of single spherical gold nanoparticles, single nanoparticles in suspension, and of nanoparticles targeted cells were investigated. Formation thresholds determined at different wavelengths indicate a bubble formation efficiency increasing with the irradiation wavelength. A value of 60 ± 5 mJ/cm^2 was found for 532 nm, which is close to the wavelength of the surface plasmon resonance. The minimum threshold for acute cell damage was determined to be 54 ± 13 mJ/cm^2 for a single laser pulse generating a bubble diameter of 7.6 ± 1 μm corresponding to about half of the cell diameter and 23 ± 1 mJ/cm^2 for pulse train irradiation. These values have to be compared to the maximum permissible exposure limits for skin. Our results show that the particle distribution inside the cell plays a key role for the cell damage process, which strongly influences the pathways of cell targeting. A good understanding of the cell death mechanisms is a prerequisite to translate experimental findings to the clinic. [This research was supported in part by the SNF (No. 205320-116343) and the EU (No. LSHC-CT-2006-018858 PROMET.)]

1:40

5pBA3. Enhancing targeted focused ultrasound therapy using light, sound, and nanoparticles. Ronald A. Roy, James R. McLaughlan (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215), and Todd W. Murray (Univ. of Colorado at Boulder, 427 UCB, Boulder, CO 80309)

The use of targeted nanoparticles for both imaging and therapeutic applications shows significant promise. Photoacoustic tomography is a non-invasive imaging technique based on the detection of broad-band acoustic emissions generated by the absorption of laser light in tissue. The introduction of light-absorbing gold nanoparticles can improve signal emission levels and, if functionalized, can promote the targeting of specific cell populations, thereby enhancing both contrast and the ability to delineate tissue types. For a sufficiently high laser fluence, a transient vapor cavity is formed which collapses inertially, generating a broadband emission and even greater contrast enhancement. However, the fluence required typically exceeds the maximum permissible exposure for tissue. By combining ultrasonic and optical pulses, the light and sound thresholds required to repeatedly generate inertial cavitation (IC) can be significantly reduced, thus reducing both the laser fluence and acoustic pressures required to generate images from these acoustic emissions. The presence of IC in continuous wave high intensity focused ultrasound (HIFU) exposures can locally enhance the heating resulting in effective tissue ablation at lower HIFU intensities. [Work supported by a Boston University COE Dean's Catalyst Award and the Gordon Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

2:00

5pBA4. Combining light and sound to guide and monitor ultrasound therapy. Emmanuel Bossy (Institut Langevin, ESPCI ParisTech UMR 7587, INSERM ERL U979, 10 rue Vauquelin, 75231 Paris Cedex 05, France, emmanuel.bossy@espci.fr)

Ultrasound therapy makes use of high-intensity focused ultrasound (HIFU) to noninvasively necrose targeted soft tissues without damage to overlaying or surrounding tissues. Two main challenges in implementing HIFU therapies are treatment guidance and treatment monitoring. In this paper, we discuss the use of techniques combining optics and acoustics to guide/monitor HIFU, as alternatives or complements to more conventional approaches based on magnetic resonance imaging or ultrasound imaging. Recently, several publications demonstrated some potential for photoacoustic imaging in the field of HIFU therapy, both for guidance and/or monitoring [Funke *et al.*, Apple Phys. Lett. **94**(5), (2009); Chitnis *et al.*, JBO **15**(2), (2010); Cui *et al.*, JBO **15**(2), (2010)]. In the first part of the presentation, we will illustrate the use of a dual-mode linear array for photoacoustic-assisted ultrasound therapy. The same elements of the probe are used both as photoacoustic detectors and HIFU emitters, driven by a dedicated multichannel electronics. In the second part of the presentation, monitoring the formation of a HIFU lesion will be demonstrated using transient optoelastography, a technique based on the optical detection of shear motion generated by the acoustic radiation force.

2:20

5pBA5. Photoacoustic imaging of ultrasound-induced displacements in ocular tissues. Ronald Silverman, Raksha Urs, Harriet Lloyd (Dept. of Ophthalmology, Columbia Univ. Med. Ctr., 160 Ft. Washington Ave., New York, NY 10032, rs3072@columbia.edu), Jeffrey Ketterling (Riverside Res. Inst., New York, NY 10038), Fanting Kong, and Y.C. Chen (Hunter College, New York, NY 10065)

Acoustic radiation force produces compression that can be used for assessment of tissue stiffness, generally by detection of displacements in the phase-resolved pulse/echo ultrasound waveform. In this report, we describe use of photoacoustics for tracking displacements produced by acoustic radiation force in the iris and retina. The probe consisted of a 20 MHz transducer with a central aperture through which 532 nm laser pulses were introduced. Laser pulses were emitted simultaneously with excitation of the transducer by a 20 MHz monocycle. For a 12 ms period, we interleaved tonebursts (generating force) between successive laser pulses/monocycles. Iris displacements averaged 26.5 and 76.7 μm photoacoustically versus 24.9 and 71.3 μm by pulse/echo at 60 and 100 W cm^{-2} force, respectively. For the retina, the displacements were 14.6 and 25.3 μm photoacoustically versus 12.4 and 25.8 by pulse/echo. Pulse/echo displacements were more difficult to discern due to their broadness and anatomic non-specificity. The photoacoustic signal is advantageous because of its broadband character and because the photoacoustic signal is only generated by specific molecules, which in the retina would correspond to melanin in the pigment epithelium. As little is known about retinal elasticity, this technique offers an avenue toward investigation of this tissue property.

Contributed Papers

2:40

5pBA6. Real-time monitoring of high-intensity focused ultrasound ablations with photoacoustic technique. Huizhong Cui and Xinmai Yang (Univ. of Kansas, 5109 Learned Hall, 1530 W 15th St., Lawrence, KS 66045)

High-intensity focused ultrasound (HIFU) has been widely used in clinics as a non-invasive technique for soft tissue ablation. Real-time monitoring of HIFU process is required in order to perform HIFU. A photoacoustic (PA) imaging system was used to monitor HIFU ablation process in this study. Single-element, spherically focused ultrasonic transducers with center frequencies of 5 and 10 MHz were used to generate a HIFU field and detect the PA signals in beef kidney during HIFU treatments, respectively. A 25-micro-diameter thermocouple was used to measure the temperature rise during the treatment as well. Thermal dose, which was used to indicate the coagulation of soft tissue, was calculated with the temperature measurement from the thermocouple. Detected PA signals were therefore related to the coagulation of soft tissue during HIFU through thermal dose calculations. In addition, PA signals from beef kidney coagulating under a constant temperature was obtained to show the changes of PA signals under a constant temperature during soft tissue coagulation. We demonstrate that thermal dose is a more appropriate way to monitor in the treatment process instead of monitoring temperature, since the monitoring of temperature change through the detected PA signals during HIFU ablation may not be feasible.

2:55

5pBA7. Acousto-optic sensitivity mapping of biological tissues under insonification by high-intensity focused ultrasound. Samuel Powell and Terence S. Leung (Dept. of Medical Phys. and Bioengineering, Univ. College London, Malet Pl. Eng. Bldg., London WC1E 6BT, United Kingdom, spowell@medphys.ucl.ac.uk)

By means of the acousto-optic effect, a focused acoustic field may perturb the optical paths within a particular region of a highly scattering medium. These perturbations cause the modulation of a speckle pattern formed when multiple optical paths from a coherent source interfere at a detector position. Through analysis of this modulation, the contrast of biological tissues at near-infrared wavelengths can be exploited to determine the optical properties of biological tissue with a spatial resolution comparable to the dimension of the acoustic focus. It has been demonstrated *ex vivo* that this technique is capable of monitoring changes in the optical properties of biological tissue in the acoustic focus region during treatments using high-intensity focused ultrasound. The sensitivity of this approach to the optical properties of regions external to the focus has not yet been examined; this is a key question in determining the clinical viability of this approach. This work presents modulation depth sensitivity maps in various clinically relevant geometries with varying background and target optical properties and high intensity, non-linear acoustic field distributions. The sensitivity maps are generated using a novel Monte-Carlo simulation code which is accelerated by highly parallel execution on graphics processing units.

FRIDAY AFTERNOON, 27 MAY 2011 CIRRUS, 1:00 TO 4:30 P.M.

Session 5pEA

Engineering Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Animal Bioacoustics: Acoustical Sensor and Array Technology II

Dehua Huang, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841-1708

Thomas R. Howarth, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841-1708

Contributed Papers

1:00

5pEA1. Wave space array methods for aeroacoustic testing. Chris Bahr, Mark Sheplak, Lou Cattafesta (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL 32611-6250, cattafes@ufl.edu), and Jian Li (Univ. of Florida, Gainesville, FL 32611-6200)

Phased array beamforming is commonly applied to assess the spatial distribution of noise sources in aeroacoustic wind tunnel testing, generally with planar microphone arrays. As classically applied with a conventional beamforming algorithm, results can suffer contamination due to cross-flow over the array face in the wind tunnel test environment. When the cross-flow fluctuations are statistically independent from microphone-to-microphone, diagonal removal can be applied to the cross-spectral matrix to negate this contamination. However, when these fluctuations are correlated, diagonal removal fails to recover the acoustic field of interest. As has been previously demonstrated, the wave space representation of an observed pressure field readily separates cross-flow hydrodynamic fluctuations from the observed acoustic field, when applied to low speed wind tunnel tests. The authors build on existing research to construct beamforming methods in wave space to leverage this separability. The implications of aeroacoustic beam-forming analysis in wave space as applied to plane and spherical waves are discussed, with focus on filtering hydrodynamic fluctuations from acoustic waves, as well as determining levels of the resultant filtered pressure field. Theoretical development is presented, followed by numerical analysis and experimental results from the University of Florida Aeroacoustic Flow Facility.

1:15

5pEA2. Time/frequency multiple signal classification beamforming based on principle component analysis for reducing platform and flow noise and identifying continuous and impulsive ground targets on UAV. Ramon A. Silva and Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843, rsilva06@tamu.edu)

When a microphone array is mounted on an UAV, most existing beamforming methods cannot be used to adequately identify continuous and impulsive targets on the ground due to high-level platform and flow noise. Here, we propose to develop a time/frequency beamforming method based on a principle component analysis and multiple signal classification (MUSIC) algorithm. This method can reduce the effects of platform and flow noise, e.g., engine and boundary layer noise, by removing noise-associated principle components from measured signals. It can also pinpoint the exact target locations while most existing methods can only detect target directions. In order to validate the proposed method, a cross-shaped microphone array is installed on the bottom surface of an UAV. The UAV is then placed in a wind tunnel operating at Mach 0.05–0.1 and its engine is turned on to simulate flight cruising conditions. Two loudspeakers are used to simulate continuous and impulsive ground targets. The target locations estimated from the proposed method are compared to the actual loudspeaker locations. Through the wind tunnel experiment, it is shown that the proposed beamforming method can be used to effectively suppress platform and flow noise and successfully identify the transient targets.

1:30

5pEA3. Micromachined reconfigurable microphone array for wind tunnel testing. Joshua S. Krause, Alfram V. Bright (Dept. of Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, joshua.krause@tufts.edu), Mark J. Moeller, Judith M. Gallman (Spirit AeroSystems, Inc., Wichita, KS 67278), and Robert D. White (Tufts Univ., Medford, MA 02155)

A surface micromachined, front-vented, 64 channel (8×8), capacitively sensed microphone array-on-a-chip devices for aeroacoustic testing is described. The arrays are fabricated using the MEMSCAP PolyMUMPs polysilicon surface micromachining process, with a Parylene-C passivation layer. The devices are packaged with low profile interconnects, presenting a maximum of 100 μm of surface topology. The array electronics allow the microphone outputs to be redirected to one of two channels, allowing dynamic reconfiguration of the effective transducer shape in software. Measured microphone sensitivity is 0.15 mV/Pa for an individual microphone and 8.7 mV/Pa for the entire array, in close agreement with model predictions. The microphones and electronics operate over the 200–40 000 Hz band. The dynamic range extends from 60 dB SPL in a 1 Hz band to greater than 150 dB SPL. Element variability is ±0.05 mV/Pa in sensitivity with an array yield of 95%. Off-chip electronics provide 80 dB off isolation. Preliminary wind tunnel testing at flow rates of up to 23 m/s indicates that the devices continue to operate in flow without damage, and can be successfully reconfigured on the fly. Analysis of measured boundary layer pressure spectra at six flow rates from 5 to 23 m/s is underway.

1:45

5pEA4. Generalized inverse beam-forming in applications to aeroacoustic problems. Takao Suzuki (Acoust. and Fluid Mech., The Boeing Co., Seattle, WA 98124, takao.suzuki@boeing.com)

Current source-detection techniques (i.e., beam-forming) still suffer with resolving aerodynamic sources, which are typically distributed, highly directive, and partially coherent. Improvement of these tools helps us evaluate new acoustic technologies. We are developing an algorithm, called L1 generalized inverse beam-forming, to resolve coherent/incoherent, distributed, and multipole sources. To extract each coherent signal, a cross spectral matrix is decomposed into eigenmodes. Subsequently, the complex source-amplitude distribution that recovers each eigenmode is solved using L1-norm generalized inverse techniques using iteratively re-weighted least squares with reference solutions including multipoles as well as a monopole. The capabilities of the proposed algorithm are demonstrated using benchmark problems by comparing with several existing beam-forming algorithms, and it is found that distributed sources as well as dipoles with arbitrary orientation can be identified regardless of coherency with another source. The resolution is comparable to existing deconvolution techniques, such as DAMAS or CLEAN, and the computational cost is only several times more than that of DAMAS2. The proposed algorithm has also been applied to aeroacoustic test data including noise associated with jet-flap interaction, noise from a single round jet, and duct acoustics with flow, and these results are discussed in this talk.

2:00

5pEA5. Direct algebraic localization of a proximate source from instantaneous or short-time Fourier transform measurements on circular array. Shigeru Ando (Dept. of Information Phys. and Computing, Univ. of Tokyo, Tokyo 113-8656, Japan, ando@alab.t.u-tokyo.ac.jp)

The purpose of this study is to obtain an inversion formula of direction and distance of a proximate source from sound pressure distribution on a circular array. The approach will be important both in theoretical aspects and in practical usage for providing rapid initial estimates in more general multi-source conditions. Our approach is based on the weighted integral method (WIM) [IEEE Trans. SP **57**, (2009), Inverse Problems **26**, 015011 (2010)] for signal/source parameter estimation. The WIM is composed of two steps: (1) describing the problem with a partial differential equation (PDE) and a measurement scheme in a finite region and (2) integrating the PDE on the measurement region with appropriate weight functions to obtain equations for solving the parameters. The integration by parts eliminates the differential-of-field terms in the PDE. The weight functions are chosen so that the differentials generated instead by the integration have simpler forms and also the integral boundary terms at edges of the region will vanish successfully. We begin with the location-constraint PDE [ASA/ASJ meeting (2006)] and obtain exact formulas, an instantaneous one including time-differentials and an STFT-based one with frequency decomposition, for a proximate source. We show several experimental results using 16-element circular microphone array.

2:15

5pEA6. Digital beamsteering system using acoustic transducer array. Chong Hyun Lee, Jinho Bae, Dong-Guk Paeng, Jaeil Lee (Dept. of Ocean System Eng., Jeju Nat'l Univ., 66 Jejudaehakno, Jeju 690-756, Republic of Korea), and Seongil Kim (Agency for Defense Development, Gyeongsangnam-do 645-016, Republic of Korea)

Acoustic signal processing can be applied to sonar, acoustic communications and multimedia applications. Especially transmitting acoustic signal to the desired direction has many applications in military and industry. In this paper, we present digital beamsteering algorithm using parametric array transducers. To generate the desired beampattern, the proposed algorithm utilizes the complex weight instead of time delay. With this complex operation, many algorithms including constant beamwidth algorithm and linear constrained beamforming algorithms can be applied to make versatile transmitting beampatterns. To verify the generated beampattern, we build acoustic beamsteering system by using commercial transducers and LABVIEW software and hardware. The system includes GUI software, which allows user to change many parameters such as number of sensors, complex weight of each sensor, type of transmit data, etc. The GUI can also be easily modified and extended to accommodate the characteristics of transducers. With the laboratory experiments, we verified the performance of the algorithm and system. The proposed algorithm and system can be used to build many application systems such as multimedia, acoustic communications, and sonar system. [Work supported by Grant No. UD100014DD.]

2:30

5pEA7. Results from a prototype tetrahedral array for tracking sound sources in shallow water. Kay Gemba and Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, gemba@hawaii.edu)

A passive acoustic system aimed at tracking sound sources in harbor and near-shore environments is currently being developed. At the expense of decreased directivity for the same number of elements, volumetric arrays have an advantage over line or planar arrays since beam patterns can be steered both vertically and horizontally. In this paper, the tracking performance of a tetrahedral array for mid- to high-frequency (1–40 kHz) sound sources is evaluated. Modeled tracking performance is compared with measured results obtained using a prototype array in a controlled environment. The performance of several different processors is being investigated, and difficulties associated with shallow-water environments including complicated and time-varying multipath structure and varying background noise conditions are considered. Primary criteria for selecting processors of interest are robustness and computational efficiency. [This work is funded by the U.S. Department of Homeland Security through the Center for Island, Maritime and Extreme Environment Security.]

2:45

5pEA8. Synchronizing data for tracking moving targets with passive acoustic arrays. Geoffrey Goldman (U.S. Army Res. Lab., Adelphi, MD, geoffrey.goldman@us.army.mil)

To accurately track targets moving at high speed with passive acoustic arrays, the propagation delay time needs to be accounted for in the signal processing algorithms. The data from each array need to be synchronized to a common reference time relative to the acoustic signal generated at the target. One approach is to synchronize the data at each array by simply delaying it, then estimates of the direction of arrival (DOA) and possibly the time of arrival (TOA) can be performed. Alternatively, non-synchronized estimates of the DOA and TOA, calculated relative to a common sample time at the sensors, can be synchronized by interpolation based upon the estimated propagation delay time. For a kinematic target model with constant acceleration, the propagation delay time from the target to the acoustic array can be determined by solving a fourth-order polynomial equation. For many scenarios, the propagation delay time can be accurately estimated by performing a second-order or third-order Machaurin series approximation. The target kinematics can be estimated using standard algorithms that initially assume no propagation delays in the data. As the delays are incorporated into the estimates of DOA and TOA, the results will improve.

