Session 4aAAa

Architectural Acoustics and Engineering Acoustics: Rooms for Reproduced Sound I

K. Anthony Hoover, Chair
McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362

Invited Papers

9:00

4aAAa1. Design of critical listening rooms: A historical perspective. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Critical listening room design has progressed to follow loudspeaker evolution from mono to two-channel to multi-channel, as well as the advances in loudspeaker quality. There are basically two fundamental approaches to two-channel room design. The non-environment room, in which the contribution of the room is minimized, and versions of live-end-dead-end, in which early reflections were minimized creating a spatio-temporal reflection free zone and diffuse reflections from the rear of the room are used to create passive surround sound. This presentation will focus on an immersive design for multi-channel loudspeaker formats, which utilizes broad bandwidth absorption and diffusion. Research with low-frequency plate resonators and a broad bandwidth diffusion chamber has contributed to the proposed design and results will be discussed. The proposed multi-channel critical listening room includes the use of low-frequency control down to 40 Hz provided by 4-in.-thick metal plate resonators on a portion of the front wall and in all corners, broad bandwidth diffusion on all wall and ceiling surfaces, and strategically placed multiple subwoofers. The intent is to minimize monophonic reflections and create lateral enveloping energy by uniformly scattering sound from all of the active sound sources in the room.

9:20

4aAAa2. Work on the reproduction of sound at the Visualization Laboratories at the King Abdullah University of Science and Technology: A case study. Steve Ellison (Meyer Sound Labs, Inc., Berkeley, CA 94702, ellison@meyersound.com) and Peter Otto (Sonic Arts, CalIT2/UCSD, La Jolla, CA 92093, potto@ucsd.edu)

Researchers at the newly opened King Abdullah University of Science and Technology are developing new ways to visualize and auralize complex data sets in immersive environments. These environments include the Interactive Media Room, a 100-seat multi-purpose facility, and Cornea, a $10 \times 10 \times 10$ in.^{3} six-sided stereoscopic cave. Both facilities are equipped for multi-channel audio generation and playback in various standard and custom formats, as well as electronically variable acoustics utilizing electroacoustic architecture. Concurrent viewing and dialogue between participants in the two spaces is supported. Fully immersive environments present unique challenges due to physical properties of projection screens and the geometries that characterize these structures. Explorations of Cornea at KAUST and StarCave at UCSD are presented, along with strategies for reproducing sound and varying acoustics in these environments. Multi-use facilities such as the IMR also have conflicting acoustical goals. For instance, the optimal acoustic for multi-channel or cinematic reproduction is less reverberant than optimized classroom environments. The use of electroacoustic architecture to improve the listening and overall immersive experience at both Cornea and the Interactive Media Room is discussed. Examples are presented including an experimental VR emulation of the acoustics of the King Abdullah Grand Mosque on the KAUST campus.

4a THU. AM

9:40

4aAAa3. Seelos Theater: A case study for a renovated multipurpose room with multiple loudspeaker systems. Matthew J. Moore, Alexander G. Bagnall, and Aaron M. Farbo (Cavanaugh Tocci Assoc., 327 F. Boston Post Rd., Sudbury, MA 01776, mmoore@cavtocci.com)

The Seelos Theater at Holy Cross College has been renovated twice since starting out as a bowling alley in the main student center on campus. This most recent renovation provided an opportunity for the school and design team to update and modernize this heavily used lecture hall and cinema. The authors will show how two different loudspeaker systems and AV systems were integrated with the architectural acoustics design built for reproduced sound. The authors found that the resultant performance of the systems performed better than was expected in computer modeling. Existing mechanical system noise mitigation, integration of existing film equipment, short construction timelines, and serving the needs of different audience types will be discussed. The authors used lessons learned with this project to help with the design of spaces for future projects.

10:00

4aAAa4. Speaker coverage in active acoustic music practice rooms. Ronald Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

Larger music practice rooms (>25 m^3) present unique challenges compared to smaller, traditional practice rooms (6–12 m^3) when employing active acoustics technology and integrated digital recording. An experiment was conducted comparing a ceiling array of 32 speakers in a large music practice room ($5.3\times5.7\times2.7$ m^3, 81 m^3) with that of eight speakers located in corners of that room (one at the

top and one at the bottom in each corner facing parallel to the walls). The goal was to determine which speaker configuration would be the most effective for creating even sound field coverage and uniform frequency response. Data were collected for both seated and standing positions in the room from a matrix of 42 locations. Frequencies from 40 Hz to 8 kHz were plotted for analysis. The data were compared to the qualitative preferences expressed by a number of musicians using the rooms for both practice sessions (with the active acoustics enabled) and listening to the playback of recorded practice sessions.

10:20

4aAAa5. Small music venues and amplified sound: High-sound pressure levels and architecture. David S. Woolworth (Oxford Acoust., Inc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

This paper is a survey of typical popular live music venue sound pressure levels observed in Oxford, MS. An attempt is made to measure clarity and articulation in the venues with increasing sound pressure, and the results are correlated with previous observations of Knudsen and others. Additional analysis of challenges to sound operators and programs to control high-stage volumes are discussed.