3:00—3:15 Break

3:15

5pEA9. Subsampled array design for compressive sensing. Charles F. Gaumond and Geoffrey F. Edelmann (Naval Res. Labortatory, 4555 Overlook Ave. SW, Code 7140, Washington, DC 20375, mardi.hastings@gatech.edu)

Optimal array design is considered for beamforming using the compressive sensing technique. Compressive sensing is an l1 technique suitable for solving inverse problems that are highly undersampled and would be considered ill-posed by conventional l2 methods. Consequently, fewer array elements are required to successfully detect targets and suppress sidelobes and noise. An array does not need to be spaced at half-wavelength intervals, nor are the beams required to be orthonormal. In fact, compressive sampling works best when sample points are at apparently random intervals (satisfying the uniform uncertainty principle). This paper considers conventional, subsampled, random, logarithmic, and Golomb spaced arrays for the purpose of target detection via compressive sensing. [Work supported by the ONR.]

3:30

5pEA10. Exploiting towed array maneuvers for online maximum likelihood field directionality estimation. Jeffrey S. Rogers (Naval Res. Lab., Washington, DC 22375) and Jeffrey L. Krolik (Duke Univ., Durham, NC 27708)

Passive estimation of the time-varying field directionality using a towed acoustic array remains a central problem in passive sonar. Typically, array processing algorithms are designed around the assumption of a linear array and fail to take advantage of the dynamics associated with a towed array. Performance of such systems can often be complicated by left-right ambiguities associated with the array as well as poor resolution near endfire directions. In this paper, the dynamics of the towed array are exploited allowing for left-right disambiguation as well as improved endfire resolution. A new method for online spatial spectrum estimation is presented. The maximum likelihood (ML) of the time-varying field is solved for using a single expectation maximization step after each received data snapshot. A multi-source, dynamic simulation is used to illustrate the proposed algorithm's ability to suppress ambiguous towed-array backlobes and resolve closely spaced interferers near endfire which pose challenges for conventional beamforming approaches, especially during array maneuvers. Simulation results suggest the online ML method offers improved M-of-N detection performance over that of conventional beamforming. [This work was supported by ONR.]

3:45

5pEA11. Optimum array design to maximize fisher information for bearing estimation in spatially correlated noise environment. Saurav R. Tuladhar and John R. Buck (Dept. of ECE, Univ. of Massachusetts Dartmouth, 285 OldWestport Rd., North Dartmouth, MA 02747, stuladhar @umassd.edu)

Source bearing estimation is a common application of linear sensor arrays. The Cramer–Rao bound (CRB) sets a lower bound on the achievable mean square error (MSE) of any unbiased bearing estimate. In the white noise case, the CRB is minimized by placing half of the sensors at each end of the array [MacDonald and Schultheiss, JASA (1969)]. However, many realistic ocean environments have a mixture of both white noise and correlated noise. In many shallow water environments, the correlated ambient noise can be modeled as cylindrically (two-dimensionally) isotropic. This research designs a fixed aperture linear array to maximize the fisher information for bearing estimation under these noise conditions. The optimum array is designed to maximize the minimum fisher information over an *a priori* bearing range. The elements of the optimal array are located closer to the array ends than uniform spacing, but are not so extreme as in the white noise case. The optimal array results from a trade off between minimizing the measured noise and maximizing the array bearing sensitivity. Depending on the source bearing, the resulting improvement in MSE performance over a uniform array is equivalent to a 3 to 5 dB improvement in input SNR. [Work supported by ONR.]

4:00

5pEA12. Hermetic array processing for underwater echo-ranging systems. Harvey C. Woodsum (SoneSys, LLC, 21 Continental Blvd., Merrimack, NH 03054)

Hermetic array processing provides enhanced beam width resolution, as well as correspondingly greater gain against ambient noise and reverberation, without the requirement of data-adaptive processing. The technique derives from the discrete hermetic (spectral) transform, and has been applied to an underwater echo-ranging system deployed in an underwater security system. Theoretical and empirical results are also described, and the impact on sonar system performance is presented. A description of practical implementation issues and a comparison to traditional adaptive beamforming are shown.

4:15

5pEA13. Affects of nearby bubbles on underwater array gain. R. Lee Culver and J. Daniel Park (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804)

Combining multiple sensor signals coherently (i.e., beamforming) improves spatial or angular resolution and increases signal to noise ratio (SNR). When the array is steered, signals arriving from the steering direction add in phase, while signals arriving from other directions do not (proper choice of signal frequency assumed). Array gain (AG) is a measure of how much the SNR at the array output is increased relative to array input SNR. The degradation in underwater acoustic array AG by scattering from nearby bubbles was measured at the AB Wood tank located at the Institute of Sound and Vibration Research (ISVR), University of Southampton, in June 2008. AG degradation is separate from the effects of bubbles in water to attenuate acoustic signals. Measured statistics of signal phase at the individual sensors show that as bubble density increases, phase differences between the elements increase and AG is degraded. We present a theory and numerical simulation that attributes the phase shifts to scattering from nearby bubbles and provides a way to predict AG degradation from the bubble density. [Work sponsored by the ONR Undersea Signal Processing.]

Session 5pPA

Physical Acoustics and Biomedical Acoustics: Nonlinear Acoustic Waves and Their Characterization

Oleg A. Sapozhnikov, Cochair
Moscow State Univ., Acoustics Dept., Moscow, 119992 Russia

Robin O. Cleveland, Cochair
Boston Univ., Dept. of Mechanical Engineering, 110 Cummington St., Boston, MA 02215

Invited Papers

1:00

5pPA1. Polyvinylidene fluoride membrane hydrophone low-frequency response to medical shockwaves. Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Adam D. Maxwell (Univ. of Michigan, Ann Arbor, MI 48109), Yuri A. Pishchalnikov (Indiana Univ. School of Medicine, Indianapolis, IN 46202-5120), and Oleg A. Sapozhnikov (Univ. of Washington)

Lithotripsy shockwaves are particularly difficult to measure because of their wide signal bandwidth and large pressures. A polyvinylidene fluoride (PVDF) membrane hydrophone and preamplifier were built and tested. A broad-focus electromagnetic lithotripter was used to calibrate the PVDF hydrophone. A fiber optic probe hydrophone (FOPH) with known impulse response was used as a measurement standard for secondary calibration. A low-frequency circuit model for the PVDF membrane electrodes in an infinite conductive medium was developed. The model response was compared with signals recorded by the FOPH and PVDF hydrophone at different levels of water conductivity ranging from 1 to 1300 μS/cm. Measured waveforms were distorted by high-pass filtering effects of the water conductivity. The model results showed good agreement with the measured waveforms and provided a correction for the system. When the input impedance was altered appropriately or the hydrophone was submerged in a nonconductive fluid, the PVDF and FOPH waveforms appeared nearly identical. The PVDF hydrophone is capable of measuring lithotripsy shockwaves accurately when the low-frequency response is properly taken into account. [Work supported by NIH DK43881, NIH EB007643, and NSBRI through NASA NCC 9-58.]

1:20

5pPA2. Advantages and limitations of the fiber-optic probe hydrophone for characterization of shock waves in water. Yuri A. Pishchalnikov, D. Felipe Gaitan, Mark S. Einert (Impulse Devices, Inc., 13366 Grass Valley Ave, Grass Valley, CA 95949, yurapish@gmail.com), Michael R. Bailey, Oleg A. Sapozhnikov (Univ. Washington, Seattle, WA), and James A. McAteer (Indiana Univ. School of Medicine, Indianapolis, IN)

The fiber-optic probe hydrophone (FOPH) (RP Acoustics, Leutenbach, Germany) is the standard for shock wave measurement, as it is omnidirectional with a flat frequency response ranging from static pressure to several megahertz. The FOPH calibration is determined from the equation of state of water, the optical refractive index of the glass/water interface, and the dc level of reflected light. We tested the accuracy of this calibration by placing the sensitive tip of the FOPH under static pressure up to 140 MPa. The FOPH gave accurate readings of applied static pressures provided there were no defects in the fiber. Defects (cracks and chips) in the glass fiber were difficult to control and could occur during routine handling: stripping, cleaving, or mounting. Such defects led to spurious spikes in measured waveforms. Defects were also caused by cavitation damage to the fiber. In addition, cavitation bubbles on the fiber compressed the fiber and resulted in distorted waveform measurement. Thus, although the FOPH is omnidirectional and accurate from zero to tens of megahertz, it is also susceptible to minute defects in the fiber and to cavitation bubble collapse along the fiber. [This work was supported in part by NIH-DK-43881 and by Impulse ACPT Contract No. W9113M-07-C-0178.]

1:40

5pPA3. Nonlinear propagation and shock waves in histotripsy ultrasound therapy. Adam D. Maxwell, J. Brian Fowlkes, and Zhen Xu (Dept. of Biomedical Eng., Univ. of Michigan, 1107 Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, adamdm @umich.edu)

Histotripsy is a noninvasive therapy that applies short duration, highly focused pulses of ultrasound to mechanically break down targeted tissues by acoustic cavitation. These pulses become strongly distorted due to nonlinear propagation, and shock fronts are formed at the transducer focus. Focal pressure levels at which cavitation clouds are achieved can exceed 20 MPa peak negative pressure and 100 MPa peak positive pressure. Current methods for characterizing histotripsy pulses are discussed. Single-mode and multi-mode fiber optic hydrophones have been constructed in our laboratory for measuring histotripsy waveforms. Accurate measurement of acoustic waveform shape and amplitude is important because nonlinear distortion contributes greatly to generating cavitation in histotripsy. High-speed photographic observations suggest that cavitation clouds are not formed directly by the negative pressure of the wave, but by scattering of shocks from single cavitation bubbles within the focal region. Experiments and simulations describing the role of shock waves in forming cavitation clouds during histotripsy are presented. [Work supported by NSF GRFP and NIH R01 EB008998 and S10 RR022425.]

2:00

5pPA4. Experimentally validated multiphysics computational model of refracting shock wave lithotripter. Dan Fovargue, Sorin Mitran (Dept. of Mathematics, Univ. of North Carolina at Chapel Hill, 445 Chapman Hall, Chapel Hill, NC 27599), Georgy Sankin, Nathan Smith, and Pei Zhong (Duke Univ., Durham, NC 27708)

We will present a computational model of the focusing of an electromagnetically induced pressure pulse by an acoustic lens and subsequent shock wave formation. Linear elasticity equations for the lens are solved simultaneously with Euler hydrodynamic equations for water considered to be a compressible medium with a Tait equation of state. The adaptive mesh refinement model allows both three-dimensional and two-dimensional axisymmetric computations. A number of coupling approaches at the lens-water interface are investigated by comparison to experimental results: transfer only of pressure boundary condition, coupling of displacement velocities, and a buffer linear elasticity region in the water immediately adjacent to lens. The model is validated against single-medium measurements (water or lens material), and the complete experimental shock wave formation process. Initial results from a crack propagation model in stone simulants placed at the lithotripter focus will also be presented.

2:20

5pPA5. Measurements of spark-generated *N*-waves in air using a combination of acoustical and optical methods. Philippe Blanc-Benon (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Universit de Lyon, 36 Ave. Guy de Collongue, F-69134 Ecully Cedex, France, philippe.blanc-benon@ec-lyon.fr), Petr V. Yuldashev (Moscow State Univ., Moscow 119991, Russia), Sbastien Ollivier (Universit de Lyon, F-69134 Ecully Cedex, France), Mikhail V. Averiyanov, Oleg Sapozhnikov, and Vera A. Khokhlova (Moscow State Univ., Moscow 119991, Russia)

Accurate measurement of broadband acoustic signals in air, particularly *N*-waves, remains a challenge. Bandwidth of existing microphones typically does not exceed 150 kHz, which results in significant overestimations of the shock rise time. To better resolve the shock thickness, it is proposed to use a focused optical shadowgraphy technique. The approach is tested experimentally. A spark source is used to generate high amplitude *N*-waves in air. Acoustic measurements are performed using conventional microphones (3 mm diameter), and optical shadowgrams are made using a collimated light beam from a 20-ns flash source. The results of modeling based on the generalized Burgers equation are in a good agreement with the microphone measurements in respect to the wave peak pressure and duration. However, the measured rise time of the front shock is ten times longer than the calculated one, which is attributed to the limited bandwidth of the microphone. The recorded optical shadowgrams in the vicinity of the shock front are compared with shadow patterns predicted theoretically. It is shown that a combination of microphone measurements and focused optical shadowgraphy is a reliable way of studying evolution of spark-generated shock waves in air. [Work supported by RFBR, French Government, and CNRS PICS 5603.]

2:40

5pPA6. Airborne shocks: Measurement methods and challenges. Thomas B. Gabrielson (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, MS 6120D, State College, PA 16804)

While most nonlinear phenomena are intriguing, the nature and behavior of acoustic shocks are particularly captivating. Rather ordinary sources can produce traveling wave fronts with pressure rises approaching 1% of atmospheric pressure on microsecond time scales. The resulting pressure signature can be challenging to record accurately; such measurements often require special sensors and always require an understanding of the influence of the sensor on the pressure field. Without careful design, such a measurement can become a characterization of the microphone itself with little information about the shock. There are a number of techniques for correcting the aberrations introduced by measurement microphones and a number of approaches to constructing special sensors. This presentation will summarize a subset of these techniques primarily in the context of measurement of weak shocks—shocks that travel at approximately the ordinary speed of sound but that have rise times from 10 μs down to a few tenths of a microsecond.

3:00—3:15 Break

Contributed Papers

3:15

5pPA7. Characterization of nonlinearly distorted ultrasound waves in water using broadband laser vibrometry. Oleg A. Sapozhnikov (Dept. of Acoust., Moscow State Univ., Moscow, Russia, oleg@acs366.phys.msu.ru), Bryan W. Cunitz, and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105 bailey@apl.washington.edu)

Laser vibrometry is a practical method to detect surface displacement. The method enables a direct measurement of acoustic field parameters such as acoustic particle displacement or acoustic particle velocity. Unlike other sensors, e.g., hydrophones, laser vibrometers are completely non-contact. Such devices are capable of measuring displacements from centimeters to sub nanometers at frequencies from near dc to 10 s of megahertz and have been proven to establish a primary standard for calibrating hydrophones [Bacon, IEEE Trans. UFFC, **35** (1988)]. In this technique, an ultrasonic transducer radiates an acoustic wave which is detected by a thin plastic membrane—a pellicle. The pellicle is effectively transparent to the acoustic beam so that the vibration of the pellicle follows the particle motion in the sound wave, but is reflective to the optical beam of the vibrometer allowing for a measurement. The present talk will report on measurements of nonlinearly distorted sawtooth waves in water performed with two commercial Polytec laser vibrometers: a scanning 24 MHz bandwidth system and a non-scanning 600 MHz bandwidth system. It is shown that appropriately chosen optical targets—pellicle or thick glass block with flat sides—allow resolution of both shock front and the smooth part of the waveform. [Work supported by NIH EB00764, NIH DK43881, and NSBRI through NASA NCC 9-58.]

3:30

5pPA8. Acoustical filters and nonlinear acoustic wave propagation in liquids. Cristian Pantea and Dipen N. Sinha (Los Alamos Natl. Lab., Mater. Phys. and Applications, MPA-11, MS D429, Los Alamos, NM 87545)

Generation of nonlinear acoustic waves of low frequency, less than 100 kHz, with associated low beam spread has potential imaging applications in highly absorptive media as encountered in the downhole geothermal and oil exploration environment. One of the particular advantages of the nonlinear acoustics approach of wave generation is the narrow beam spread of the dif-

ference frequency component. The difference frequency beam spread is determined by the beam profile of the primary high-frequency components used in the nonlinear mixing process. In the present study, we investigate the influence of acoustical filters in the beam path on the propagation of the generated difference frequency beam. Primary frequencies of approximately 1 MHz are used, resulting in a difference frequency component of around 100 kHz. Our results show that the difference frequency beam maintains its collimation even after passing through different acoustical filters where the primary frequencies are blocked and there is no more frequency mixing of the primaries. However, the amplitude of the difference frequency component starts decaying in an exponential fashion as predicted.

3:45

5pPA9. Statistical analysis of a finite-amplitude sinusoid. Micah R. Shepherd, Amanda D. Hanford (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804), and Kent L. Gee (Brigham Young Univ., Provo, UT 84604)

Pressure amplitude statistics have been used in several experimental studies of jet and rocket noise to indicate and study nonlinear acoustic propagation. Statistics of pressure amplitude time derivatives have also been shown to be sensitive to nonlinearity in noise. To illustrate the sensitivity of statistics to nonlinearity, the nonlinear propagation of a pure sinusoid is considered. The probability density function, standard deviation, skewness, kurtosis, and crest factor are computed for both the amplitude and amplitude time derivatives as a function of distance. The amplitude statistics vary only in the postshock realm, while the amplitude derivative statistics vary rapidly in the preshock realm. Using statistical analysis also suggests that the sawtooth onset distance can be considered to be earlier than previously realized.

4:00

5pPA10. Experimental characterization of shock formation distance in broadband noise propagation. Michael B. Muhlestein and Kent L. Gee (BYU Dept. of Phys. and Astronomy, N283 ESC, Provo, UT 84602)

Unlike initially sinusoidal waveforms, the definition of the "shock formation distance" for broadband noise is complicated by the fact that not all shocks form at the same rate. Examination of the concept of a "characteristic" shock formation distance for broadband noise raises some questions: Is there some generalization of the shock formation distance for sinusoidal signals or narrowband noise that can be applied to broadband noise? If so, is it inversely proportional to amplitude and frequency, as the pure tone and narrowband noise distances are? Possible methods to investigate these questions, such as statistical analyzes and examining spectral slopes and time waveform derivatives, are discussed. These techniques have been studied using experimental data acquired from finite-amplitude broadband noise propagation in an anechoically terminated plane-wave tube. Preliminary results from these experiments are discussed.