10:40

4aAAa6. Mixing studio isolation. Anthony K. Hoover (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Westlake Village, CA 91362)

A new suite of mixing and teaching studios had been discussed for many years at Berkley College of Music in Boston. The authorization to proceed with design and construction of the studios finally came with a new Dean of the Music Technologies Department, who, without benefit of much of the background and previous discussions, contacted an old colleague for the final studio design, and then contracted with a reputable full-service contractor for construction services. Some unusual problems developed during construction, which sensitized the client to subsequent situations, in turn requiring acoustical consulting services. This paper will review some of the project history, outline the problems, and discuss solutions and recommendations, and will address some of the concerns with the specialized construction and unusual complications.

11:00

4aAAa7. Edison phonograph recording and reproduction in a concert hall. Alex U. Case (Sound Recording Tech., UMass Lowell, 35 Wilder St., Lowell, MA 01854, alex_case@uml.edu)

A recent recording session at the University of Massachusetts Lowell merged the birth of sound recording with the state of the art. A small ensemble performed an original piece of music—composed in the style of early 1900s American music—on stage in Durgin Concert Hall. It was recorded live to both high-definition digital multitrack and an original Edison wax cylinder phonograph. As the acoustical influence of the phonograph is substantial, several takes were needed to adjust the balance of the live performance to maximize the musical impact of the reproduced sound realized through wax cylinder playback. Multiple sonic viewpoints were simultaneously recorded: stereo, surround sound, close microphones, and Edison phonograph. The resulting recordings are played and discussed.

11:20

4aAAa8. Soul spaces: How acoustics influence the music production process and the recordings that result. Matthew B. Zimmern (8 Illsley Hill Rd., West Newbury, MA 01985, matthewzimmern@gmail.com)

Spaces in modern recording studios are quite different than those of 40 years ago. While today it is common for recording facilities to have various multi-purpose, acoustically designed spaces, many recordings from the middle of the 20th century were often captured in a single room, one that was not constructed with acoustical principles in mind and often times featured acoustical deficiencies. Soul music production practices of the late 1960s were researched, and an original composition was recorded in two different ways: by using past recording techniques and by using contemporary musical practices. Through listening to audio examples of the same song recorded in two different spaces, the sonic qualities of the different recording environments will be discussed, as well as how the different production techniques affected the recording, musical, and aesthetic attributes of the resulting audio recordings.

11:40

4aAAa9. Obtaining a frequency independent reverberation time in listening rooms. Niels Werner Adelman-Larsen (Flex Acoust., SCION-DTU, Diplomvej 377, 2800 Kgs. Lyngby, Denmark, nwl@flexac.com), Cheol-Ho Jeong, and Jiazi Liu (Tech. Univ. of Denmark, 2800 Kgs. Lyngby, Denmark)

High-frequency sound emitted from loudspeakers is very directive and can be easily aimed at the listeners. Investigations show that even a few people decrease the mid-/high-frequency reverberation time of a room significantly both due to actual absorption but also due to the scattering of the sound that they introduce which makes the placement of absorptive material important. People absorb approximately one-sixth of the sound at low frequencies than at high frequencies, but still much reproduced music contains very loud levels of bass frequency sound. In order to avoid loud, reverberant bass sounds masking the direct mid-/high-frequent sound, the room needs to be designed with quite large amounts of bass absorption in order to properly serve its purpose. Helmholtz and porous type absorbers need a large cavity in order to also achieve bass absorptive qualities and this is a challenge in many smaller rooms. Membrane absorbers require less space and due to the omni-directional behavior of bass sound their placement is not so critical. This paper will present new measurements of absorption coefficients of standing persons as well as a new type of modular bass absorber system: 2.6 in. deep, absorption coefficient of 0.6 from 50–125 Hz, and designed to readily incorporate into the architectural design.

Session 4aAAb

Architectural Acoustics: Hidden Gems

Andrew N. Miller, Chair
BAi, LLC, 4006 Speedway, Austin, TX 78751

Chair's Introduction—8:00

Invited Papers

8:05

4aAAb1. The Festival Hill concert hall. Richard Boner and Pamela Harght (BAi, LLC, 4006 Speedway, Austin, TX 78751 rboner @baiaustin.com)

The Festival Institute at Round Top, TX, was established in 1971, and began with ten pianists performing on an open stage, under the stars. Following the dream of the founder, Concert Pianist James Dick, Festival Hill has grown to be a premier summer venue for classical music, with over 30 concerts performed each June and July. The August-to-April Series, the International Guitar Festival, the Theatre Forum, The Poetry Forum, and the Herbal Forum bring the total number of year-round events to more than 50. This paper focuses on the excellent and visually unique 1000 seat concert hall, which was built in stages over a 26-year period from 1981–2007. Also discussed are the renovated 19th-century homes which have been moved to Festival Hill, serving as housing and practice spaces for musicians.

8:30

4aAAb2. Whispering arches as intimate soundscapes. E. Carr Everbach (Swarthmore College, Swarthmore, PA 19081) and David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683)

Whispering archways are whispering galleries. They are sometimes found around smooth arches, doorways, and even bridges. Persons positioned on opposite sides of a whispering arch communicate privately by whispering into the arch. In that way they function as private and even intimate soundscapes. Some whispering arches that have stood for centuries have inspired local folk legends. Such acoustical architecture can help to establish a sense of place and strengthen emotional ties. It is said that a whispering arch at the entrance to a 10th c. Gothic cathedral at the historic monastic site of Clonmacnoise in the Irish midlands figured in local courtship for centuries. There, shy young lovers are said to have uttered vows of love and proposals of marriage. The arch was also used by lepers to give confession without infecting the priest. Whispering arches were almost certainly unintended in the past but can also be created intentionally as an intimate form of public art. Recent measurements on a whispering archway at West Chester University show that high background noise levels from traffic or air handlers can ruin an otherwise intimate space; however, so modern architects should plan the entire soundscape carefully.

Contributed Papers

8:55

4aAAb3. A performers guide to venue acoustics. David J Zartman (Graduate Program in Acoust., Penn State Univ., 405 EES Bldg., State College, PA 16801)

Whether in high school or the Vienna Symphony, musicians have little control over their performance venues. An instrument can be upgraded or replaced if there is an undesirable aspect, but replacing a venue is often far too expensive to be a tenable solution. A musician's perfect venue will also change piece to piece—an incredible venue for lush Wagner operas can cause issues for someone performing an intricate Paganini concerto and vice versa. What then are these effects? Why do they take place? How can a performer anticipate and combat undesirable aspects of venues while drawing out their featured characteristics for an optimized performance?

9:10

4aAAb4. Vineyard shape variations of the Concertgebouw, Amsterdam. Weihwa Chiang (Natl. Taiwan Univ. of Sci. and Technol., 43, Keelung Rd., Section 4, Taipei 106, Taiwan, Republic of China) and Wei Lin (Hwa Hsia Inst. of Technol., Chung Ho, Taipei, Taiwan, Republic of China)

The Concertgebouw, Amsterdam is a unique one of its kind because of the extra width and size for a classical rectangular hall and the significant portion of seats behind the stage. These characteristics make the hall as "semi-surround" as the vineyard halls of the present day. Computer simulation was performed to analyze the possibility of deriving design variations out of the Concertgebouw. Clarity, strength, sectional balance, and stage support of various settings of stage position and overall proportion were compared as the first step. Variations that contain the geometrical features of vineyard shape halls were further developed to optimize the three acoustical qualities and to reduce the differences among seats. The architectural features evaluated included surface treatments, arrangement of individual seating blocks, and the geometry of upper walls and the ceiling.

Session 4aAAc

Architectural Acoustics and Physical Acoustics: Physical Acoustics in Boston Symphony Hall: A Guide for the Perplexed

James B. Lee, Chair
6016 SE Mitchell, Portland, OR 97206

Chair's Introduction—9:25

Invited Papers

9:30

4aAAc1. Back to vitruvius? James B. Lee (6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

There is no generally accepted theory of acoustics of concert halls based on the physics of sound as waves. Lord Rayleigh, founder of the science, remarked but little about sound in rooms. Wallace Clement Sabine made an analogy of sound in rooms to the kinetic theory of gases; although this is theoretically untenable, Sabine designed the superb concert hall which is the subject of this session. Leo Leroy Beranek published the first systematic examination of concert halls, demonstrating that subjective evaluations of acoustic quality are consistent: everyone agrees which are good and which are poor; nonetheless Beranek encountered severe difficulty in applying his experimental results. What aspects of the physics of Rayleigh explicate the data of Beranek and the design of Sabine?

9:55

4aAAc2. Simple acoustic source radiation near a large wall. Michael J. Moloney (Dept. of Phys. and Optical Eng., Rose-Hulman Inst. of Technol., 5500 Wabash Ave., Terre Haute, IN 47803, moloney@rose-hulman.edu)

An acoustic simple source is predicted to radiate twice as much sound near an infinite rigid wall as it does in free space. As the simple source becomes more distant from the wall, its radiated power is predicted to decrease in an oscillatory way toward the value it has in free space. This behavior can be observed indirectly by fitting a 10-s signal of decaying amplitude from a tuning fork resonator box at various distances from a large unobstructed wall. Each fit determines the damping constant b, which reflects power losses, including radiated acoustic power. In a plot of b versus distance from the wall, the oscillatory part of the damping constant closely matches the theoretical curve of total radiated power from a simple acoustic source near an infinite rigid wall.

10:20—10:30 Break

10:30

4aAAc3. Evaluation of a boss model and subtraction technique for predicting wideband scattering phenomena in room acoustics. Georgios Natsiopoulos (WSP Acoust., Rullagergatan 4, 41526 Gothenburg, Sweden, georgios.natsiopoulos@wspgroup.se)

Scattering from a hard hemisphere on an infinite plane is studied experimentally and theoretically in a laboratory environment using a subtraction technique. With this technique, the scattered field is determined by subtracting the transient responses from the plane with and without the boss present. Scattering for different combinations of transducer and hemisphere positions are then classified into a number of cases, denoted in-plane/off-plane and back-/forward/mixed scattering. Theoretical and experimental results are compared and evaluated from an auralization point of view using a one-third octave band smoothing of the contribution from the hemisphere. With the smoothing used, average intensity level discrepancies typically less than 3 dB are obtained for a six octave wide ka-interval, where a equals the hemisphere radius.