4:15

5pPA11. Propagation of radially polarized shear wave beams with cubic and quadratic nonlinearities. Kyle S. Spratt, Mark S. Wochner, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

This presentation describes an extension of the work by Wochner *et al.* [J. Acoust. Soc. Am. **123**, 2488–2495 (2008)], wherein a coupled pair of nonlinear parabolic equations was derived for the two components of the particle motion perpendicular to the axis of a shear wave beam in an isotropic hyperelastic medium. Although the equations derived in that work contain both cubic and quadratic nonlinearities, the latter were discarded based on the fact that they are not present in plane shear waves, and hence assumed negligible within the quasiplane paraxial region of linearly polarized shear wave beams. However, the experimental observation of prominent second harmonic generation by nonplanar shear waves in soft solids [Jacob *et al.*, J. Acoust. Soc. Am. **122**, 1917–1926 (2007)] has motivated a more thorough understanding of these quadratic nonlinear terms. In the present work, the quadratic nonlinearity is explored, and, in particular, it is found that the form of this nonlinearity simplifies for axisymmetric radially polarized beams. Numerical results are presented comparing the effects of the cubic and quadratic nonlinearities on the generation of harmonics along the beam axis and the directivity patterns of those harmonics. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:30

5pPA12. Optical freezing of Faraday waves for precise pore distributions in tissue scaffolds. A. Sampathkumar (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, ashrex@bu.edu), C. M. Klapperich (Boston Univ., Boston, MA 02215), J. R. Saylor (Clemson Univ., Clemson, SC 29634), and R. G. Holt (Boston Univ., Boston, MA 02215)

Collagen-glycosaminoglycan (GAG) tissue scaffolds have been widely employed to function as the structural support and nutrient-supply for *in vitro* and *in vivo* cell growths. The traditional scaffold fabrication process may take up to 3 days to complete, with the pattern formation step requiring nearly 24 h. We describe a bench-top optical curing system that implements a novel fluid-acoustic technique that uses Faraday waves for rapid fabrication/prototyping of tissue scaffolds on the time-scale of minutes. The scaffold patterns and spacings are easily variable by varying the frequency and the amplitude of the Faraday waves. The experimental setup incorporates a 450 mJ/pulse Nd:YAG laser operating at 532 nm to photopolymerize the Faraday wave patterns generated using 30 kHz vibrations on the GAG emulsions. Preliminary results on freezing Faraday patterns on photoresists and a study of the acoustic driving and the optical curing parameter spaces will be presented.

4:45

5pPA13. Characterization of the acoustic field of a clinical electromagnetic shockwave therapy device. Camilo Perez, Hong Chen, and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Extracorporeal shockwave therapy devices are used in clinical settings for different medical applications such as orthopedics and urology. Having several clinical devices out in the field creates a challenge when comparing treatments and energy deposition mechanisms between different devices. In this work, the field of an electromagnetic shockwave device (Duolith SD1 T-Top, Storz) was characterized using a fiber optic hydrophone (FOPH2000, RP Acoustics). The acoustic field from two hand-held probes was measured: one probe was focused (with different length coupling cones) and the second one was a ballistic (radial therapy) probe. With the focused probe, measured pressures ranged from 45 MPa peak-positive to 12 MPa peak-negative. Axial and transverse beam profiles were acquired while analyzing the peak-positive and peak-negative pressures at each machine energy level and pulse repetition frequency. The focused source showed an extended -6 dB peak-positive focal region along the axis of propagation and shorter in the orthogonal planes to the propagation ($30 \times 3 \times 3$ mm^3). Linear scans along the axis of propagation showed quadratic decay distal to the focus. Measured peak-negative pressures were higher pre-focal than post-focal. The results compared qualitatively, but not quantitatively with manufacturer specifications. [Work supported by NIH AR053652.]

Session 5pSC

Speech Communication: Structure and Rhythm (Poster Session)

Benjamin R. Munson, Chair
Univ. of Minnesota, Dept. of Speech, Languages, Hearing Science, 164 Pillsbury Dr., S.E., Minneapolis, MN 55455

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

5pSC1. English and Russian listeners perceive cues to lexical pitch accent differently. Irina A. Shport and Susan Guion-Anderson (Linguist. Dept., Univ. of Oregon, Eugene, OR 97403-1290)

This study examines the difference in perception of the fundamental acoustic cue to Japanese lexical pitch-accent—a sharp F0 fall following the F0 maximum—by native listeners of two stress languages, English and Russian. In both languages, F0 fall is not a cue to stress, but intonational high-low pitch accents exist. Previous work showed that English listeners were sensitive to the F0 fall in a discrimination task; however, they did not use the F0 fall information in a pitch pattern categorization task [Shport and Guion-Anderson, J. Acoust. Soc. Am. **127**, 2024 (2010)]. Similar results were expected for native Russian listeners. Preliminary data suggest that Russian and English listeners differ in their perception of Japanese pitch patterns. Russian listeners are slightly less sensitive to the F0 fall in a discrimination task than English listeners; however, they appear to employ this acoustic cue for pattern categorization. Specifically, patterns without the F0 fall tend to be perceived as having a prominent first syllable as compared to patterns with the F0 fall. This response tendency is the reverse of Japanese listeners. The difference between English and Russian listeners is discussed in terms of native-language prosody shaping the perception of foreign-language prosody.

5pSC2. Effect of language, speaking style and talker on a spectral rhythm measure. Gayatri Rao (Dept. of Psych., Univ. of Texas at Austin, 1 University Station A8000, Austin, TX 78712-0187) and Rajka Smiljanic (Univ. of Texas at Austin, Austin, TX 78712)

Recently, two spectro-temporal approaches have been proposed for characterizing cross-language and disordered speech rhythm: power spectrum [Tilsen and Johnson (2008)] and speech envelope modulation spectrum (EMS) [Liss *et al.* (2010)]. These measures quantify the rhythmicity of speech within specified frequency bands. In the present study, the EMS measure was employed to assess the effect of language, speaking style, and talker on rhythm. In order to examine the rhythmic characteristics of conversational and speaking styles in English and Croatian, utterances produced by two female speakers of each language in clear and conversational speaking styles were analyzed [Smiljanic and Bradlow (2005)]. The spectral centroid, a weighted mean of a spectrum, was used to measure the center of mass of the EMS extracted for each utterance. Results from nested factorial analyzes revealed that the spectro-temporal characteristics differed for English and Croatian, for the conversational and clear speaking sentences, and for individual talkers. These results suggest that the low rate amplitude modulations of a signal can be used to distinguish rhythmic properties across languages, speaking styles, and talkers. Furthermore, these automated spectral measures provide a new means for characterizing rhythm that bypasses laborious manual measurements and phonologically dependent segmentation criteria.

5pSC3. Stable production rhythms across languages for bilingual speakers. Kathy M. Carbonell (Dept. of Speech, Lang. & Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721-0071, alotto@email.arizona.edu), Kaitlin L. Lansford, Rene L. Utianski, Julie M. Liss (Dept. of Speech & Hearing Sci., Arizona State Univ., P.O. Box 870102 Tempe, AZ 85287-0102), Sarah C. Sullivan, and Andrew J. Lotto (Univ. of Arizona, Tucson, AZ 85721-0071)

There has been a great deal of work on classifying spoken languages according to their perceived or acoustically-measured rhythmic structures. The current study examined the speech of 12 Spanish-English bilinguals producing sentences in both languages using rhythmic measures based on the amplitude envelopes extracted from different frequency regions—the envelope modulation spectrum (EMS). Using discriminant factor analysis, EMS variables demonstrated a moderate ability to classify the language being spoken suggesting that rhythmic differences between languages survive even when speaker is controlled. More interesting is the fact that EMS variables could reliably classify which speaker produced each sentence even across languages. This result suggests that there are stable rhythmic structures in an individual talker's speech that are apparent above and beyond the structural constraints of the language spoken. The EMS appears capable of describing systematic characteristics of both the talker and the language spoken. [Work supported by NIH-NIDCD].

5pSC4. Comparison of prosodic properties of intonation in beijing Mandarin and Taiwan Mandarin. Jianjing Kuang and Grace Kuo (Dept. of Linguist., Phonet. Lab., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095-1543, kuangjj@gmail.com)

In many studies, Beijing Mandarin and Taiwan Mandarin are considered to be identical because they share the same lexical tone system (except for a subtle difference in tone 3). However, native speakers of the two dialects can easily differentiate one from the other. This study investigates the intonation patterns of the two dialects; it will be shown that there are important prosodic differences in the boundary cues and use of pitch. The data were collected from recordings of "Little Red Riding Hood" read by eight native speakers (four from Beijing, and four from Taiwan). The preliminary results show that Beijing Mandarin and Taiwan Mandarin have different boundaries cues: at the phrase level, although both Beijing and Taiwan Mandarin have final lengthening, Taiwan Mandarin has no intensity reduction and Beijing Mandarin has a high boundary tone. At the sentence level, obvious lengthening and stress placement on final particles are observed in Taiwan Mandarin, while those particles are reduced in Beijing Mandarin. Furthermore, the pitch trajectory differed in some sentence types, e.g., pitch declination after nuclear-accented wh-words in Taiwan Mandarin but not Beijing Mandarin.

5pSC5. Rhythmic speech perception predicts novice musical composition in English and French. John G. Neuhoff (Dept. of Psych., The College of Wooster, Wooster, OH 44691) and Pascale Lidji (McGill Univ., Montreal PQ H3A 1B1, Canada)

Music and speech have long been thought to have common cognitive underpinnings, and recent work demonstrates that the music of expert composers reflects the speech rhythm of their native language [H. Ollen (2003), P. Daniele (2003)]. In the current study, monolingual English speaking music novices composed simple "English" and "French" tunes on a piano keyboard. The rhythms produced reflected speech rhythms perceived in English and French, respectively. Yet, the pattern was opposite that produced by expert English and French composers and opposite that predicted by the acoustic determinants of speech rhythm that specify English speech as more rhythmically varied than French. Surprise recognition tests 2 weeks later confirmed that the music-speech relationship remained over time. Participants then rated the rhythmic variability of French and English speech samples. We found that native English speakers perceived French as more variable than English despite the measured greater variability of English. Finally, we repeated these procedures with a sample of monolingual French speakers and found similar, but opposite effects. The results suggest that common cognitive underpinnings of music and speech rhythm are more widespread than previously thought, and that novice rhythm production in music is concordant with perceived speech rhythms.

5pSC6. Measuring linguistic and functional performance of hand-held speech-to-speech interpreters. Jared Bernstein (Pearson, 299 S. California Ave., Palo Alto, CA 94306, jared.bernstein@pearson.com) and Elizabeth Rosenfeld (Tasso Partners, 1330 Tasso St., Palo Alto, CA 94301)

In 2011, voice-in/voice-out interpretation in several language pairs is available for mobile phones. Computers with internet access can translate text between many language pairs. These machines interpret and translate quickly, but not always accurately. If current portable devices support basic bidirectional communication for travel or in commercial transactions, what is the future of second language education? This work evaluates the performance of hand-held interpretation analytically and functionally. A study was conducted using current hand-held devices for English-Spanish and Spanish-English interpretation. Ten Spanish and ten English monolinguals performed three simple interactive functional tasks designed to assess different levels of spoken language proficiency. The resulting performances were transcribed then processed by a machine translator and spoken into a hand-held machine interpreter. Analytic measures (word error rate, Bleu, intelligibility) were made comparable for the three processes (recognition, translation, synthesis), in both the Spanish-English and English-Spanish directions to find the weak links in the interpretation chain. The data also provided an estimate of the functional level of current hand-held devices with reference to second language communication standards used in applied linguistics.

5pSC7. A cross-linguistic study of intonation: English learners' production of Mandarin intonation. Yanhong Zhang (Dept. of Asian Lang. and Cultures, Rutgers Univ., NB, NJ 08901, yzhang648@gmail.com) and Jiahong Yuan (Univ. of Pennsylvania, Philadelphia, PA)

Learners of non-tonal languages have to learn to manipulate complicated F0 patterns arising from the interaction between lexical tones and sentential intonation. This study examines how native English speakers deal with the effect of lexical tones in producing Mandarin intonation. Considering the boundary tone is indicative of statement-question contrasts, this study compares tonal features of sentential final syllable carrying four lexical tones in different intonations by native Mandarin and native English speakers. Acoustic analyzes indicated that both English and Mandarin speakers produced higher F0 of the boundary tone in question than in statement regardless of the lexical tones, but no difference in duration and intensity between statement-question contrasts was found. Pitch values in English speakers' utterances were generally lower than that of Mandarin speakers. Perceptual test showed that the accuracy rate of statements across four lexical tones for English and Mandarin speakers were close to each other, but English speakers' question utterances were poorly perceived in comparison to Mandarin speakers' utterances, suggesting that English speakers had difficulty in producing Mandarin question sentences, especially when the final syllable carries tone 1 or tone 4. English speakers' poor performance on question intonation was attributed to the influence of their native prosodic system.

5pSC8. Speech rhythm and pitch patterns in Bengali: Implications for prosodic acquisition. Sameer ud Dowla Khan (Dept. of Cognit., Linguistic, and Psychol. Sci., Brown Univ., 229 Waterman St., Providence, RI 02912, sameeruddowlakhan@gmail.com), Kristine Yu, and J'aime Roemer (Univ. of California Los Angeles, Los Angeles, CA 90095, jaime.roemer@gmail.com)

Rhythmic timing, measured as durational variability of consonantal and vocalic intervals, is thought to be a cue for phonological chunking in infant acquisition [Ramus *et al.* (1999)], i.e., for learning stress-timing (English) versus syllable-timing (Spanish). Another source of rhythm that has been less studied in infant speech development is "macrorhythm" in pitch contours, regularity in $f0$ marking of prosodic structure [Jun (2005)]: Head-marking languages with variable contours on each word (English) are distinguished from edge-marking languages with a recurring pattern on each word (Korean). Bengali is a language with a moderately constrained syllable structure, and both head- and edge-marking: weak stress and a recurring rise on each content word [Khan, (in press)]. To explore how Bengali can be best categorized in terms of rhythmic timing and macrorhythm, and how speakers may adjust these properties to aid infant language acquisition, we analyzed various durational and intonational measurements in recordings of ten speakers reading a passage in both laboratory speech and simulated infant-directed speech (IDS) styles. Preliminary results suggest that while Bengali has syllable timing and strong macrorhythm, speakers do not highlight these rhythmic properties in IDS, but instead counteract them by increasing consonantal interval variability and interrupting the regular rising $f0$ patterns.

5pSC9. Perceptual study of Hawaiian in relation to rhythm classes. Diana Stojanovic (Dept. of Linguist., Univ. of Hawai'i at Manoa, Honolulu, HI 96822)

The existence of language rhythm-classes has long been disputed. While early criticisms focused on disproving claims of isochronous units, later studies examined whether various durational measures are able to classify languages in the manner that corresponds to results of perceptual studies. Most recently, the evidence coming from perceptual studies is re-examined by testing additional languages and different tasks, in some cases failing to show support for rhythm-class hypothesis. Tasks and format of stimuli differ across perceptual studies; in particular, infant studies employ long speech samples while studies with adults use short stimuli. Longer samples allow generalization and may be required for a successful discrimination. The present study investigates perceptual classification of Hawaiian (classified with English based on durational metrics, but with Japanese based on syllable structure) with respect to three posited rhythm types. English, French, and Japanese were used as class representatives. Two variants of AAX task were used: In one, 60 s samples for both A and X language were presented at the beginning of the experiment. Classification results for Hawaiian are discussed in relation to the 60 s learning period and typicality of the stimuli.

5pSC10. Interaction of acoustic and neighborhood density parameters on spoken word recognition. Kathryn L. Cabbage and Thomas D. Carrell (Univ. of Nebraska-Lincoln, 318Q Barkley Ctr., Lincoln, NE 68583, klcabbage@huskers.unl.edu)

The neighborhood activation model (NAM) [Luce and Pisoni (1998)] predicts the speed and accuracy of word retrieval in the mental lexicon based on word frequency and neighborhood density. Data from NAM experiments have demonstrated processing advantages of dense words over sparse words. Little is known about how acoustic, versus phonetic, variability influences processing strategies used by listeners of acoustical-degraded speech such as those using cochlear implants and hearing instruments. In this experiment, neighborhood density was investigated using speech-like sentences which preserved detailed frequency information but removed most amplitude information. Results yielded no difference between dense and sparse words when presented in this time-varying sinusoidal format [Remez *et al.* (1980)]. Consistent with prior research, amplitude modulation increased overall intelligibility of the stimuli; however, the effect of neighborhood density was in the opposite direction from that found in natural speech. We hypothesize that this may be indicative of the neighborhood density effect, becoming secondary to a phonotactic probability effect which has previously been associated with sublexical processing tasks. These results indicate that interactions between acoustic and phonetic factors must be accounted for when designing hearing instruments and cochlear implants. Follow-up investigation with other acoustic variables is currently underway in our laboratory.

5pSC11. The role of lexical processing in the discrimination of English vowel contrasts by Spanish learners. Teresa Lopez-Soto and Daniel Calvo-Carmona (Dept. of English Lang., Univ. of Seville, Palos de la Frontera, s/n 41004 Sevilla, Spain, teresals@us.es)

In our study we present the results of measuring the discrimination of English vowel contrasts that prove to be particularly complex for Spanish learners of English in a lexical versus non-lexical context. We have developed a series of discrimination tasks in which we have used both lexical and non-lexical (non-words) stimuli for three minimal pairs: cat-cut, seat-sit, and sit-set. A group of 30 Spanish speakers trained on the discrimination of monosyllabic CVC English words and non-words in a perception task that included vowel discrimination. Lexical and non-lexical minimal pairs were randomly presented and participants received feedback after their response. Results showed the following. (1) For all three vowel contrasts, perception accuracy as being measured in a discrimination task proved to be higher for lexical words (mean 79.2%) than for non-words (mean 57.6%) in the pretest. (2) After four 1-h perception training sessions, all participants improved in every vowel contrast, but the results were comparatively higher for non-lexical (mean 80.1%) than for lexical words (85%). A post-training questionnaire was used to make sure that participants could correctly identify both lexical and non-lexical words. We believe that this study lays ground for further investigation on the limits of phonological processing and lexical access.

5pSC12. Lexical effects on nasal coda neutralization in Taiwan Mandarin. Yingshing Li (Inst. of Linguist., 128, Section 2, Academia Rd. 115, Taipei, Taiwan, Republic of China, lngysli@gate.sinica.edu.tw)

Recent disputes concerning lexical effects on speech variability in speech production are whether these effects are due to lexically selective encoding processes or post-access phonological processes. This study investigates how lexical variables (inherent frequency and contextual predictability) affect nasal coda neutralization (alveolar /n/ versus velar /ŋ/) in Taiwan Mandarin, as measured in degree of vocalic nasalization (A1-P0 and A1-P1), formant transition (F1, F2, and F3), and nasal consonant duration and intensity. Participants read a random list of 527 monosyllabic Chinese morphemes in a speeded naming task. The results of linear regression reveal that (1) while sociolinguistic (genders and home languages), phonological (adjacent segmental features), orthographic (familiarity of characters and accuracy of transcription), and processing variables (response times and speaking rates) are factored out, lexical variables are found to enhance nasal coda neutralization; (2) however, a larger proportion of the variance is predicted by the correlation of both lexical and processing variables, suggesting that phonological planning and articulatory velocity are the means of achieving lexically-driven nasal coda neutralization.