10:55—11:00 Break

11:00—12:00 Panel Discussion

Session 4aAB

Animal Bioacoustics, Signal Processing in Acoustics, and Noise: Topical Meeting on Signal Processing for Subtle and Complex Acoustic Signals in Animal Communication I: Automated Classification of Animal Acoustic Signals I

Ann E. Bowles, Cochair
Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109

Sean K. Lehman, Cochair
Lawrence Livermore National Lab., Livermore, CA 94550

Chair's Introduction—8:55

Invited Papers

9:00

4aAB1. Model-based signal processing approach to animal bioacoustics: A brief overview. James V. Candy (Lawrence Livermore Natl. Lab., P.O. Box 808, L-151, Livermore, CA 94551)

The model-based approach to signal processing is generally founded on the fundamental concept of incorporating any *a-priori* knowledge of the underlying phenomenology from which the signal evolved along with measurement instrumentation and uncertainty (noise, parameters, etc.) in the form of mathematical models that are embedded in the processor. In this way, the phenomenologist, experimenter, and signal processor combine all of their possible knowledge into a scheme enabling each to think within their own comfort zones while developing a powerful approach to extract the illusive information they desire. In this overview, we present the concepts required to develop mode-based processing schemes that can be used in a wide variety of animal bioacoustics applications ranging from signal estimation, tracking, identification, detection, and classification. We discuss the development of this approach incorporating acoustic applications that can be extrapolated to animal bioacoustics problems. We can express all of these techniques in terms of a model-based framework enabling the use of such powerful estimation techniques such as Kalman filters (Gaussian case) to Bayesian particle filters (non-Gaussian case) for solution to a wide variety of problems.

9:20

4aAB2. Detecting and identifying cetaceans species from their acoustic emissions. Whitlow Au and Julie Oswald (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734)

In the past decade, the use of autonomous remote passive acoustic recorders to detect the presence of cetaceans has grown enormously throughout the world. Since these devices digitize acoustic signals at sampling rates in the tens of kHz, a single device can quickly accumulate many Gbytes of data in a short time. Therefore, automatic detection and recognition algorithms must be developed to handle the large amount of data. There is no general signal method processing method that can be applied to all species of cetacean. The algorithm by the name of ROCCA (real-time odontocete call classification algorithm) developed by Dr. Julie Oswald is probably the most general one that has been applied to the whistles of dolphins. The program, extensible bio acoustics tool (XBAT) developed by Harold Figueroa, is probably the most general tool for use with baleen whales. However, investigators also tend to develop their own algorithms to suit their requirements. The problems and difficulties in developing automatic detection and recognition algorithms will be the focus of this presentation. General principles associated with the geography, biology, and the characteristics of sound emissions in relation to the development of algorithms will be discussed along with examples of different approaches.

9:40

4aAB3. Overcoming the idiosyncrasies of spectrogram correlation detection. Kurt M. Fristrup (Natural Sounds Program, Natl. Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525)

Spectrogram correlation has been widely used to detect animal sounds. Effective application of this method to identify Black-capped Vireo (*Vireo atricapillus*) sounds in 22 000 h of environmental recordings required the use of multiple spectrogram templates for each general type of sound to ensure that substantial subsets of these sounds were not missed by the detector. Over 5 million candidate sounds were identified, but the majority of these were not confirmed by expert review. Accordingly, acoustical features were extracted from a stratified random sample of 8284 candidate sounds that were identified by experts, and a random forest classifier was developed to winnow out the false alarms. The estimated classification error varied by site from 0.5% to 6%. When this classifier was applied to the entire data set, approximately 740 000 vireo sounds were identified. This project illustrates both the feasibility of monitoring birds over large spatial and temporal scales and the challenge of adequately sampling the range of variation in a very restricted class of biological signals.

10:00—10:15 Break

10:15

4aAB4. Individual identification using hidden Markov models for population monitoring and assessment. Michael T. Johnson (EECE, Marquette Univ., 1515 W. Wisconsin, Milwaukee, WI 53233) and Patrick J. Clemins (Assoc. for the Advancement of Sci., Washington, DC 20005)

Individually distinct acoustic features are present in a wide range of animal species, just as they are in humans, and the widespread success of speaker identification in humans suggests that robust automatic identification of individual animals from their vocalizations is attainable. Despite this, only a few studies to date have yet attempted to use individual distinctiveness to help assess population structure, abundance, and density patterns. Here we present an approach, based on individual identification and clustering using hidden Markov models (HMMs), which enables a more direct mechanism for using individual vocal variability to monitor and assess populations. Current results indicate that the new method is able to give good estimates of local abundance based on vocalization clustering, which can in turn be used in an acoustic mark-recapture framework to estimate population. Limitations to this approach currently include the need for explicit call-type separation prior to individual clustering, which is possible in many species but can create a problem in species with unknown or variable repertoires. Overall, it is hoped that this new technique may lead to a more accurate understanding of population structure and abundance on a larger scale.

10:35

4aAB5. Identifying delphinid whistle contours using graph search. Marie A. Roch, Bhavesh Patel (Dept. of Comp. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720), Shannon Rankin (NOAA SW Fisheries Sci. Ctr., La Jolla, CA 92037-1022), Yvonne Barkley (Bio-Waves Inc., Encinitas, CA 92024), Melissa S. Soldevilla (Duke Univ., Beaufort, NC 28516), and John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093-0205)

The automated characterization of delphinid whistle contours can lead to insights (both potential and realized) into biological questions such as habitat use and behavior. Prior to the 1990s, most measurements of whistle contours were conducted manually by trained analysts, and even today the noisy signal environment offers significant challenges to fully automated classification systems. In this talk, we provide an overview of challenges and historical approaches to this problem. We present our recent research using graph search algorithms to characterize complex auditory scenes involving numerous animals vocalizing simultaneously. We end by discussing techniques that have the potential to further advance this area of research. [Work sponsored by ONR.]

10:55

4aAB6. Wavelets: A comparison with the spectrogram and other methods for time-frequency analysis. Patrick Loughlin (Dept. of Bioengineering, Univ. of Pittsburgh, 745 Benedum Hall, Pittsburgh, PA 15261, loughlin@pitt.edu) and Leon Cohen (Hunter College, CUNY, New York, NY)

Many natural signals exhibit spectral content that changes over time. Methods for time-varying spectral analysis first emerged in the 1940s with the development of the “sound spectrograph” at AT&T Bell Laboratories. Since then, the spectrogram has become the primary method for time-frequency analysis. Originally implemented as a bank of band-pass filters, today the spectrogram is typically computed digitally via the short-time Fourier transform. Recently, wavelets have been proposed as a superior method for time-frequency analysis. The usual argument is that the spectrogram uses a fixed window length, whereas the wavelet approach uses windows that are longer for lower frequencies and shorter for higher frequencies. While the benefits of this approach are usually taken as self-evident, we explore in critical detail the aims of time-frequency analysis, and the benefits afforded by wavelets versus the spectrogram and modern methods such as the Choi–Williams distribution. In particular, since a primary aim of time-frequency analysis is to study the local spectral and temporal characteristics of signals, we examine the local moments of the various methods. Local moments are related to important signal features such as the instantaneous frequency and bandwidth. We show the effect of fixed windowing versus variable wavelet windowing on these features.

11:15—12:00 Demonstrations and Discussion

Session 4aAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Inversions in Ocean Environments

Yong-Min Jiang, Cochair
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Alexandra I. Tolstoy, Cochair
ATolstoy Scientific, Inc., 1538 Hampton Hill Cir., McLean, VA 22101

Contributed Papers

8:30

4aAO1. Variation of uncertainty and resolution with problem formulation in continuous geoacoustic inversion. Andrew A. Ganse and Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aganse@apl.washington.edu)

In continuous geoacoustic inversion, the resolution and uncertainty of the estimated ocean bottom are inherently linked by a tradeoff with each other. They also depend on the experiment geometry and the choice of formulation for the data and for the bottom model. For example, the resolution and variance will differ if the data are represented by an intensity envelope timeseries or a tau-p timeseries, or if wave slowness or bottom impedance is estimated. Previous work by the authors investigated variations in resolution and uncertainty with geometry and problem formulation based on linearization of the geoacoustic problem at a given solution. However, the problem is further complicated by the fact that the solution point itself can also depend on the geometry and data and model formulation so that the resolution and uncertainty vary for that reason also. This aspect of the nonlinear geoacoustic inverse problem is explored here. [Work supported by ONR.]

8:45

4aAO2. The estimation of geoacoustic parameters via frequencies 25–100 Hz. A. Tolstoy (ATolstoy Sci., Inc., 1538 Hampton Hill Cir., McLean, VA 22101)

This work will discuss recent efforts to extend a previously discussed low-frequency (LF) geoacoustic inversion method to slightly higher frequencies. In particular, the earlier method was "successful" at frequencies 25–50 Hz where an exhaustive search was possible even in the presence of errors in other (assumed known) parameters. However, some of the test data analyzed offered only one appropriate frequency (53 Hz) and did not converge to a unique solution. In fact, thousands of data fits were found. Here, we shall examine efforts to improve convergence by allowing for slightly higher frequencies. Range may also be a consideration at these LFs.

9:00

4aAO3. Travel time inversion of broadband data from Shallow Water 2006 experiments. Yong-Min Jiang (minj@uvic.ca) and N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper presents geoacoustic inversions of broadband signals collected by the L-shaped array SWAMI32 during the Shallow Water 2006 Experiments. The L-shaped array was deployed in 70 m of water off the coast of New Jersey. The vertical leg of the array (VLA) has 10 even spaced sensors which expends 53.55 m in the water column, while the horizontal leg (HLA) has 20 uneven spaced bottom moored sensors that give 256.43 m of the aperture. An acoustic source was maintained at a depth of 35 m and towed along a circle around the VLA at a speed of 0.5 knots. The distance between the acoustic source and the VLA was around 190 m. Mid-frequency (1100–2900-Hz) chirps received at the VLA and HLA were analyzed for investigating the variability of the sea bottom properties around the circle. The data at the HLA were analyzed to assist the identification of the bottom layer structure while the data at VLA were employed to carry out the geoacoustic inversion. Environmental data collected in the vicinity were used in the inversion to account for the variable water column environment. [Work supported by ONR Ocean Acoustics.]