5pSC13. Lexical influences on response times in Japanese listeners' recognition of spoken English words. Kiyoko Yoneyama (Dept. of English, Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi, Tokyo 175-8571, Japan, yoneyama@ic.daito.ac.jp) and Benjamin Munson (Univ. Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455)

One index of phonological knowledge of an L2 is the magnitude of phonological neighborhood density (PND) effects on word recognition. Imai *et al.* [JASA (2005)] showed that effects of PND on word-recognition accuracy were only present in advanced Spanish-speaking L2 English learners and not in beginning learners. This poster presents a follow up to Imai *et al.*'s study using Japanese L1 acquiring English as an L2 (JL1-EL2) and Native English listeners (NELs). Previous reports of this work [Yoneyama and Munson, J. Phonetic. Soc. Japan (2010)] failed to replicate Imai *et al.*'s findings, in that we observed strong effects of frequency and neighborhood density on the performance of three groups of listeners: both beginning and advanced JL1-EL2 speakers, and native English speakers. Here we investigate the influence of lexical factors on reaction times (RTs) in spoken-word recognition in the same listeners. A preliminary analysis of a subset of data comparing the least-proficient JL1-EL2 speakers to the NELs shows larger effects of PND on RTs for the JL1-EL2 listeners than for the NELs. Together, the findings suggest that lexical and phonetic factors influence second-language word recognition in complex, language-specific ways. [Work supported by NSF Grant No. BCS 0729277.]

5pSC14. Individual differences in spoken word recognition: Regional dialect variation. Terrin N. Tamati (Dept. of Linguist., Indiana Univ., Memorial Hall 322, Bloomington, IN 47405, ttamati@indiana.edu), Jaimie L. Gilbert, and David B. Pisoni (Indiana Univ., Bloomington, IN 47405)

Regional dialect is an important source of variation in the speech signal. However, little is known about individual differences in the ability to perceive and accommodate for this type of variability. In the current study, individual listeners' performance on a speech recognition task with different American English (AE) dialects was explored. Ninety-eight listeners completed a novel high-variability sentence recognition task (PRESTO), which contains sentences produced by talkers from seven AE dialect regions. Correlational analyzes were carried out on performance on the task and scores from a self-report questionnaire on executive function. Results revealed significant correlations between measures of behavioral regulation (executive function) and performance accuracy for several of the standard and non-standard talker dialects. This was especially the case for non-mobile listeners, who had lived in only one dialect region before the age of 18, as opposed to mobile listeners, who had lived in more than one dialect region before the age of 18. These findings indicate possible group differences based on both residential history and basic underlying cognitive abilities in adapting to differences in regional dialect variation. The results will be discussed with respect to individual differences in the perception of indexical properties of speech under adverse listening conditions.

5pSC15. The effect of semantic context on speech intelligibility in reverberant rooms. Nirmal Kumar Srinivasan and Pavel Zahorik (Dept. of Psych. and Brain Sci., Univ. of Louisville, Louisville, KY 40203)

Although it is well known that semantic context affects speech intelligibility and that different reverberant rooms affect speech intelligibility differentially, these effects have seldom been studied together. Revised SPIN sentences in a background of Gaussian noise in simulated rooms with reverberation time (T60) of 1 and 0.25 s were used. The carrier phrase and the target word of the speech stimuli were manipulated to be either in the same room or in different rooms. As expected, intelligibility of predictable sentences was higher compared to unpredictable sentences—the context effect. The context effect was higher in the low-reverberant room as compared to the high-reverberant room. When the carrier phrase and target words were in different rooms, the context effect was higher when the carrier phrase was in the low-reverberant room and target word in the high-reverberant room. For predictable sentences, changing the target word from high-reverberation to low reverberation with a high reverberant carrier increased intelligibility. However, with a low-reverberant carrier and different rooms for the target word, there was no change in intelligibility. Overall, it could be concluded that there is an interaction between semantic context and room acoustics for speech intelligibility.

5pSC16. Perception of acoustic cues in contrastive focus. Nadya Pincus (Univ. of Delaware, Ling. & Cog. Sci., 812 Lehigh Rd., Newark, DE 19711, npincus@udel.edu)

This research examines the ability of native English speakers to identify meanings associated with contrastive focus. Two conditions are compared: adjective and noun focus. Subjects hear either "I want the RED shirt" or "I want the red SHIRT," where the focused element receives an L+H* pitch accent and a longer duration. They must then choose which of two continuations best follows the auditory stimulus (e.g., "I don't like the blue shirt" versus "I don't like the red pants"). Preliminary results, based on 23 subjects listening to 10 sentences each, show that despite substantial acoustic differences between the focused and nonfocused words for both the adjective and noun conditions, each condition's result is vastly different. As a group, they excel at identifying the adjective-focused condition, but do poorly in the noun-focused condition. Closer examination reveals a bimodal distribution on the noun condition: One group has a high success rate (between 80 %–100%) and another has a low success rate (between 20%–40%). For the low success group, conflicting acoustic cues may result in a misinterpretation as phrase-final lengthening and/or natural focus on the object, while these cues are being overridden in the high success group.

5pSC17. Parsing the ambiguity of casual speech: "He was like" or "He's like"? Dan Brenner (Dept. of Linguist., U. Arizona, Douglass Bldg., Tucson, AZ 85721), Natasha Warner (U. Arizona, Tucson, AZ 85721), Mirjam Ernestus (Radboud U., Nijmegen, The Netherlands), and Benjamin V. Tucker (U. AB, Edmonton, Canada)

Reduction in casual speech can create ambiguity, e.g., "he was" can sound like "he's." Before quotative "like" ("so she's/she was like…"), it was found that there is little accurate acoustic information about the distinction in the signal. This work examines what types of information (acoustics of the target itself, speech rate, coarticulation, and syntax/semantics) listeners use to recognize such reduced function words. We compare perception studies presenting the targets auditorily with varying amounts of context, presenting the context without the targets, and a visual study presenting context in written form. Given primarily discourse information (visual or auditory context only), subjects are strongly biased toward past, reflecting the use of quotative "like" for reporting past speech. However, if the target itself is presented, the direction of bias reverses, indicating that listeners favor acoustic information within the target (which is reduced, sounding like the shorter, present form) over almost any other source of information. Furthermore, when the target is presented auditorily with surrounding context, the bias shifts slightly toward the direction shown in the orthographic or auditory-no-target experiments. Thus, listeners prioritize acoustic information within the target when present, even if that information is misleading, but they also take discourse information into account.

5pSC18. Effects of distal pitch and timing of speech and nonspeech precursors on word segmentation. J. Devin McAuley (Dept. of Psych., Michigan State Univ., East Lansing, MI 48824, dmcauley@msu.edu), Laura C. Dilley (Michigan State Univ., East Lansing, MI 48824), Prashanth Rajarajan, and Krista Bur (Michigan State Univ., East Lansing, MI 48824)

Recent work shows that word segmentation is influenced by distal prosodic characteristics of the input several syllables from the segmentation point. However, the conditions under which these distal prosodic effects operate in segmentation remain obscure. In experiment 1, participants heard eight-syllable sequences with a lexically ambiguous four-syllable ending (e .g., crisis turnip versus cry sister nip). The duration and/or pitch characteristics of the initial five syllables were resynthesized in a manner predicted to favor parsing of the final syllables as either a monosyllabic or a disyllabic word; the acoustic characteristics of the final three syllables were held constant. Experiment 1 replicated earlier results showing that utterance-initial prosody influences segmentation utterance, finally, using different utterance-initial prosodic characteristics than shown previously. In experiment 2, the first four syllables of the eight-syllable sequences were replaced with complex tones, which maintained the pitch and timing of the original syllables. Initial results suggest that the presence of tones induced participants to adopt a more conservative response criterion but did not affect their sensitivity to distal pitch and timing cues relative to a control, speech-only condition. These findings have implications for understanding how listeners segment words from fluent speech. [Research funded by NSF Grant No. BCS-0847653.]

5pSC19. Explaining boundary perception in spontaneous French conversation: Acoustic and phrasal metrics. Caroline L. Smith (Linguist., Univ. of New Mexico, MSC 03 2130, Albuquerque, NM 87131-0001, caroline@unm.edu)

Prosodic structure in speech derives from both linguistic structure, particularly syntax, and performance factors such as speech rate. The combination of influences means that predicting the structure of a spontaneous utterance remains a major challenge. One approach has been to survey listeners and take their perceptions as the basis for analysis [e.g., Cole *et al.*, Lab.Phonol. (2010)]. A study of this type in French compared perceptions of two types of spontaneous speech: informal conversation and more formal journalistic discussion. 25 naive native speakers of French listened to recordings of both types and indicated where they perceived boundaries between groups of words. Their responses were then compared to acoustic measurements and analyzes of the phrasal structure of the conversations. The presence of a pause and a rise in $F0$ were the most reliable acoustic correlates of a perceived boundary. Preliminary analyzes of phonological phrase structure suggest that the thresholded boundary weight hypothesis of Breen *et al.* [2010 Lang. Cog. Proc.] is a promising metric for explaining the pattern of boundary perception in these data. Locations were more often perceived as boundaries when the sum of the sizes of just-completed and upcoming constituents (measured in numbers of phonological phrases) is larger.

5pSC20. Dyadic postural coordination and discourse structure. Martin A. Oberg, Adriano Barbosa, and Eric Vatikiotis-Bateson (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, oberg @interchange.ubc.ca)

This study reports on the relationship between coordinated behavior in dyadic conversation and discourse boundaries in audio-visual and audio only conditions over various conversation tasks. Instantaneous inter- and intraspeaker correlations of time varying head motion and body posture are analyzed for cues marking turn-taking behavior and are compared against human-labeled discourse boundaries. Transitions in two-dimensional (time and delay lag) instantaneous correlation are compared across conditions. Implications for the structure of interaction and development of entrainment in dyadic conversation are discussed. Preliminary analyzes of correlations from mismatched conversations suggest the existence of rhythmic behaviors characteristic of conversation in general. Differences between AV v A only conditions suggest the importance of visual feedback in postural entrainment.

5pSC21. Vowel laxing in Indonesian as a test case for interaction of morphological and syllabic structure. Daniel R. McCloy (Dept. of Linguist., Univ. of Washington, Box 354340, Seattle, WA 98195-4340, drmccloy@uw.edu)

Lax (also known as, centralized) vowel allophones are attested in Indonesian for non-low vowels in closed syllables [e.g., Sneddon (1996)]. In consonant-final stems with vowel-initial suffixes (*ke+apik+an*), phonological theory (the maximal onset principle) predicts the stem-final consonant to syllabify with the suffix (*ke.a.pi.kan*) and the preceding vowel to manifest as unreduced. This study compares speaker-normalized formant values for such vowels against formant values for vowels in stem-final open syllables with obstruent-initial affixes (*men+jadi+kan*) and vowels in monomorphemic contexts (*tikam*). Word-final open and closed syllables (*jadi*, *cerdik*) are included as reference points in the vowel space. Male and female L1 speakers of standard Indonesian are recorded reading three randomizations of the word list. Data collection is ongoing, but preliminary results suggest that for front (unround) vowels in stem-final closed syllables with vowel-initial affixes (*ke+apik+an*), the formant values fall between the values for prototypical lax and non-lax Indonesian vowels; no clear pattern has yet emerged for back (round) vowels. Findings suggest that morphological structure (i.e., whether a segment "belongs" to stem or affix) may constrain syllabification or cause deviations from preferred syllable structures.

5pSC22. Effects of morpheme boundaries on the timing of fundamental frequency (F0) peaks. Seung-Eun Chang (Dept. of East Asian Lang. and Cultures, Univ. of California, Berkeley, 3413 Dwinelle, Berkeley, CA 94720, sechang71@berkeley.edu)

This research examines the effects of morpheme boundaries on the timing of fundamental frequency (f0) peaks and demonstrates that phonetic realization is influenced by morphological structure. It reports the timing of f0 peak and peak plateau in monomorphemic HH (high, high tone) and HL (high, low tone) as well as bimorphemic H+H and H+L in South Kyungsang Korean, spoken in the southeastern part of Korea. The results show that f0 peak came significantly later in bimorphemic H+H than in monomorphemic HH and peak plateau was significantly longer for monomorphemic HH than for bimorphemic H+H. It suggests that f0 peak plateau is implemented more stably within a single lexical item than across a boundary between lexical items. However, a significant morpheme effect was not found in monomorphemic HL and bimorphemic H+L. In accounting for these asymmetrical results, several hypotheses were discussed including speakers hyperspeech effort [H&H theory, Lindblom (1990)], intrinsic vowel height on f0 [Hombert *et al.* (1979)], and f0 peak delay effect on phonologically sug-

gested H tone spreading [de Jong and McDonough (1993)]. It is proposed that the bimorphemic H+H in this language is the phonetic image of the f0 peak delay effect of high tone.

5pSC23. Grouping and the verbal transformation effect: The influence of formant transitions. Marcin Stachurski, Robert J. Summers, and Brian Roberts (Psych., Sch. of Life & Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom)

Listening to a recorded word repeated many times produces verbal transformations (VTs) to different forms; this effect may be due in part to perceptual re-grouping of the speech sounds. Six CVC words producing strong formant transitions between the initial consonant and vowel (e.g., "short") were recorded and monotonized at F0=130 Hz; each word was paired with another producing weak formant transitions (e.g., "fort"). Each set of six words was used to derive another in which the CV transitions were edited out and replaced with samples selected from the neighboring steady-state portions. For each word pair, the amount of splicing applied to the weak-transition stimulus (30–70 ms) was matched to that required for its strong-transition counterpart, and the VTs obtained for 3-min sequences of the edited and unedited versions were compared. Listeners reported more VT forms in the spliced than in the unedited case for the strong-transition words, but not for those with weak transitions. Hence, the effect is not an artifact of splicing, but arises from the removal of the strong transitions. This finding supports earlier studies suggesting that formant transitions play an important role in binding disparate speech segments together into a single auditory stream. [Work supported by EPSRC.]

5pSC24. Acoustic correlates of stress and their use in diagnosing syllable fusion in Tongan. James White and Marc Garellek (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jameswhite@ucla.edu)

The first goal of this study is to determine which acoustic cues correlate with stress in Tongan. Preliminary data indicate that pitch ($F0$), duration, and vowel height ($F1$) are the best cues to stress. Differences are also found for vowel frontness ($F2$), intensity, and voice quality. The second goal is to use these acoustic stress correlates to test for syllable fusion, an alleged phonological process in Tongan by which sequences of vowels in separate syllables fuse into a single syllable (e.g., Poser 1985). Lower-to-higher sequences (e.g., /ai/) are said to undergo fusion while higher-to-lower sequences (e.g., /ia/) are not. However, no empirical studies have been conducted testing these claims. Using the acoustic stress cues identified for Tongan, these two types of sequences are compared, where the first vowel of each sequence falls in a stressed position. For higher-to-lower sequences (e .g., /ia/), the first vowel should show acoustic cues of stress whereas the second vowel should appear unstressed. For lower-to-higher sequences (e.g., /ai/), if syllable fusion occurs, acoustic stress cues should be present over the entire sequence; otherwise, stress cues should be seen only on the first vowel as in nonfusing higher-to-lower sequences. Preliminary data indicate that syllable fusion does occur.

5pSC25. Boundary tones as gestures: Coordination relations at boundaries. Argyro Katsika (Dept. of Linguist., Yale Univ., 370 Temple St., New Haven, CT 06520-8366, argyro.katsika@yale.edu)

While the temporal properties of prosodic boundaries are fairly well understood, tonal events at boundaries (phrase accents and boundary tones) have received less attention. The present study, conducted within the articulatory phonology framework [cf. Browman and Goldstein (1989)], is a first investigation of the coordination of boundary tone gestures to constriction gestures. An articulatory magnetometer study probes this coordination in a variety of constructions in Greek (causatives, yes-no questions, imperative requests, and wh-questions). The effects of stress (levels: zero, one or two syllables before the boundary) and accent (levels: accented versus deaccented utterance-final word) are examined. Results from two speakers analyzed to date indicate that the boundary tone coordinates with the onset's release gesture or the nucleus' opening gesture of the utterance-final syllable. The presence of accent does not affect these coordination relations, while stress does. Specifically, the latter affects the lag between the onset of the consonant gesture and the onset of the vowel gesture, and the lag between the onset of the consonant gesture and the onset of the boundary tone gesture. Such findings imply that boundary tones, albeit postlexical events, affect the intrasyllabic gestural coordination. An alternative possibility is that tonal and temporal events at prosodic boundaries interact. [Work supported by NIH.]

FRIDAY AFTERNOON, 27 MAY 2011 METROPOLITAN A, 1:00 TO 4:50 P.M.

Session 5pSP

Signal Processing in Acoustics, Engineering Acoustics, Underwater Acoustics, and Animal Bioacoustics: Detection and Classification of Buried and Proud Targets II

Jason E. Summers, Cochair
Applied Research in Acoustics, 1222 4th St., SW, Washington, D.C. 20024

Patrick J. Loughlin, Cochair
Univ. of Pittsburgh, Dept. of Bioengineering, Pittsburgh, PA 15261

Chair's Introduction—1:00

Invited Papers

1:05

5pSP1. Low frequency scattering from elastic objects embedded in a waveguide by the finite element technique. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037)

Computation of scattering from an object in a waveguide is a challenging task because of the enormous size of the computational domain. It requires the solution of the wave equation by simultaneously satisfying the boundary conditions on the surface of the object and the boundaries of the waveguide. While the finite element solution of scattering from an object in free space can be managed by limiting the size of the computational domain by the use of infinite elements or perfectly matched layers for an object in a waveguide, the waveguide itself is part of the computational domain. Since an accurate solution requires that the computational domain be discretized at least at one-tenth of the acoustic wavelength for a computational domain tens or hundreds of wavelength large, this results in very large systems of equations, whose solution is a daunting numerical task. To cope with the numerical complexity, in this paper we use the principles of fluid-structure interaction to compute scattering from the object using the finite element technique and couple it to the waveguide using the Helmholtz–Kirchhoff integral, where acoustic propagation models are used to compute the environment Green's functions. The technique is applied to buried and proud objects.

1:25

5pSP2. Acoustic scattering from underwater munitions near a water-sediment interface. Steven G. Kargl, Kevin L. Williams, Aubrey L. España (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, kargl@apl.washington.edu), Jermaine L. Kennedy, Timothy T. Marston, Joseph L. Lopes, and Raymond Lim (Panama City Div., Panama City, FL 32407-7001)

Monostatic and bistatic scattering measurements were conducted on a set of targets near a fresh water-sand sediment interface. The measurements were performed during March 2010 and are referred to as the Pond Experiment 2010 (PondEx10). Monostatic synthetic aperture sonar (SAS) data were collected on a rail system with a mobile tower, while a stationary sonar tower simultaneously collected bistatic SAS data. Each tower is instrumented with receivers while the sources are located only on the mobile tower. For PondEx10, 11 targets, including 6 underwater munitions, were deployed at 2 ranges from the mobile tower system. Initially, the data were processed using standard SAS techniques, and then, the data were further processed to generate acoustic templates for the target strength as a function of frequency and aspect angle. Results of the data processing from proud targets are presented. Finite element model (FEM) predictions of the scattering from an ordnance in the free field and proud on the interface are also discussed. A processing technique that separates an individual target's response from nearby targets is also briefly discussed. [Research supported by the SERDP and the ONR.]