9:15

4aAO4. Low-frequency attenuation and sound-speed measurements in marine sediments using an impulsive source. Altan Turgut, Jeff Schindall, and Steven Means (Acoust. Div., Naval Res. Lab., Washington, DC 20375)

Low-frequency (200–1000-Hz) sediment attenuation and sound-speed measurements were conducted at a site on the New Jersey Shelf where additional data sets from a chirp-sonar bottom profiler (2–12 kHz) and an acoustic-probe system (10–80 kHz) are available. Impulsive sound signals, generated by automated light-bulb implosions, are received by a 16-element vertical line array at short ranges (<500 m). Precursor arrivals and signals reflected from the R-reflector are used to estimate sediment sound-speed and attenuation. Attenuation in dB/m/kHz is estimated using the spectral-ratio technique and sound-speed is estimated from the travel-time analysis. Frequency dependency of sound-speed and attenuation is also investigated within a wide frequency band (200 Hz–80 kHz) using the results from impulsive source, chirp-sonar, and acoustic-probe measurements. Measured attenuation and sound-speed values seem to be well predicted by an extended Biot theory for sediments with distributed pore sizes. [Work supported by ONR.]

9:30

4aAO5. Studies on the effect of shear on compressional wave attenuation. Gopu R. Potty and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882)

Results of a modeling study to examine the effect of shear on compressional wave attenuation are presented. This study also investigates a shear inversion algorithm based on interface waves using synthetic data. Recent studies suggest that inclusion of shear speed is necessary to explain the correct frequency dependence of attenuation. Synthetic data will be generated for elastic bottom, with different shear speeds, and these data will be inverted for compressional wave attenuation. This could provide insight into the effect of shear speed on the attenuation coefficient obtained from inversion at various frequencies. In addition to investigating this effect, we also develop inversion algorithms for shear speed. One of the most promising approaches is to invert the relation between seismo-acoustic interface waves (Scholte waves) that travel along boundaries between media and shear wave speed. The propagation speed and attenuation of the Scholte wave are closely related to shear-wave speed and attenuation over a depth of 1–2 wavelengths into the seabed. The dispersion characteristics of the Scholte wave has been successfully used for inversion of sediment shear properties. Synthetic data will be used in our study to develop an inversion scheme. [Work supported by Office of Naval Research.]

9:45

4aAO6. Toward passive acoustic remote sensing of ocean-bottom gas seeps. Chad A. Greene and Preston S. Wilson (Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292)

Cold seeps are gas vents that are found in the ocean, often along continental shelves near sediment-borne methane hydrates. Methane hydrates and methane gas seeps are of particular interest both for their potential use as an energy source and for their possible contribution to global climate change. This work is an initial step toward passively locating cold seeps and quantifying their gas flow rates using acoustic remote sensing techniques. Results are presented from laboratory experiments in which gas fluxes were determined from the radiated acoustic signature of a model seep. The physical principle that supports the technique and its accuracy will be discussed. [Work supported by ONR.]

10:00—10:15 Break

10:15

4aAO7. The card-house structure of mud: Energy between particles and its effect on bubble formation. Joseph O. Fayton (Rensselaer Poly. Inst., Troy, NY 12180, faytoj@rpi.edu), Allan D. Pierce, William M. Carey (Boston Univ., Boston, MA 02215), and William L. Siegmann (Rensselaer Poly. Inst., Troy, NY 12180)

A model of mud as a lyophobic colloid [Verwey and Overbeek (1948)] leads to a card-house structure of platelets, typically kaolinite or smectite particles. Because of isomorphous substitution, each platelet carries a distributed negative charge. The structure is immersed in water which, especially so for sea water, carries positive and negative ions, and the positive ions tend to cluster on both sides of the platelets, so that the composite platelet charge is zero, but each platelet is analogous to a sheet of longitudinal electrical quadrupoles. The charge distribution causes parallel platelets to repel each other, but which can attach so that one platelet edge can touch the face of another platelet. The energy associated with such attachments is discussed, and it is conjectured that it is associated with van der Waals and London attraction forces. Recent sound transmission experiments by Carey at Dodge Pond in Connecticut support the contention that sediment mud invariably contains vapor bubbles, and other work such as that of Boudreau *et al.* (2005) suggests that the bubbles are considerably flattened, with shapes that have been described as corn-flakes. Present paper conjectures as to whether these shapes are associated with the card-house structure of mud.

10:30

4aAO8. An investigation of the combustive sound source. Andrew R. McNeese, Jason D. Sagers, Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292), and David P. Knobles (The Univ. of Texas at Austin, Austin, TX 78713-8029)

The combustive sound source (CSS) is a versatile impulsive underwater sound source with an adjustable bandwidth and output amplitude. Unlike typical impulsive acoustic sources, CSS can maintain a wide bandwidth at low amplitude; hence, it is a more environmentally friendly impulsive source for ocean acoustics experiments and surveys. The CSS consists of a submersible combustion chamber, open to the water, which is filled with a combustible fuel/oxidizer mixture. The mixture is ignited and a combustion wave propagates through the mixture. During this process the ensuing bubble expands due to an increase in temperature and can collapse to a smaller volume than before ignition. This bubble activity radiates acoustic pulses. The CSS can be used as a source for low-frequency sediment characterization and TL measurements, and when deployed on the bottom can produce seismic interface waves. In addition to stationary deployments in the water column, CSS can be deployed in a tow body and as an array. Discussion will focus on the latest CSS design including functionality, acoustic output, long-term operational stability, and future development plans.

10:45

4aAO9. Detection and classification of sub-surface features based on their seismo-acoustic signatures. Irena Lucifredi and Miaki Ishii (Harvard Univ., 20 Oxford St., Cambridge, MA 02138, irenav@alum.mit.edu)

Complex subsurface structure makes seismo-acoustic imaging difficult, raising the need for novel detection, classification, and imaging approaches. For the representation of range-dependent seismo-acoustic propagation, we employ a hybrid, coupled wavenumber integration approach to range-dependent seismo-acoustic modeling based on the OASES environmental modeling framework. In feature identification and classification, time series analysis frequently provides the sought-after recognition clues. Nevertheless, the geometry and the structure of the subsurface features generate a variety of complexities to the wave field caused by different physical mechanisms, geometric constraints, and intrinsic properties. Based on the time series attributes such as the time of arrival, waveform amplitude, and the frequency content, a set of identification features is extracted and a set of corresponding classes is formed. Using the simulated data scenarios, general rules incorporating the identification features of subsurface structures are deduced. This classification methodology is then applied to the data recorded by an array of seismographs and hydrophones on and off the coast of Japan. The results create a basis for classification and description of the structures of interest such as the presence or absence of very low-velocity range-dependent layers below the water-seabed interface.

11:00

4aAO10. Numerical experiments on weak shock propagation in marine sediments. B. Edward McDonald (Naval Res. Lab., Washington, DC 20375, mcdonald@sonar.nrl.navy.mil)

NRL is planning a set of laboratory and field experiments to investigate weak shock propagation in saturated marine sediments. Questions to be addressed are as follows: (1) How much does the granular content of the sediment increase the effective nonlinearity coefficient? (2) Does the 3/2 order Hertzian nonlinearity inherent to grain contacts play a significant role? Numerical and analytic solutions involving the Hertzian nonlinearity reveal maximum nonlinearity coefficient near zero stress, whereas in a fluid, nonlinearity increases with stress. Numerical experiments are presented using the NPE model [McDonald and Kuperman, J. Acoust. Soc. Am. 81, 1406 (1987)] to determine whether shocks resulting from Hertzian nonlinearity can be observed in the presence of nominal values of frequency-linear attenuation common to granular media. [Work supported by the Office of Naval Research.]

11:15

4aAO11. Tank experiments for validation of volume scattering models. Jorge E. Quijano and Lisa M. Zurk (Dept. Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201)

Mid-frequency scattering from random media is of interest for active sonar applications and, in particular, in shallow water environments, where ocean bottom interaction may affect the performance. To estimate the contribution of volume scattering from the sea bed, models based on the wave equation have been developed and more recently, an approach based on transport theory was proposed [J. Acoust. Soc. Am. **126**, 1711–1723 (2009)]. Validation of this model is difficult to achieve in field experiments due to environmental uncertainties and in many cases to the lack of ground truth revealing the structure of the random media. As an alternative, tank experiments were implemented in a controlled environment at scaled frequencies to explore the range of applicability as a function of different input parameters such as frequency of operation, scatterer size, and concentration. For this work, gel slabs containing random distributions of micron-sized glass beads were manufactured, and broadband pulses at the frequency band 200–600 kHz were utilized to characterize the scatterer contribution. Acoustic propagation through this heterogeneous media is analyzed using the proposed radiative transfer method and the results are compared to the analytical solution of the wave equation for the long-wavelength approximation.

11:30

4aAO12. Inverting for surface displacement fields using directly measured point-to-point sensitivity kernels. Jit Sarkar, Shane Walker, Bruce Cornuelle, William A. Kuperman (Scripps Inst. of Oceanogr., UCSD, Mail Code 0238, 9500 Gilman Dr., La Jolla, CA 92093-0238, jit@mpl.ucsd.edu), Philippe Roux, and Christian Marandet (LGIT, BP 53, 38041 Grenoble Cedex, France)

The effect of surface perturbations on underwater acoustic fields has been directly measured in an ultrasonic tank experiment. High-frequency transducer arrays of 64 elements each are placed 600 mm apart, submerged in ~50 mm of water. A point perturbation is used to displace the surface sequentially in 1-mm steps between the two arrays. At each position a 3.8-MHz signal is transmitted from the source array (round-robin), and the response recorded on the receive-aray, both with and without the presence of the surface probe. The difference between these two measurements yields the point-to-point acoustic sensitivity to that perturbation. These data are explored with respect to inferring the surface structure through both linear full-wave inversions and double-beamforming.