1:45

5pSP3. Reversible line-scan and circular synthetic aperture sonar processing with applications to object classification. Timothy M Marston (110 Vernon Ave., Panama City Beach, FL, 32407, timothy.m.marston@navy.mil) and Philip L. Marston (Dept. of Phys. & Astronomy, Washington State Univ., Pullman, WA, 99164-2814))

Spectral response as a function of aspect, often called "acoustic color," is sometimes used to classify objects. The ability to utilize reversible imaging algorithms to enhance spectral features and to isolate the acoustic color plots of adjacent targets were previously demonstrated in the line-scan case [Marston *et al.*, J. Acoust. Soc. Am. **128**, 2461 (2010)]. Sometimes the resonant or ringing part of an object's response is isolated from the spatially localized background. This was verified by experiments at Washington State University with a metal cylinder where the ringing part was modeled using quantitative ray theory. An overview of multiple inversion techniques and the importance of the assumptions behind the beam forming processes involved will be given. Current research additionally concerns the application of reversible circular synthetic aperture sonar algorithms to enhance our understanding of the relationship between CSAS images of objects and particular spectral features present in color plots. An imaging approach based on the Fourier-slice theorem, and the subsequent reversal process will be given, along with examples of the reversal process being used to associate object features with specific spectral features. [Work supported by ONR.]

Contributed Papers

2:05

5pSP4. Acoustic scattering from proud and buried unexploded ordnances in a cluttered environment. Aubrey L. Espana, Kevin L. Williams, Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aespana@apl.washington.edu), and Mario Zampolli (TNO Defense Security and Safety, The Hague, Netherlands)

Details of the surrounding environment, for examples, sediment conditions and nearby clutter, influence the ability to successfully detect and classify proud or buried targets. These issues were investigated during experiments conducted in March 2010 in a fresh water pond, during which targets were placed at varying distances from each other, in proud and buried configurations within a sand sediment. This paper will focus on a subset of these experiments involving the acoustic scattering from unexploded ordnances (UXOs) in proud and fully buried configurations in which the incident grazing angle of the sonar onto the water-sediment interface is above the critical angle. Monostatic synthetic aperture sonar (SAS) data will be presented for the case of a single, isolated UXO, as well as the situation where multiple UXOs are in close proximity to each other, hence simulating a cluttered environment. To supplement the data, finite element models have been developed for the UXO with varying levels of complexity in both target specifications (shape and material composition) and general experimental setup. These simulations reveal the level of fidelity required to achieve good data-model agreement. [Research supported by the ONR.]

2:20

5pSP5. Multistatic scattering by scaled metallic targets: Experimental results and modeling. Jon R. La Follett and Raymond Lim (Naval Surface Warfare Ctr., Panama City, FL 32407)

Multistatic scattering measurements provide multiaspect target information that can potentially be used for classification. In the present work, bistatic and monostatic scattering measurements are conducted on multiple solid metal scaled targets using a previously described small scale test bed facility [Malvoso *et al.*, J. Acoust. Soc. Am. **127**, 1748 (2010)], and comparisons are made. Line-scan configurations with fixed and moving sources are used to construct monostatic and bistatic synthetic aperture sonar images for targets in the free field and proud on scaled sediment. One bistatic configuration uses a fixed source and two receivers that are scanned along parallel lines. The alternate bistatic configuration uses a translating source-receiver and a fixed receiver to obtain scattering information in quasi-forward directions while simultaneously acquiring monostatic measurements. Comparing monostatic and bistatic results demonstrate that, for certain cylinder orientations, bistatic scattering is dominated by specular reflection while monostatic scattering collected along the same receiver scan line is mostly due to elastic target responses. Several elastic scattering responses can be interpreted using ray models involving leaky-Rayleigh waves. Data-model comparisons are also made with T-matrix calculations for solid aluminum cylinders and stainless steel spheres. [Research supported by ONR.]

2:35

5pSP6. Mode properties and coupling conditions for scattering by circular cylinders: Comparison of liquid-filled shells and solid cylinders. Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA 99164-2814)

Simple circular cylinders are sometimes used as proud test objects when developing and testing sonar systems in realistic environments [Williams *et al.*, J. Acoust. Soc. Am. **127**, 3356–3371 (2010)]. In endeavors of this type it remains important to have a good physical understanding of the coupling of sound to the modes of infinite and finite tilted and broadside circular cylinders. The present talk reviews some experimental and theoretical investigations of the free field scattering properties including some results of quantitative ray theory for scattering amplitudes. Plotting the frequency response as a function of the tilt angle of a cylinder reveals coupling loci normally associated with axial modes of the cylinder. In certain limiting cases the coupling loci for helical modes of solid metallic cylinders resemble the

coupling loci for liquid-filled shells having appropriate boundary conditions. It is also possible for internal modal properties of long liquid-filled metallic shells to modulate the backscattering amplitude of meridional ray features. [Work supported by the ONR.]

2:50

5pSP7. Scattering by square and rectangular metallic cylinders: Circular synthetic aperture sonar images and spectral coupling loci. Anthony R. Smith, Daniel S. Plotnick, Timothy M. Marston, and Philip L. Marston (Dept. Phys. and Astronomy, Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Reversible circular synthetic aperture sonar (CSAS) is one approach to displaying and studying the evolution of monostatic sonar data with changes in the orientation of the viewed object. The present investigation concerns the evolution of backscattering by a solid slender square brass cylinder and a rectangular brass cylinder, both objects being examples of a horizontal rectangular parallelepiped elastic waveguide. In both cases, substantial elastic glints are visible in CSAS images for orientations suitable for generating surface guided elastic waves. The square cylinder is also useful for examining the effect of changing the azimuthal orientation relative to the cylinder's axis and the direction of the horizontal illumination. While rotating the azimuthal angle by 45 deg greatly modifies the magnitude of the backscattering, some of the frequency domain coupling loci (functions of rotation of the cylinder's axis about a vertical axis) are in common for both orientations. After azimuthal rotation, an apex line of the cylinder is viewed by the transducer instead of the cylinder's long flat side. This kind of spectral investigation is relevant to characterizing the coupling loci for elastic waveguides lacking the rotational symmetry of a circular cylinder. Additional insight is obtained from the CSAS images. [Work supported by ONR.]

3:05—3:20 Break

3:20

5pSP8. Finite cylinder mode identification in scattering using finite elements, Fourier decomposition, and background subtraction. David B. Thiessen, Timothy M. Marston, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, thiessen@wsu.edu)

Finite elements are widely used for calculating the acoustic scattering by objects for which a partial wave series solution is not available. When interpreting the results of such calculations for the purpose of developing signal processing methods, it can be helpful to identify the elastic modes of the object excited by the incident sound. One approach to this is examined here for a finite metallic circular cylinder. Since the object is rotationally symmetric it is convenient to use a Fourier basis to speed up the calculations [Zampolli *et al.*, J. Acoust. Soc. Am. **122**, 1472–1485 (2007)]. This also has the advantage of using the scattering amplitude associated with a given Fourier index to extract mode information. It is also helpful to subtract from the computed complex amplitude the corresponding scattering by a rigid or nearly rigid object having the same size and shape as the cylinder. For some situations this can be compared with quantitative ray theory and experimental results using reversible synthetic aperture sonar processing. [Work supported by ONR.]

3:35

5pSP9. Reversible bistatic and monostatic synthetic aperture sonar filtering. Grant C. Eastland, Timothy M. Marston, and Philip L. Marston (Dept. Phys. and Astronomy, Washington State Univ., Pullman, WA)

Recent investigations into understanding scattering features of proud and partially exposed cylinders have revealed many interesting properties of scattering mechanisms. In the present study, the effect of cylinder location was explored by lowering a horizontal solid aluminum cylinder through a flat free surface into a water tank while performing a vertical transducer line scan to monitor the evolution of the scattering. Interactions with the air-water interface partially simulate various aspects of interactions with flat sediment. The presence of the interface contributes many more paths for scattering to occur, even in the case of broadside illumination considered. Reversible SAS filtering of monostatic and bistatic data is able to identify different specular mechanisms involving the target and free surface, in addition to their corresponding elastic responses. Studying how these contribute to SAS images enables a clearer understanding of target classification and identification. It is also helpful to view the evolution of signal time-domain loci as a function of transducer location and target location. This investigation emphasizes broadside and near-broadside illumination. A delayed multiple scattering feature is visible involving one interfacial reflection and double reflected from the target. [Work supported by ONR.]

3:50

5pSP10. Boundary enhancement of a solid cylinder backscattering response: Experiments and modeling. Jon R. La Follett (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL 32407) and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

The presence of a reflecting boundary modifies the free-field backscattering response of an object. Boundary reflections increase the number of available paths for scattered sound to return to a source-receiver. In the present work, high-frequency tank experiments on solid aluminum cylinders demonstrate a strong boundary related backscattering response that is present for a broad range of aspect angles. To study the effects of close proximity to a flat reflecting boundary, cylinders were suspended in water near the tank's air-water interface. Targets were insonified from below at grazing incidence. Monostatic measurements were made as the cylinders were rotated in a plane parallel to the boundary. Some aspects of this response could be understood by modeling the region of water bounded by the cylinder top and the air-water interface as a quasi-2-D waveguide. Sound incident on this region of decreasing water thickness can be reflected as from a seamount. The backscattered amplitude is strongly affected by the threshold frequency for propagation in the gap between the cylinder and the adjacent flat surface. Experiments show that the behavior of this response, as a function of target depth and aspect angle, is in qualitative agreement with this waveguide model. [Work supported by ONR.]

4:05

5pSP11. Comparisons of evanescent wave and propagating wave scattering based on circular synthetic aperture sonar and spectroscopy. Daniel S. Plotnick, Timothy M. Marston, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

A convenient way to generate acoustic evanescent waves is to use a tank containing immiscible liquids and an appropriately directed acoustic beam [Osterhoudt *et al.*, IEEE J. Oceanic Eng. **33**, 397–404 (2008)]. This is useful for exploring the nature of scattering induced by evanescent acoustic waves by suspending an object (or set of objects) close to the horizontal interface between the two liquids. In the present work the scattering is studied for a pair of metal spheres and other metallic objects using reversible circular synthetic aperture sonar in which the object (or set of objects) is spun around a vertical axis while the scattering is measured. In some cases it is also possible to investigate spectral features and make comparisons with measurements using ordinary propagating waves or the interference pattern produced by sets of such incident waves. These experiments are useful for gaining insight into scattering phenomena and signal processing options. [Work supported by ONR.]

4:20

5pSP12. A model for under-ice-target acoustic scattering. Garner C. Bishop (Naval Undersea Warfare Ctr., Code 8211, Bldg. 1371, 1176 Howell Ave., Newport, RI 02841)

A model is described that has been developed to calculate the scatter of a high-frequency acoustic pulse that originates from a stationary source and is incident on and is scattered from the under-ice surface characteristic of the Arctic and from a stationary target. The under-ice surface is modeled by an ensemble of ice keels and ice keels are modeled by an ensemble of ice blocks with rectangular facets. The Helmholtz–Kirchhoff integral in the Kirchhoff approximation is used to calculate the scatter from an ice facet. The scatter from an ice keel is given by a coherent sum of the scatter from all the ice facets. A T-matrix is used to calculate the scatter from the stationary target. The first order terms in the multiple scattering process between the target and the ice surface are also calculated, i.e., the scattering from the target of the free field scatter from ice facets and the scatter from ice facets of the free field scatter of the target. Numerical results show the extent to which the scatter from the target in the under-ice environment differs from its free field scatter.

4:35

5pSP13. Characterization of non-Gaussian, bistatic, echoes from a shipwreck. John R. Preston (Appl. Res. Lab., The Penn State Univ., P. O. Box 30, MS 6110D, State College, PA 16804)

Sonar clutter is one of the primary limitations to active ASW. This work focuses on statistical analysis of clutterlike echoes from some bistatic measurements. Non-Gaussian characterizations of bistatic echoes from a shipwreck are presented. The received data are taken from the five octave research array (FORA) that has been used to collect extensive monostatic and bistatic data in two recent sea trials on the Malta Plateau off Sicily called Clutter 07 and BASE 07. This work uses data from the above mentioned sea trials to characterize non-Gaussian behavior of bistatic echoes from a shipwreck using pulsed sources in the 800–3500 Hz band. The page test is used to isolate the shipwreck echoes before processing. K-distributions with their shape and scale parameters are used to describe non-Gaussian behavior together with the models of Abraham and Lyons to infer physical descriptors from the echoes. The ability to georeference key statistical measures of clutter allows CFAR processors to adaptively set thresholds and reduce false alarms. Examples are shown to demonstrate this. Also included are presentations of the shape parameter versus bistatic aspect angle. [Work supported by ONR Code No. 321US].

ACOUSTICAL SOCIETY OF AMERICA

Special Workshop on
Acoustic Challenges in Aquatic Ecosystem Assessment

25 May 2011 • Sheraton Seattle Hotel
26–27 May 2011 • Hyatt Hotel

TECHNICAL PROGRAM

Wednesday, 25 May
Sheraton Seattle Hotel, Metropolitan Ballroom Foyer 3rd Floor

Exhibit Opening Reception — 5:30 p.m. to 7:00 p.m.
All workshop registrants and ASA meeting attendees are welcome

Thursday, 26 May
Hyatt Hotel, 721 Pine Street
Open to Workshop Registrants only

4aFW	Resource Monitoring, Assessment, and Management I
4pFWa	Resource Monitoring, Assessment, and Management II
4pFWb	Resource Monitoring, Assessment, and Management III (Posters)

Friday, 27 May
Hyatt Hotel, 721 Pine Street
Open to Workshop Registrants only

5aFWa	Technological Applications and Innovations I
5aFWb	Technological Applications and Innovations II (Posters)
5pFW	Technological Applications and Innovations III

Organizing Committee
John K. Horne, University of Washington, Seattle, WA
Robert McClure, BioSonics Inc., Seattle, WA
Martin Siderius, Portland State University, Portland, OR

Session 4aFW

Fisheries Workshop: Resource Monitoring, Assessment, and Management I

John K. Horne, Chair
Univ. of Washington, School of Aquatic and Fishery Sciences, P.O. Box 355020, Seattle, WA 98195

Invited Papers

8:30

4aFW1. Acoustics in support of policy. John Simmonds (European Commission, JRC Ispra VA 27021, Italy, ejsimmonds@gmail.com)

Policy in the United States and Europe requires development of targets and monitoring regimes for ocean exploitation. The current policies require a precautionary approach (PA)to exploitation, maximum sustainable yield (MSY) targets, and exploitation under ecosystem approach to fisheries management (EAFM). Currently underwater acoustics stock estimation is used as a major input for a number of fish stock assessments, and occasionally used directly to assess the state of a small number of stocks. This paper links policy requirements with management needs and explores how acoustics can deliver crucial information for environmental protection and safe exploitation. The current uses of acoustics in fisheries management are briefly reviewed, and the future requirements and uses required by these policies are explored.

Contributed Papers

9:30

4aFW2. Acoustic-derived indicators for marine reserve zoning, assessments, and monitoring in coral reef ecosystems. J. Christopher Taylor, John Burke, Shay Viehman, and Erik Ebert (Ctr. for Coastal Fisheries and Habitat Res., NOAA Ocean Service, 101 Pivers Island Rd., Beaufort, NC 28516, chris.taylor@noaa.gov)

The Tortugas Ecological Reserve was established in 2001 as a no-take research reserve extending the area of the Florida Keys National Marine Sanctuary. The Sanctuary is evaluating the effects of management areas, including the TER. In 2008 we initiated a fisheries hydroacoustic survey to compliment the visual census for reef fishes and coral communities that was initiated in 2000. The hydroacoustic survey was specifically designed to map the distribution and abundance of reef fishes in relation to the reserve boundary, and over a broader spatial extent that is possible using scuba divers. Biomass of large, exploited species observed by divers was significantly higher within the reserve, and biomass increased with proximity to the boundary. Fish densities assessed using hydroacoustics showed a similar pattern. We developed an indicator from the acoustic data analogous to the biomass size spectrum and found that biomass of larger size classes was higher within the reserve. Indicators produced from fisheries hydroacoustic surveys cannot only inform marine reserve design and monitoring, but can also enhance the interpretation of smaller-scale visual surveys by providing landscape-scale maps of fish densities at high spatial resolution.

9:50

4aFW3. Multiple frequency acoustic backscatter discrimination of zooplankton and fish scatterers in Cape Cod Bay in Spring 2010. Joseph D. Warren and Melissa Patrician (School of Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, joe.warren@stonybrook.edu)

During the late winter and early spring, Cape Cod Bay has large numbers of marine mammal and seabird predators that feed on aggregations of zooplankton (copepods) and fish (sand lance and herring). As part of a study investigating the foraging behavior of these top predators, we sampled the waters of Cape Cod Bay using multiple frequency (38, 120, 200, and 710 kHz) echosounders and a 600 kHz acoustic Doppler current profiler. To ground truth our data net tows and video plankton recorder casts were used to sample zooplankton, primarily small copepods, while a drop camera system was used to identify aggregations of fish. Combining the backscatter data with target strength estimates for the different scatterer types allowed us to estimate the identity, distribution, and abundance (numerical density) of zooplankton and fish in this area at spatial and temporal scales relevant to the foraging behavior of baleen whales. This method worked best when scattering aggregations were monospecific and target scattering spectra were distinct.

10:10—10:40 Break

10:40

4aFW4. Acoustic-trawl surveys to assess walleye pollock in Alaska: challenges faced and progress made. Christopher D. Wilson (NOAA Fisheries, Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Bldg. 4, Seattle WA 98155)

Large-scale acoustic-trawl surveys have been regularly conducted for over 3 decades by researchers at the NOAA, Alaska Fisheries Science Center (AFSC) to assess walleye pollock (Theragra chalcogramma) in Alaska. Research within the AFSC acoustics program (MACE) has addressed several of the challenges associated with both the acoustics and trawling aspects of these surveys. Acoustic issues include species classification of the acoustic data, fish avoidance to survey vessels, and *in situ* target strength measurements. Large trawls typically form an integral part of acoustic surveys to confirm the species identity of backscatter and to collect other biological information needed to convert the echo integral into numbers and weight of animals per unit area. MACE staff have also made significant progress in evaluating the selectivity of the large midwater trawls used during their acoustic surveys as well as in developing other direct sampling tools to help interpret the acoustic backscatter data. This presentation will highlight these and other recent advanced technologies and methods that are being developed and used by the MACE program to improve acoustic-trawl surveys in Alaska and elsewhere.

11:00

4aFW5. Acoustic biomass estimation and uncertainty of Pacific hake and Humboldt squid in the Northern California current in 2009. Rebecca Thomas, Ian Stewart, Dezhang Chu, John Pohl (NOAA Fisheries, Northwest Fisheries Sci. Ctr., Fishery Resource Anal. and Monitoring Div., 2725 Montlake Boulevard East, Seattle, WA 98112, rebecca.thomas@noaa.gov), Ken Cooke, Chris Grandin (Fisheries and Oceans Canada, Pacific Biological Station, Nanaimo, BC, V9T 6N7, Canada), and Stephen de Blois (NOAA Fisheries, Northwest Fisheries Sci. Ctr., Fishery Resource Anal. and Monitoring Div., 2725 Montlake Boulevard East, Seattle, WA 98112)

The population of Humboldt squid (*Dosidicus gigas*) has seen an explosion in the Eastern North Pacific over the last several years. The species has gone from being rarely seen in the waters off OR, WA, and BC to becoming a major predator in the marine food web in this area. This population explosion has the potential to cause large impacts in major fish stocks. The biennial 2009 Joint U.S.-Canada Pacific hake (*Merluccius productus*) acoustic trawl survey also noted large amounts of Humboldt over much of the survey area. Because Humboldt squid could be acoustically confused with Pacific hake, and because the presence of Humboldt squid disrupted the normal shoaling pattern of hake, an estimated depth threshold was used to help distinguish Humboldt squid from hake. Accordingly, the biomass estimate of Pacific hake for 2009 was less certain. Several methods were explored to quantify the uncertainty and assess the reliability of the hake biomass estimate.

11:20

4aFW6. Acoustic data from fishing vessels: What can be obtained that research vessels cannot offer? Francois Gerlotto (I.R.D., Ave. Jean Monnet, B.P. 171, 34203 Ste Cedex, France, francois.gerlotto@ird.fr), Mariano Gutierrez (TASA, Lima, Peru), Erwan Josse Josse (UMR LEMAR, Brest, France), and Ronan Fablet (IRD, Lima, Peru)

Use of fishing vessels and platforms of opportunity to collect acoustic-based population distribution and abundance data has been considered since the 1990s. The idea was that low data quality was compensated by large sampling throughout the year, and in areas where research vessels did not sample. We present results and hypotheses on the use of such data from the anchovy and jack mackerel fishing industry in Peru, where many fishing vessels are equipped with SIMRAD ES60 echosounders. First, fishing vessels are selected according to a criteria set. Once "optimal" fishing vessels are selected, we show that exploiting acoustic data from these ships may provide new information. The first is a correlation of fishing activity with fish biomass. Ee analyzed echograms preceding trawl sets. Data on fish biomass and spatial distribution were correlated with catch and catch structure. Additionally, environmental information in an area of 1 nm^2 centered on the trawl location was added to the acoustic and fishing data. The potential to obtain information on fish length within the exploited school before fishing was explored, using direct (TS measurements around the school) or indirect (environmental signature giving probabilistic information) methods.

11:40

4aFW7. Introduction of multi-frequency acoustics and mid-water trawl sampling to the CALCOFI program. Ana Lara Lopez, Peter Davison, and J. Anthony Koslow (Scripps Inst. Ocean, Univ. Cal San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0218, alaralopez@ucsd.edu)

The CALCOFI program has evolved over time, continually expanding its interdisciplinary study of the southern California Current ecosystem. Historically, CALCOFI comprised studies of physical and chemical oceanography, phytoplankton, zooplankton, ichthyoplankton, marine mammals, and seabirds. However, there was little focus on the mid-trophic level and the mesopelagic micronekton. In 2010, multi-frequency acoustics, combined with mid-water trawl sampling, was introduced. Data were collected with an EK-60 equipped with five frequencies (18, 38, 70, 120, and 200 kHz) and a Matsuda–Oozeki–-Hu Trawl (MOHT). Frequency spectrums were generated for all organisms caught in the net using target strength models and categorized into different acoustic groups. Inverse techniques and dB differencing were subsequently used to estimate their abundance and distribution across the CALCOFI grid. [Work supported by NOAA and The Moore Foundation.]

THURSDAY AFTERNOON, 26 MAY 2011 — HYATT AMPHITHEATRE, 1:00 TO 5:00 P.M.

Session 4pFWa

Fisheries Workshop: Resource Monitoring, Assessment, and Management II

Orest I. Diachok, Chair
Johns Hopkins Univ., Applied Physics Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099

Contributed Papers

1:00

4pFWa1. Evaluating freshwater habitat restoration with active acoustics. Laura E. Madden (School of Forest Resources and Appl. Res. Lab., The Penn State Univ., 435 Forest Resources Bldg., University Park, PA 16802, lem230@psu.edu) and Jennifer L. Miksis-Olds (The Penn State Univ., State College, PA 16804)

The effectiveness of adding a submerged physical structure in order to increase fishery production is uncertain. Measuring fishery response to these alterations with conventional techniques is difficult. Electrofishing is a typical assessment method in freshwater fishery management and is often limited in sample size and sampling frequency. This study used active acoustic technology to evaluate the distribution and behavior of fish assemblages associated with added submerged rock structures in a reservoir currently undergoing habitat improvement. An acoustic water column profiler was deployed for three 1-week intervals at each of three replicate sites consisting of adjacent treatment areas with added rock structures and control areas without added structures. Electrofishing was conducted during each sampling interval. Fish abundance and behavior at each site were assessed from the volume backscatter time series and electrofishing data. Differences between areas with and without structures were compared. Combining acoustic technology with conventional assessment methods has enabled a more thorough evaluation of habitat restoration projects and helps guide the development of future conservation efforts.

1:20

4pFWa2. Comparison of DIDSON sonar based estimates of Chinook salmon escapement with other methods in the Coweeman River, Washington. Dan Rawding and Martin Liermann (Wash. Dept. of Fish and Wildlife, 600 Capitol Way N., Olympia, WA 98501-1091)

The Coweeman River supports a small spawning population of Tule fall Chinook salmon (200–2000 adults), which are listed for protection under the US Endangered Species Act but harvested in mixed stock Pacific Ocean and mainstem Columbia River fisheries. The Coweeman Chinook stock has a low proportion of hatchery spawners and has been used as an index population for other less intensively monitored Tule populations. Over the last decade the Washington Department of Fish and Wildlife (WDFW) and partners have funded numerous studies to determine effective methods to accurately estimate Chinook salmon escapement for this population. From 2007 to 2009, a DIDSON sonar was operated below the major spawning area to estimate escapement. Concurrently, independent escapement estimates were made using more traditional methods such as mark-recapture, red counts, and area-under-the curve surveys. Bayesian approaches to estimate salmon escapement for all methods will be presented, with an emphasis on the use of mixture models for sonar-based estimates and a comparison of the sonar estimates with more traditional approaches.

1:40

4pFWa3. Observations of 5-MHz acoustic backscattering from *Cochlodinium polykrikoides* blooms in coastal waters. Jee Woong Choi, Eunhye Kim, Jungyul Na (Dept. of Environ. Marine Sci., Hanyang Univ., Ansan 426-791, Korea), and Donhyug Kang (Korea Ocean Res. and Development Inst., Korea)

Measurements of 5-MHz volume-backscattering strength of harmful algal blooms were made in summer 2008 at sites off the coast of Geumo island in southern Korea. The most abundant phytoplankton in the blooms was the chain-forming, toxic dinoflagellate *Cochlodinium polykrikoides*, which accounted for about 70% of total abundance. The measured backscattering strengths are compared with the abundance of phytoplankton estimated using an optical microscopy from water samples collected simultaneously with the acoustic measurements. The results show that the scattering strengths are highly correlated with dinoflagellate concentrations, suggesting that the acoustic method may be a useful tool for detecting the harmful algal blooms in their initial stage. Finally, the measurements are compared with the scattering strengths predicted by a fluid-sphere scattering model using a single-cell size of *C. polykrikoides* and the equivalent cell size of a sphere the same size as the chain-forming cells. The results imply that the chain-forming cells should be considered as single body with an equivalent cell size. [Work supported by the Korea Ocean Research and Development Institute.]

2:00—3:10 Break and Posters

3:10

4pFWa4. Assessing rockfish abundance in complex habitats using acoustics and cameras. Darin T. Jones (NMFS/AFSC, 7600 Sand Point Way NE, Bldg. 4, Seattle, WA 98115, darin.jones@noaa.gov), Thomas C. Weber (UNH-CCOM, Durham, NH 03824), Christopher N. Rooper (NMFS/AFSC, Seattle, WA 98115), John L. Butler (NMFS/SWFSC, La Jolla, CA 92037), Christopher D. Wilson, and Alex De Robertis (NMFS/AFSC, Seattle, WA 98115)

Many species of rockfishes (*Sebastes* spp.) are difficult to assess using trawl surveys due to their propensity to aggregate near the seafloor in rocky high relief areas. A feasibility study was conducted during October 2009 in such an area south of Kodiak Island, AK, to evaluate the use of standard fisheries acoustic survey methods in conjunction with stereo-video cameras for estimating the distribution and abundance of dusky and northern rockfishes. Uniformly spaced parallel transects were repeatedly surveyed using single beam echosounders over several days. A multibeam echosounder was used to characterize the seafloor as trawlable or untrawlable and these designations were corroborated by camera. Rockfish abundance was estimated using a combination of acoustic and camera measurements. At least 80% of the rockfish detections were observed in untrawlable habitat areas, and within 2.0 m of the seafloor. Over half of the rockfish seen by the camera were within the acoustic dead zone. Repeat passes exhibited high precision and there was no significant difference in fish abundance or height off bottom between night and day. Future work is planned during summer 2011 to evaluate the feasibility of using these methods in broader areas and for other rockfishes in the Gulf of Alaska.

3:30

4pFWa5. Acoustic sea bed classification of Pacific Sand Lance habitat. Ben R. Biffard, N. Ross Chapman, Stephen Bloomer (Marine Acoust. Remote Sensing Facility, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada), and Cliff Robinson (Parks Canada Agency, Vancouver, BC V6B6B4, Canada)

This paper describes the results of preliminary acoustic sea bed classification surveys in three areas in the southern Gulf Islands of British Columbia to develop methods for mapping habitat of Pacific Sand Lance (PSL). Little is known about this important species, and much less is known about their use of subtidal burying habitat for overwintering. Grid surveys were run using a dual-frequency (24 and 200 kHz) single beam echosounder, and the data were classified using conventional statistical segmentation procedures using QTC IMPACT. This type of seabed classification separates the seabed into self-consistent regions called seabed classes based on acoustic diversity—primarily on the basis of echo shape. Sediment grab samples and video were also obtained to provide ground-truth and labels for the acoustic seabed classes. The acoustic seabed classification surveys were successful in identifying subtidal sands with low fractions of fines (silts) and sand wave fields among a variety of seabed types identified. Grab samples captured PSL buried in the medium to coarse subtidal sands, mostly in sand wave fields, and mostly in winter. Future work will be to observe diel migrations and develop methods to estimate biomass of buried PSL based on seabed classification.

3:50

4pFWa6. Acoustic observations of the deep scattering layer during the Deepwater Horizon oil spill. Alex De Robertis (Natl. Oceanic and Atmospheric Administration, Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Seattle, WA 98115, alex.derobertis@noaa.gov), Thomas C. Weber, Larry Mayer (Univ. of New Hampshire Durham, NH 03824), and Christopher D. Wilson (Alaska Fisheries Sci. Ctr., Seattle, WA 98115)

The explosion of the Deepwater Horizon drilling rig on April 20, 2010 resulted in the release of large quantities of oil and gas from the damaged wellhead into the deep waters of the Gulf of Mexico. During the monitoring effort that ensued, a large body of acoustic measurements with scientific echosounders was collected from May to October with the goal of mapping subsurface oil and gas and monitoring the integrity of the well head. These measurements will be used to observe the deep scattering layer (DSL), a ubiquitous community of sound-scattering mesopelagic organisms in the vicinity of the spill site. Preliminary observations of reduced backscatter in the near-field of the rising oil indicate that the DSL is perturbed by the rising oil close to the well head. It is unclear whether this is a highly localized effect occurring only near the well, or whether the DSL was also perturbed by a deep oil plume that spreads from the well site. The acoustic measurements of the DSL will be related to fluorometric indices of hydrocarbons from CTD profiles to gauge if there was a larger-scale perturbation on the abundance, behavior, and distribution of the midwater community.

4:10

4pFWa7. Influence of biophysical coupling on age-0 pollock survey results. Sandra Parker-Stetter (Univ. of Washington, School of Aquatic and Fishery Sci., Box 355020, Seattle, WA, 98195-5020, slps@uw.edu), John Horne (Univ. of Washington, Seattle, WA, 98195-5020), Edward Farley, and Lisa Eisner (Alaska Fisheries Sci. Ctr., Auke Bay Lab., Natl. Marine Fisheries Service, Juneau, AK 99801-8626)

Age-0 walleye pollock relative abundance and distribution in the eastern Bering Sea has been characterized with data from a surface trawl survey [Bering Aleutian Salmon International Survey (BASIS)]. Based on surface trawl catches since 2003, age-0 pollock abundances appeared to be highest during climatic warm years and lowest in cold years, with predictable distributions based on water column stratification. In 2008–2010, all cold years, acoustics, and midwater trawling were added to the BASIS survey. Acoustics confirmed that low numbers of age-0 pollock were found in the surface waters, but revealed the presence of previously unsampled, large (over 50 m high, several km long) aggregations in deep (greater than 100-m bottom depth) water. Acoustic and midwater trawling results suggest that the deep biomass may exceed that found in surface waters. Relationships are now being evaluated between oceanographic characteristics (water column stability, extent of the bottom cold pool) and biomass of age-0 pollock in surface and deep aggregations. Clarifying these biophysical linkages is important for evaluating potential gear-specific biases in survey data and for informing survey design decisions under changing climate conditions.

4:30—5:00 Panel Discussion

Joe Warren, Moderator

Session 4pFWb

Fisheries Workshop: Resource Monitoring, Assessment, and Management III (Poster Session)

Jules S. Jaffe, Chair
Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:10 p.m. to 2:40 p.m. and contributors of even-numbered papers will be at their posters from 2:40 p.m. to 3:10 p.m.

4pFWb1. Estimating total uncertainty for abundance at age estimates from acoustic-trawl surveys. Paul D. Walline and Mathieu Woillez (Natl. Marine Fisheries Service, Alaska Fisheries Sci. Ctr., 7600 Sandpoint Way NE, Bldg. 4, Seattle, Washington 98115, paul.walline@noaa.gov)

A comprehensive quantitative treatment of acoustic-trawl survey uncertainty that includes all major components of measurement and sampling error is needed to help specify the level of risk in stock assessment models. Acoustic and trawl sampling errors can be addressed by geostatistical simulations, which account for autocorrelation, non-independence, and non-random sampling, while additional sources of uncertainty, such as ship motion and instrument errors, can be incorporated into uncertainty estimates by bootstrapping. A conceptual model is presented in the form of a flow diagram, which identifies sources of uncertainty and recommends analytical methods to quantitatively assess the uncertainty associated with abundance at size estimates. Sequential geostatistical simulations of the spatial distribution of backscatter attributed to walleye pollock from Eastern Bering Sea acoustic-trawl surveys show that the acoustic sampling error is one of the two largest sources of uncertainty in estimates of total abundance. Sampling error associated with the spatial distribution of fish size is the next largest uncertainty source, and was simulated using co-Kriging for spatial functional data, which extends the simulation method to allow estimation of uncertainty by age-class.

4pFWb2. Accounting for uncertainty in acoustic estimates. Patrick Sullivan and Lars Rudstam (Dept. of Natural Resources, Cornell Univ., Ithaca, NY 14853, pjs31@cornell.edu)

The process of converting raw acoustic signals into estimates of fish abundance can be noisy and complicated. Is the method we use to calculate mean abundances the best that is statistically achievable? If we are to put effort into increasing sample size or refining estimates, where should we focus our attention? Once we understand the uncertainty associated with a given step in the conversion process, how do we include that uncertainty in characterizing the final estimates? In this paper, we develop a mechanism that is part Monte Carlo and part Bayesian to address known sources of uncertainty in the acoustic assessment process. Estimation steps are followed sequentially from Sv to abundance and the variances are carried along through each step. The structure of the analysis facilitates examination of the effects of the various parts of the analysis on the quality of the final outputs. Acoustic scientists can use this approach to help prioritize their own research and assessment designs.

4pFWb3. Uncertainty in hydroacoustic-derived fisheries data: An analysis of simultaneous hydroacoustic-trawl surveys. Paul W. Simonin, Samuel A. Tideman (Dept. of Natural Resources, Cornell Univ., Tower Rd., Ithaca, NY 14853, pws44@cornell.edu), Bernard Pientka (Vermont Dept. of Fish and Wildlife, Junction, VT 05402), Lars G. Rudstam, Patrick J. Sullivan (Cornell Univ., Ithaca, NY 14853), and Donna L. Parrish (Univ. of Vermont, Burlington, VT 05405)

Fish biomass estimates from hydroacoustic surveys contain uncertainty that must be quantified before such estimates are of use. To better understand uncertainty associated with down-looking hydroacoustics, analyzes were conducted on simultaneously collected trawl and hydroacoustic data from over 200 sampling events in Lake Champlain, a mesotrophic freshwater lake located between New York, Vermont, and Quebec (Canada). A model set, which used a number of factors to explore sources of variance, was analyzed. Factors explored were acoustic density estimation method, sampling time, sampling season, water temperature, light, invertebrate presence, fish size, fish species, and collection depth. Trawl avoidance was higher during the day, and varied with trawl type. Sources of variance in hydroacoustic estimates were the size and depth of analysis regions, presence of invertebrates, and fish shoaling behavior. All fish species could not be differentiated using acoustic target strength, thus necessitating trawl collections to estimate species-specific abundance. Trawl collection showed vertical separation of adult pelagic species in Lake Champlain, and suggested that side- or up-looking hydroacoustic and surface gillnet surveys may be necessary to reduce uncertainty in abundance estimates. Surveys using multiple sampling methods likely result in abundance estimates closest to reality.

4pFWb4. Estimation of the take of fish in a riverine environment by sound generated by confined explosions. Thomas J. Carlson, Gary E. Johnson (Pacific Northwest Natl. Lab., Portland, OR 97204-1423), and Christa M. Woodley (Marine Sci. Lab., Sequim, WA 982382)

The Columbia River Navigation Channel Improvement Project required removal of a large shelf of basalt rock from the Columbia River near Portland, OR. Blasting was conducted over a 2 month period to fracture the rock for removal. Nearly 100 blast events, each consisting of up to 130 individual charges, were detonated. Monitoring was conducted to estimate the take of adult and juvenile salmonids and sturgeon. The number and distribution of fish exposed to blast pressure were estimated by applying active acoustic methods to observe the flux of fish into a region surrounding blast events. Blast pressures were measured and a propagation model used to estimate the level of blast pressure through the region monitored for take. A likelihood model was fitted to data for the response of adult fish to impulsive sound exposure and used to estimate the take of fish given the estimated distribution and exposure of fish through the river volume impacted by blast events. The explosive charges, which were located at depth in the massive rock formation and further confined by stemming in addition to time delayed detonation of individual charges, resulted in low peak pressures in the water column and negligible take of fish.

4pFWb5. Challenges in the quantitative assessment of anthropogenic sound in marine environments. Martin Siderius and Lisa Zurk (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201)

Concerns about the effects of anthropogenic sounds in aquatic environments have increased over the past 2 decades. There is particular concern with the impact to species that are endangered or those that rely on sound for foraging, navigating, and to detect prey and predators. One of the challenges to quantitative assessment is simply characterizing all environments with potentially harmful sound sources and identifying the impacted species (type and population density). In addition, there are challenges to setting the criteria for physiological and nonphysiological (i.e., behavioral) impacts.

Environments vary widely from the deep ocean where navy and commercial sonar systems mostly operate to shallow water (including rivers) where construction sites are often located and activities such as pile driving occur. There are challenges related to both the measurement and the prediction of impact in these environments. Measurements are needed to determine if sound levels and species type and density are as expected. Prediction is required to extrapolate beyond the direct measurements and to define a zone-of-influence to determine the overall impact. Acoustic analysis includes accurately describing the sound mechanisms and the sound propagation within the zone-of-influence. The focus of this presentation will be on the acoustic challenges related to quantitative assessment.

4pFWb6. Using acoustic data from fishing vessels to estimate walleye pollock abundance in the eastern Bering Sea. Taina Honkalehto, Patrick H. Ressler, Richard H. Towler, and Christopher D. Wilson (Natl. Oceanic and Atmospheric Administration, Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Seattle, WA 98115)

Eastern Bering Sea walleye pollock (*Theragra chalcogramma*) supports one of the world's largest fisheries. Due to pollock's high recruitment variability and relatively short life span, timely and accurate abundance indices are needed for fisheries management. Annual bottom trawl (BT) surveys track the demersal portion of the pollock population using chartered commercial fishing vessels, while biennial acoustic-trawl (AT) surveys use NOAA research vessels to track the younger, midwater portion of the population. More frequent, annual indices of walleye pollock stock size may be obtained using acoustic backscatter data collected from the commercial fishing vessels conducting the BT survey at relatively low cost. A retrospective analysis of 38 kHz, AT survey data (1999–2004) identified a suitable index area to track midwater pollock abundance. The BT survey acoustic data (2006–2009) collected in that area tracked both the AT survey pollock abundance and the large-scale pollock distribution patterns. This study is unique because commercial vessel acoustic data were used to estimate a new annual abundance index whose performance can be evaluated by a biennial research vessel survey. The new index will benefit managers by providing more accurate information on near-term abundance trends when dedicated research ship time is not available.

4pFWb7. Estimating Chinook salmon passage in the Kenai River using dual frequency identification sonar. Debby L. Burwen, Steven J. Fleischman, and James D. Miller (Alaska Dept. of Fish and Game, 333 Raspberry Rd., Anchorage, AK 99518, debby.burwen@alaska.gov)

Chinook salmon passage has been estimated in the Kenai River using fixed location, side-looking split-beam sonar since 1996. Estimation of Chinook salmon passage is complicated by the presence of smaller, more abundant species of salmon. Accurate estimates depend on the ability to distinguish large from small fish. Originally, a target strength threshold was used to classify fish species. However, *in situ* tethered-fish experiments revealed that target strength was a poor predictor of fish size for side-looking sonar. Measurements based on echo envelope length (duration) provided better predictive ability for tethered fish, but this could not be conclusively verified for free-swimming migratory fish. In this paper we describe our current efforts to estimate Chinook salmon passage in the Kenai River using dual frequency identification sonar (DIDSON). Length is manually measured from DIDSON still images, and from fish captured by gillnets drifted onsite. A species/age mixture model is fitted to the data to provide estimates of Chinook salmon passage. Passage estimates for large Chinook salmon can be produced using a simple threshold applied to DIDSON lengths.

4pFWb8. Characterizing and monitoring marine nekton at Puget Sound's renewable energy site. John K. Horne, Sandra L. Parker-Stetter (School of Aquatic and Fishery Sci., Univ of Washington, Box 355020, Seattle, WA 98195, jhorne@u.washington.edu), Brian Polagye (Univ. of Washington, Seattle, WA 98195), Jim Thomson (Univ. of Washington, Seattle, WA 98105), Kurt L. Fresh, and M. Brad Hanson (NOAA Northwest Fisheries Sci. Ctr., East Seattle, WA 98112)

Marine hydrokinetic energy sites have the potential to impact distributions, dynamics, and abundances of macroinvertebrates, fish, and marine mammals (i.e., nekton). Potential impacts include changes in aggregation, avoidance, and occurrences of strikes or impingements. Understanding potential impacts requires knowledge of species-specific distributions over relevant spatial and temporal scales. Configurations and integration of technologies capable of providing images and data are not well-established, and the application of monitoring technologies is complicated by extreme flows at marine hydrokinetic energy sites. An echosounder, multibeam sonar, and acoustic camera will be used to detect, categorize, and enumerate nekton at a proposed renewable energy site in northern Admiralty Inlet, Puget Sound, WA. This is the site selected by Snohomish Public Utility District for the deployment of two OpenHydro turbines. Data from stationary instrument deployments will be compared to data from a mobile acoustic and midwater trawling survey to determine how well each technology captures spatiotemporal variation in nekon density distributions. Results of the design, deployment, retrieval, and analysis of data from these three instrument classes will be used to formulate recommendations for instrument choice, configuration, and the characterization and monitoring of pelagic nekton at any renewable energy site.

4pFWb9. Bioacoustic absorption/scintillation spectroscopy based classification of fish by size and depth. Orest Diachok (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099)

Bioacoustic absorption/scintillation spectroscopy (BASS), which exploits frequency selective attenuation and scintillation due to swim bladder resonances, promises classification of fish by size (and with ancillary information and species) and depth and unbiased (by avoidance and proximity of fish to boundaries) estimates of number densities versus fish length. A BASS experiment in the Santa Barbara Channel, which employed a fixed ultra-broadband (0.3–10 kHz) source and a fixed vertical array separated by 4 km, demonstrated the power of this method. The source, which was employed during this experiment, had an environmentally friendly source level of 170 dB. Absorption lines in frequency/depth space, which were associated with year classes of physostomes, sardines, and anchovies, were evident at night when the fish were dispersed. BASS results were consistent with trawl and echo sounder data. Interpretation of absorption lines during daytime was complicated by "bubble cloud" resonances associated with fish schools. Displays of the scintillation index in frequency-depth space facilitated classification of absorption lines associated with species/year classes. Future experiments will focus on hake, a physoclist. [This work was supported by the Office of Naval Research.]

Session 5aFWa

Fisheries Workshop: Technological Applications and Innovations I

John K. Horne, Chair
Univ. of Washington, School of Aquatic and Fishery Sciences, P.O. Box 355020, Seattle, WA 98195

Invited Paper

8:30

5aFWa1. Marine Ecosystem Acoustics, a conceptual approach to enhanced process understanding and system evaluation. Olav Rune Godoe (Inst. of Marine Res., P.O. Box 1870, 5817 Bergen, Norway)

Institute of Marine Research has participated in developing marine acoustics for fisheries and ecosystem quantification since the mid 1930s. This development has motivated the recent focus on the concept, Marine Ecosystem Acoustics. This concept defines acoustics as the principal tools in ecosystem-based studies of marine organisms and their environment. Only acoustics can provide-information with appropriate resolution at spatial and temporal scales across the range of the fundamental biophysical processes as well as at the scales needed to resolve trophic interactions from individuals to populations. Exploiting acoustic band and beam widths as well as advanced platform technologies ecosystem processes can be observed on scales at which they occur, an essential requirement for quantitative ecosystem understanding and modeling. Further, the approach also supports the knowledge and information needed to establish a better fundament for an ecosystem-based fisheries management. Acoustics is presently underutilized in ecosystem research and still needs substantial development. The concept and its potential are exemplified with results from field experiment.

Contributed Papers

9:30

5aFWa2. Field applications of broadband acoustic scattering techniques for quantifying zooplankton distribution, abundance, and size. Gareth L. Lawson, Andone C. Lavery, Peter H. Wiebe, and Nancy J. Copley (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, glawson@whoi.edu)

Discriminating among sources of scattering remains a key problem in ecological applications of active acoustics. This is particularly true in studies of zooplankton, which are often found in communities of heterogeneous species composition, and in the deployment of acoustics on autonomous platforms, where independent sampling with nets to ground-truth acoustic observations is often not feasible. Broadband measurements can provide substantial improvements in species discrimination by characterizing more fully the frequency spectrum of scatterers relative to traditional single- and multi-frequency narrowband techniques. Here, we present findings from a series of three cruises to the margins of Georges Bank examining patchiness in the distribution of krill and other zooplankton, as well as interactions with predators (fish, seabirds, and marine mammals). A heavily modified commercially-available broadband system spanning a frequency band of 30–600 kHz, with some gaps, was deployed in a towed body to depths of 200 m. Concurrent measurements made with a surface-towed multi-frequency system (43, 120, 200, and 420 kHz) along with ground-truthing information from depth-stratified net sampling and a video plankton recorder allow an assessment of the strengths and limitations of broadband methods for remotely discriminating among sources of scattering and for estimating the abundance and size of animals.

9:50

5aFWa3. A new wide-band sonar for zooplankton studies. Jules Jaffe, Paul Roberts, Christian Briseno-Avena (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA 92093, jules@mpl.ucsd.edu), and Amatzia Genin (Hebrew Univ. and Interuniversity Inst. Mar. Sci., Eilat 88103, Israel)

A wide-band (1.5–2.5 MHz) multi-transducer sonar system has been developed for zooplankton studies. The system consists of multiple transmit /receivers that can be configured in either a multiple mono-static or multi-static mode. In the latter case, one transmitter and multiple receivers permit multi-angle backscatter measurements. The wide bandwidth permits resolving individual echoes separated by less than 2 mm. With an optical imager that records high resolution (25 μm), high contrast, images of animals over short exposures; 10–40 μm, the system records both acoustic reflections and optical images at 1 Hz. In multi-monostatic mode, the volume of interrogation for the sonar is 4 li, while that of the optical imaging system is 150 ml. Initial tests consisted of towing the system in multi-angle mode. During a recent cruise in September 2010, vertical profiles to 500 m revealed dense aggregates of *Calanus pacificus californicus* in the Santa Barbara Basin as recorded by the sonar and verified via the imaging system. A set of such multi-transducer systems, deployed in a coral reef environment, are being used to quantify *in situ* rates of zooplankton predation by fish and invertebrates, from the level of single predators to the whole community.

10:10–11:20 Break and Posters

11:20

5aFWa4. Exploratory measurements using a broadband, split beam echo sounder system. Egil Ona (Inst. of Marine Res., P.O. Box 1870, 5817 Bergen, Norway, egil.ona@imr.no), Lars N. Andersen, Haakon Solli (Simrad, 3191 Horten, Norway), Gavin Macaulay, Lucio Calise, Ruben Patel, Rolf Korneliussen, and Tor Knutsen (Inst. of Marine Res., 5817 Bergen, Norway)

A prototype broadband echo sounder has been used for measuring zooplankton and fish in a new collaborative project between the Institute of Marine Research and Kongsberg Maritime-Simrad. The prototype system used is a Simrad broadband system which includes a transceiver operating from 10–500 kHz. We have mainly used the transceiver together with four separate Simrad ESXX-7CD standard pressure resistant transducers, the ES70-7CD, ES120-7CD, ES200-7CD, and ES333-7CD, covering the band from 50 to 450 kHz. The research has so far concentrated on data output formats, calibration methods, and measurements from single targets in *ex situ* and *in situ* situations. Examples of recorded reflected spectra for selected calibration and biological targets will be shown and the potential for improved aquatic ecosystem assessment discussed.

11:40

5aFWa5. The advantage of FM slide signals for active acoustic systems used for fisheries studies. John E. Ehrenberg and Samuel V. Johnston (Hydroacoustic Technol. Inc., 715 NE Northlake Way, Seattle, WA 98105, jehrenberg@HTIsonar.com)

A key requirement for many active acoustics systems used for fisheries or biological oceanographic data collection is to provide high spatial resolution while maximizing the range for which useful data are obtained. Unfortunately, most systems in use today use cw pulse transmit signals for which the user must trade-off range resolution with output signal-to-noise, which determines the maximum useful range at which data can be collected. This paper describes the application of a different type of signal, the FM slide or chirp, for fisheries and oceanographic acoustic assessment. Compared to the cw pulse based systems, acoustic systems that use FM slide signals with the corresponding matched filter processing can achieve significantly better spatial resolution without sacrificing maximum range. The explanation of why the FM slide signal achieves this improved performance as well as a discussion of some of the factors that must be considered when implementing this signal type are described. Laboratory measurements and field results are presented that show the advantages of the FM slide for real world assessment studies.

12:00

5aFWa6. Standard-target calibration of sonars for marine ecosystem assessment. Kenneth G. Foote (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Remote acoustic observation and sampling of multiple trophic levels in the marine ecosystem generally require s ubstantial bandwidth, if not the coordinated use of two or more sonars. Calibration is essential. The standard-target method has a history of application, spanning the total frequency range 1–2500 kHz, with precision solid elastic spheres used as standard targets. The theory of measurement of volume backscattering and form functions of arbitrary targets is elaborated, also revealing the basis for the standard-target method. [Work supported by ONR Award No. N000140910482.]

FRIDAY MORNING, 27 MAY 2011 HYATT AMPHITHEATRE, 10:10 TO 11:20 A.M.

Session 5aFWb

Fisheries Workshop: Technological Applications and Innovations II (Poster Session)

Sandra Parker-Stetter, Chair

Univ. of Washington, School of Aquatic and Fishery Sciences, P.O. Box 355020, Seattle, WA 98195

Contributed Papers

All posters will be on display from 8:30 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 10:20 a.m. to 10:50 a.m. and contributors of even-numbered papers will be at their posters from 10:50 a.m. to 11:20 a.m.

5aFWb1. Development of real three-dimensional sonar for observing small scale dynamics of fish inside a school. Francois Gerlotto (I.R.D., Ave. Jean Monnet, B.P. 171, 34203 Ste Cedex, France, francois.gerlotto@ird.fr), Philippe Roux (CNRS, LGIT, Grenoble, France), Jean Guillard (CAARTEL, Thonon, France), Erwan Josse (LEMAR, Brest, France), Ronan Fablet, Yannick Perrot, and Patrice Brehmer (ENSTB, Brest, France)

We present concepts for building and operating a high-frequency, three-dimensional sonar (operating range 5–20 m) that will allow observation and analysis of small scale dynamics within aggregations (e.g., microgroups), which is essential to understand physical and behavioral mechanisms within schools. Such a multibeam sonar requires nonconventional concepts as traditional sonars are limited by physical constraints, making observational improvements impossible. Exploiting a new generation of multichannel electronics and array processing tools, we aim to (1) reaching an acquisition rate of up to 10 images/s, while (2) keeping a spatial resolution of $\sim$10 cm within the aggregation for a 10 m^3 water volume. There are three activities: building a prototype to observe a small group of fish inside a school and deliver spatiotemporal images of interfaces inside and outside the school; develop tools for visualization, analysis, and data collection to provide information usable in behavioral ecology; and to define system capabilities and testing the applicability of the prototype to ecological and behavioral questions on schooling and representation of schooling adaptations as indicator of interactions between a population and the environment. Experiments showed that discriminating targets within noisy backgrounds was possible and gave workable results.

5aFWb2. Predator–prey interactions and fish schooling behavior revealed through the lens of a sonar. Kevin M Boswell (Dept. Oceanogr. and Coastal Sci., Louisiana State Univ., 2243 Energy Coast and Environment Bldg., Baton Rouge, LA 70803), Nils O. Handegard (Inst. of Marine Res., P.O. Box 1870 Nordnes, 5817 Bergen, Norway), Simon Leblanc, and Iain D. Couzin (Princeton Univ., Guyot Hall, Princeton, NJ 08544)

Acquiring information about the schooling dynamics and direct predator-prey interactions *in situ* is more than a difficult challenge. Moreover, when attempting to understand mechanisms that structure predator-prey responses and transfer of information across aggregated individuals in waters with limited visibility, the task can be near impossible. Recent developments in acoustic imaging and processing techniques afford the ability to quantify interactions among predators and schooling prey in low-visibility waters and provide insight into the structure of schooling fish, their response to predators. Using an imaging sonar, we were able to observe schooling behavior of common estuarine fishes in response to predator activity. We present the development of the collection and analysis techniques for acquiring relevant metrics to better understand fine-scale predator-prey dynamics in a shallow-water high-turbidity ecosystem.

5aFWb3. Estimating fish orientation from multiple view, limited angle, wide band reflections. Jules Jaffe (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093, jules@mpl.ucsd.edu) and Paul Roberts

Multiview, broadband (635–935 kHz), nearly monostatic, acoustic reflections recorded from lateral views of juvenile fish were used to infer animal orientation. Calibrated acoustic data were recorded from live fish in a laboratory while orientation was measured simultaneously via optical images. Using eight animals, 2-D data sets of target strength as a function of frequency and orientation were obtained. Fish length, lateral thickness, and dorsoventral thickness ranged from 24 to 48 mm, 3 to 7 mm, and 10 to 20 mm, respectively. Preliminary estimates of orientation were computed from the direction of the gradient of the local autocorrelation function in the target strength image. These local estimates were then median-filtered over the full system bandwidth (but still limited angle) to improve accuracy. Angular estimates were then corrected for systematic bias via a simple, one-demensional model that approximated the animals' reflection by that of a bar target. Taken over all orientations, the average absolute error in orientation estimation is 5.6 to 17 deg, depending on the data set. Results indicate that, for most sets of views, reasonable estimates of lateral orientation can be obtained from broadband, multiview data over a set of limited angular reflections.

5aFWb4. Passive acoustics as a monitoring tool for evaluating oyster reef restoration. Hilde P. Zenil, Vincent G. Encomio (Res. Dept., Florida Oceanograph. Society, 890 NE Ocean Blvd., Stuart, FL 34996, hzenilbe@yahoo.com), and R. Grant Gilmore (Estuarine Costal and Ocean Sci. Inc., Vero Beach, FL 32968)

Passive acoustics uses naturally occurring sounds produced by marine organisms to study their behavior, biology, and location. Ambient marine sounds are known to vary from place to place, and these sounds can be used to detect differences in habitats. Oyster toadfish, naked goby, mud crabs, and snapping shrimps inhabit oyster reefs, and they are known to produce sounds. In an oyster reef, the combination of sounds produced by organisms' communication, feeding, or moving may produce a unique acoustic signature. Therefore, individual acoustic signatures of oyster reefs may convey information about the habitat quality and the organisms that inhabit them. Three sites along the Saint Lucie Estuary, Florida were acoustically monitored. Restored and natural reefs were recorded for 5-min using a hydrophone. Acoustic signatures were compared using spectra [(frequency (Hz) versus intensity (dB)] overlays. Preliminary results showed that shortly after restoration, acoustic signatures from the natural and restored reefs differed. As time progresses, the acoustic signature of a fully restored reef may resemble that of the natural reef, representing a convergence of restored and natural habitats. Passive acoustics has the potential to provide a new methodology to rapidly monitor oyster reefs and other ecosystems such as coral reefs and rocky reefs.

5aFWb5. Some consideration about the management and economic aspect on the pinger use for mitigation the depredation phenomena in an MPA Mediterranean area. Vincenzo Maccarrone, Francesco Filiciotto, Gaspare Buffa, Antonio Bellante, Di Stefano Vincenzo, Giorgio Tranchida, Carmelo Buscaino, Salvatore Mazzola, and Giuseppa Buscaino (IAMC Capo Granitola, Natl. Res. Council of Italy, via del Mare 3, 91021 Granitola, Trapani, Italy)

This study was carried out in a fishing area off the coast of southern Italy, where *Tursiops truncatus* is frequently observed to interact with bottom gill nets. The experiments aimed to assess the economic efficiency of the DDD pingers during the interaction between dolphins and monofilament bottom gill nets. A net was equipped with pingers and another without to measure the different biomass catch and costs. The economic effectiveness with an estimate of the costs/benefits was evaluated. We modeled a catch per unit effort (CPUE) both in economic terms as euros per 50 m of net set and in biomass terms as kilograms caught per 50 m of net set with and without pingers. The analysis of the data allowed us to evaluate the advantage of the economic aid provided by the European Fisheries Fund (EFF) for the use of pinger. The CPUE of the net equipped with pingers was statistically higher than the control net. The costs of investment for pingers could fully comeback in 74 days and with the economic assistance of EFF in 34 days of fishing. The use of pingers represents productive and economic benefits, even if their potential impact noise should be considered.

5aFWb6. Euphausiids in the Bering Sea: better ecosystem information through acoustics. Patrick H. Ressler and Alex De Robertis (Resource Assessment and Conservation Eng. Div. NMFS, NOAA, 7600 Sand Point Way NE, Seattle, WA 98115, patrick.ressler@noaa.gov)

Better information on the distribution and abundance of euphausiids (*Thysanoessa* spp.), a key group of crustacean zooplankton, meets a specific assessment need in the Bering Sea ecosystem. Multifrequency acoustic classification, backscatter modeling, and net capture were recently used to develop, apply, and validate a method of surveying euphausiid distribution and abundance during regular acoustic-trawl surveys of walleye pollock (*Theragra chalcogramma*), an important commercial fish stock. These observations of euphausiids have both ecological and management implications. Summer surveys indicate that pollock predation may control euphausiid abundance: the stocks are inversely correlated in space and time, and estimated predation by pollock is sufficient to influence the euphausiid standing stock. Spring observations show that euphausiids and pollock are spatially segregated by ice cover and water temperature, which might mediate predation by pollock. This information on euphausiid abundance and distribution is being used as an index of prey availability in pollock stock assessment, and as a way of monitoring the status of euphausiids in assessment of the Bering Sea ecosystem. [Work supported by Alaska Fisheries Science Ctr., the North Pacific Res. Board, and the NSF.]

5aFWb7. Multi-frequency species classification of acoustic-trawl survey data using semi-supervised learning and class discovery. Mathieu Woillez, Patrick H. Ressler, Chris D. Wilson (RACE-MACE, AFSC, NOAA Fisheries, 7600 Sand Point Way NE, Bldg. 4, Seattle, WA 98115, mathieu.woillez@noaa.gov), and John K. Horne (Univ. of Washington, Seattle, WA, 98105)

Data from towed samplers (e.g., trawls) are invaluable for validation of species classifications in fisheries acoustics surveys but may be sparse or unavailable and not all backscatter is classified. This situation is often encountered when little sampling effort is focused on confirming identities of organisms other than target species. To address this situation, a generalized Gaussian mixture model was developed to classify multi-frequency acoustic data under the assumption that some classes of organisms may be unknown. The classification algorithm, based on semi-supervised learning with class discovery, was applied to acoustic survey data collected in the eastern Bering Sea during summers of 2004 and 2006–2008. The major known scattering classes (pollock and euphausiids) were objectively detected and mapped, and compared well with traditional subjective classification methods (i.e. scrutinizing) by acoustic survey analysts. Two other major classes with distinct frequency responses were also identified, occurring mostly in the upper 15–60 m of the water column. The classification results are used to describe these unknown groups, to optimize a field sampling plan to identify their constituents, and to determine whether these groups provide important information about the Bering Sea ecosystem. [Work supported by Alaska Fisheries Science Center, North Pacific Research Board and the NSF.]

5aFWb8. The use and efficacy of acoustic survey methods as applied to a mixed aquatic community: The deep scattering layer of the California Current. Peter Davison, Ana Lara-Lopez, and J. Anthony Koslow (Scripps Inst. of Oceanogr., Univ. of Cal. at San Diego, 9500 Gilman Dr., La Jolla, CA, 92093-0208, pdavison@ucsd.edu)

Concurrent acoustic and trawl data were collected in 2008 near Point Conception to compare the abundance estimates of mesopelagic micronekton obtained with these two methods. Acoustic data were obtained with a calibrated EK-60 equipped with four frequencies (38, 70, 120, and 200 kHz). Animals were collected with a midwater trawl. Target strength (TS) models were created of the animals collected by the trawl, and animals were assigned to acoustic groups based on the frequency spectrum of their modeled TS. The mean spectra of the acoustic groups were used with non-negative least squares inverse methods to estimate abundance from the multi-frequency volume scattering. The resulting acoustic abundances of each group were compared to the trawl catches, and capture efficiency of the net was estimated. The accuracy of the inverse method was tested with a Monte Carlo analysis of an artificial echogram. The accuracy of the forward modeling was tested via comparison to single targets recorded by the EK-60. [Work supported by NASA, CCE-LTER, and The Moore Foundation.]

5aFWb9. Resonance classification of fish with swimbladders using a modified commercial broadband echosounder. Dezhang Chu (2725 Montlake Blvd. E., Seattle, WA 98112, dezhang.chu@noaa.gov), Timothy K. Stanton (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), J. Michael Jech (NOAA/NMFS/NEFSC, Woods Hole, MA 02543), and James D. Irish (Univ. of New Hamshire, Durham, NH 03824)

A towed commercial sub-bottom profiler acoustic system has been adapted for studying fish with swimbladders. It contains broadband acoustic channels collectively spanning the frequency range 1.7–100 kHz, with some gaps. Using a pulse-compression technique, the range resolution of the echoes can be significantly improved, allowing high-resolution imaging of patches and resolving fish near the seafloor. Measuring the swimbladder resonance at the lower frequencies eliminates major ambiguities normally associated with the interpretation of fish echo data: (i) The resonance frequency can be used to estimate the volume of the swimbladder (inferring the size of fish), (ii) signals at the lower frequencies do not depend strongly on the orientation of the fish, and (iii) spectral analysis of backscattered signals can be used to classify different sizes and/or species of fish. At-sea studies of Atlantic herring and other fish and zooplankton species demonstrate the po-

tential for routine measurements of fish size and density, with significant improvements in accuracy over traditional high-frequency narrowband echosounders. New techniques for quantitative use of broadband systems are presented, including broadband calibration and relating target strength and volume-scattering strength to quantities associated with broadband signal processing.

FRIDAY AFTERNOON, 27 MAY 2011 HYATT AMPHITHEATRE, 1:20 P.M. TO 4:30 P.M.

Session 5pFW

Fisheries Workshop: Technological Applications and Innovations III

Martin Siderius, Chair
Portland State Univ., Electrical and Computer Engineering Dept., 1900 SW Fourth Ave., Portland, OR 97207

Contributed Papers

1:20

5pFW1. Calls of the black drum (Pogonias cromis: Sciaenidae): Geographical differences in sound production between Northern and Southern Hemisphere Populations. Michael L. Fine (Dept. of Biology, Virginia Commonwealth Univ., Richmond, VA 23284-2012), Javier Tellechea, Walter Norbis, and Daniela Olsson (Instituto de Biologia)

Because of apparent reproductive isolation between Northern and Southern hemisphere populations of the black drum Pogonius cromis, we tested the hypothesis that advertisement calls from a southern population would differ from known calls of North American populations. Additionally, we quantified disturbance and advertisement calls, their changes with fish size and sex, not previously examined in this species. Unlike most sciaenids, both sexes of P. cromis possess robust sonic muscles, and both produce disturbance calls when handled. However, only males produce an advertisement call used in courtship. The disturbance call consists of a variable train of short-duration pulses (average 23 ms). The duration, interpulse interval, and dominant frequency of pulses are similar in males and females and change developmentally: pulse duration and interpulse interval increase and dominant frequency decreases with fish size. Advertisement calls, recorded in the field and in captivity, are long-duration (average 184 ms) and tonal. Based on variation in fundamental frequency, which decreases with fish size, field choruses are composed of different-sized individuals. The duration of advertisement calls, about a third of those from Florida populations, suggests genetic differentiation between northern and southern populations.

1:40

5pFW2. A cabled acoustic telemetry system for detecting and tracking juvenile salmon. Zhiqun (Daniel) Deng, Mark Weiland, Thomas Carlson (Energy and Environment Directorate, Pacific Northwest Natl. Lab., Richland, WA 99354), Brad Eppard (U.S. Army Corps of Engineers, Portland, OR 97208), Tao Fu, and Gene Ploskey (Energy and Environment Directorate, Pacific Northwest Natl. Lab., Richland, WA 99354)

The juvenile salmon acoustic telemetry system (JSATS) is a nonproprietary technology developed by the U.S. Army Corps of Engineers, Portland District for detecting and tracking small fish. The JSATS consists of acoustic microtransmitters; autonomous, cabled, or portable receivers with hydrophones; and data management and processing applications. Each microtransmitter, surgically implanted in fish, transmits a unique 31-bit binary code encoded using BPSK at 416.7 kHz. Cabled systems are deployed at dams and used to determine passage-route and near-dam behavior for fish. Each cabled system is synchronized to a universal GPS clock and waveforms are saved to the computer before being decoded. Valid detections are separated from spurious detections using filtering processes requiring a minimum of six messages with a pulse interval matching that expected from properly functioning tags within a fixed period. Time-of-arrival information for valid detections on four hydrophones is used to solve for the 3-D position of tagged fish. For the cabled system at John Day Dam, the range for 3-D tracking is more than 100 m upstream of the dam face where hydrophones are deployed. Cabled systems have been successfully deployed on several major dams to acquire information for salmon protection and to develop more fish-friendly hydroelectric facilities. [This study was funded by the U.S. Army Corps of Engineers, Portland District.]

2:00

5pFW3. Effects of construction and operation of offshore wind farms on seals and small cetaceans. Jakob Tougaard (Arctic Environment, Aarhus Univ., DK-4000 Roskilde, Denmark, jat@dmu.dk)

Extensive expansion in offshore wind energy takes place these years in European waters, with North America following. Concern has been about possible conflicts with marine ecosystems, including marine mammals. During the last 10 years, several impact studies have been conducted during construction and first years of operation of wind farms in Europe and general conclusions begin to emerge. Pronounced effects (deterrence of animals) during construction have been observed in most cases. In particular, pile driving of steel monopiles for foundations has repeatedly been demonstrated to affect porpoise behavior at great distance and effects on seal haul-out behavior has been observed in a single case. Controlled exposure studies have confirmed the results and demonstrated reactions to pile driving impact noise at levels around 140 re. 1 μPa. Effects of operation are far less pronounced and range from negative (deterrence) over neutral, to positive (attraction). Noise levels from operating turbines are very low, however, and it is unlikely that deterrence can be attributed to the noise. In general, there appears to be little conflict between marine mammals and operating offshore wind farms, but there is reason for continued attention to the construction phase, in particular, regarding pile driving operations.

2:20

5pFW4. DeepCwind: Application of active and passive acoustics to evaluate potential impacts of deepwater offshore wind technology on fish. Jason D. Stockwell (Gulf of Maine Res. Inst., 350 Commercial St., Portland, ME 04101, jstockwell@gmri.org) and Gayle B. Zydlewski (Univ. of Maine, Orono, ME 04469)

The DeepCwind Consortium is conducting research and development to bring deepwater offshore wind technology to the Gulf of Maine. As part of the process, we are monitoring the potential impacts of 1/3 scale floating wind turbine platform on pelagic fishes and species of interest (e.g., sturgeon, Atlantic salmon) using active (hydroacoustic surveys) and passive (tagging) acoustic systems. We discuss our application of before-after-control-impact designs to evaluate potential impacts, report on our preliminary monitoring activities and results from 2010, and highlight challenges encountered thus far in the context of improving monitoring protocols for future deployment(s) of full-scale floating platforms.

2:40—3:00 Break

3:00

5pFW5. Ultrasonic control of nuisance species in commercial aquaculture settings. Bradley T. Goodwiller, James P. Chambers, Shannon N. Lieblong (Natl. Ctr. for Physical Acoust., Univ. of Misissippi, 1 Coliseum Dr., University, MS 38677), and Rachel V. Beecham (Dept. of Natural Sci., Mississippi Valley State Univ., Box 7254, 14000 Hwy. 82 W., Itta Bena, MS 38941)

One of the major challenges facing commercial catfish farmers is managing water quality and nuisance organisms. High stocking densities coupled with high feeding rates support many parasitic organisms. One such organism is the trematode that lives in an obligate parasitic relationship with another nuisance species, the ram's horn snail. One method of controlling the trematodes is the use of copper sulfate to kill the snails. This is not an ideal solution as the copper will build up and eventually poison the entire pond. An alternate method, pursued here, is the use of high amplitude ultrasound to kill or mortally wound the snails. Since this method of removing the snails is mechanical, it reduces the amount of chemicals used to maintain the pond. To investigate the efficacy of this proposed technique, an ultrasound system was designed and built at NCPA to test the effects of various ultrasonic dosages on pathogens of interest. The system is designed to be tunable, allowing for the widest range of possible combinations of frequency and amplitude. The result of this setup is a well documented rubric for producing maximum efficacy for a given pathogen. Preliminary results and the experimental setup will be presented.

3:20

5pFW6. Toward a global observation and modeling system for studying the ecology of the open ocean using acoustics. Nils Olav Handegard, Geir Huse (Inst. of Marine Res., P.O. Box 1870, Nordnes, 5817 Bergen, Norway), Olivier Maury (Institut de Recherche pour le Dveloppement (IRD), France), and Nils Christian Stenseth (Ctr. for Ecological and Evolutionary Synthesis (CEES)/Univ. of Oslo)

The CLIOTOP (mid-trophic automatic acoustic sampling) project and the EurOcean consortium are organizing a workshop in May 2011 in Bergen, Norway. The presentation will summarize the discussions and conclusions from the workshop. The workshop will deal with the technological and modeling issues related to the mass deployment of acoustic sensors in the open ocean environment. The technological challenges that will be addressed are plans and concepts for novel platforms carrying acoustics and complementary techniques, energy supply and consumption, and data transfer technology. Presently, acoustics cannot provide measures of the species-specific biomass of all taxa, and complementary technologies and ecosystem models need to be tailored to the available data. The modeling part of the workshop will focus mainly on the model-data links. It will cover the topics required to design a large scale observational system and to improve the combination of models and observations through data assimilation. The use of acoustic data in marine ecosystem models is indeed often done in a fairly naive way, typically by assuming that acoustic measurements can provide acute and precise estimates of biomass. This leads to underestimated uncertainty and potentially biased results which need to be rigorously addressed in appropriate state-space frameworks.

3:40

5pFW7. Metrics to characterize vertical distributions of pelagic fauna in large acoustic datasets. Samuel S. Urmy and John K. Horne (School of Aquatic and Fishery Sci., Univ. of Washington, 1122 NE Boat St., Seattle, WA 98105, urmy@uw.edu)

Active acoustics are a valuable addition to ocean observatories, allowing the detection of animals throughout the water column with high temporal and spatial resolution. The resulting datasets are large and can be difficult to describe and visualize in their entirety. We assembled a suite of metrics to parsimoniously characterize the vertical distribution of animal densities, including measures of abundance, density, location, dispersion, occupancy, evenness, and aggregation. We also developed and tested an unsupervised algorithm for detecting and counting backscatter layers within an echogram, using a gradient-based classification. These metrics were used to analyze data from the Deep Echo Integrating Marine Observatory System (DEIMOS), a 38 kHz upward-looking acoustic package deployed at the MARS observatory node in Monterey Bay, CA. The metrics successfully captured biological dynamics across multiple time scales, including seasonal, diel, and tidal variability, in addition to episodic events such as transient aggregations. All metric series showed significant positive autocorrelations at lags less than 26 days and as long as 103 days. Their frequency spectra were power-law distributed with exponents between -0.84 and -1 .73. This approach could be expanded to two or three dimensions, providing an effective means to quantify the temporal variability of any spatial density distribution.

4:00—4:30 Panel Discussion

161st ASA Meeting - Seattle abstracts

ASA

01 **Scott Veirs** Page no. **11**

13/5/2011 12:34

Meet Julie!

02 **Scott Veirs** Page no. **225**

13/5/2011 12:40

Can't highlight! Can you?