THURSDAY MORNING, 22 APRIL 2010 GALENA, 7:55 A.M. TO 12:05 P.M.

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound Induced Cellular Bioeffects

E. Carr Everbach, Chair
Swarthmore College, Dept. of Engineering, 500 College Ave., Swarthmore, PA 19081-1397

Chair's Introduction—7:55

Invited Papers

8:00

4aBB1. Temporary modulation of vascular barriers with focused ultrasound and microbubbles. Nathan McDannold (Dept. of Radiology, Brigham & Women's Hospital, 221 Longwood Ave., Rm. 521, Boston, MA 02115)

The combination of focused ultrasound and preformed microbubbles (ultrasound contrast agents) circulating in the bloodstream provides an opportunity to utilize mechanical stimulation of endothelial cells to temporarily and locally modulate transvascular transport and permeability. This stimulation may directly alter the endothelium and increase free transport of agents out of the vasculature or it may trigger a physiological response and increase active transport. This talk will review recent work in the brain and kidney utilizing low-intensity focused ultrasound bursts and microbubbles to temporarily disrupt barriers to transvascular passage. In the brain, these sonications result in a temporary disruption of the blood-brain barrier and potentially enable a means for targeted drug delivery. Recent work will be shown demonstrating that the method is sensitive to anesthesia agent, perhaps due to differences in vasoactive effects. In the kidney, we have found that these sonications result in a temporary increase in glomerular filtration rate and urine production and the ability to temporarily pass larger molecules, potentially providing a new platform for the study, and perhaps treatments, of renal disease. Overall, the interaction between ultrasound, microbubbles, and the endothelium presents a unique means to interact with blood vessels and provides opportunities to develop novel treatments.

8:20

4aBB2. Study of sonoporation at the single cell level and cellular bioeffects associated with sonoporation facilitated by microbubbles. Cheri X. Deng (Dept. of Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109)

The interaction of ultrasound-driven microbubbles with cells results in multifaceted cellular bioeffects including physical disruption of cell plasma membrane and subsequent downstream effects. Microbubble facilitated sonoporation, the ultrasound-induced disruption of plasma membrane, enhances the transport of ions, other intracellular contents, and external agents through the membrane into the cytoplasm via diffusion, thereby making it useful for intracellular drug and gene delivery. Transport of these entities, which depends on the dynamic process of pore formation and resealing, plays important roles in many downstream cellular effects of sonoporation beyond the initial pore formation and subsequent diffusion-related transport. For example, extracellular calcium ions diffused into the cytoplasm initiate the process of membrane resealing. In addition, intracellular calcium transients are generated which may be related to many cellular processes triggered by the important second messenger molecule. Recent results of sonoporation at the single cell level and related bioeffects will be discussed. Electrophysiological techniques reveal the dynamic process of sonoporation. Real-time fluorescence imaging measurements of intracellular calcium concentration in mammalian cells subjected to sonoporation demonstrate the spatiotemporal evolution of sonoporation related calcium transients including the large influx of calcium in sonoporated cells and the complex dynamic calcium oscillations and waves. [Work supported by NIH.]

8:40

4aBB3. Photoacoustic delivery of molecules into cells. Mark R. Prausnitz, Prerona Chakravarty, Wei Qian, and Mostafa A. El-Sayed (Georgia Inst. of Technol., Atlanta, GA 30332, prausnitz@gatech.edu)

A major barrier to drug and gene delivery is crossing the cell's plasma membrane. This study presents a physically based mechanism to deliver molecules into cells with high efficiency and viability using photoacoustic effects generated by carbon nanoparticles. The results demonstrated intracellular delivery using this method in multiple cell types for uptake of small molecules, proteins, and DNA. At optimized conditions, calcein uptake was seen in up to 50% of cells with nearly 100% viability and in 90% of cells with $\geqslant$90% viability. Uptake was not observed when cells were irradiated in the absence of carbon nanoparticles. These studies suggest that uptake occurs due to transient membrane permeabilization resulting from explosive photoacoustic forces generated by laser-induced carbon-steam reaction, C(s) + H2O(*l*) = CO$(*g*) + H2(*g*). This synergistic use of nanotechnology with advanced laser technology could provide an alternative to viral and chemical-based drug and gene delivery.

9:00

4aBB4. Ultrasound standing wave fields induce endothelial cell sprouting within three-dimensional engineered tissue. Kelley A. Garvin (Dept. of Biomedical Eng., 205 Goergen Hall, Univ. of Rochester, Rochester, NY 14627, garvin@bme.rochester.edu), Denise C. Hocking, and Diane Dalecki (Univ. of Rochester, Rochester, NY 14627)

The field of tissue engineering is working to develop fully functional replacement tissues and organs. To achieve this goal, methodologies aimed at controlling the growth of new vascular systems in three-dimensional (3-D) engineered tissues are needed. We hypothesized that organizing endothelial cells into multicellular, planar bands of cells within 3-D collagen gels using the radiation forces developed in an ultrasound standing wave field (USWF) would promote an angiogenic endothelial cell phenotype. Human umbilical vein endothelial cells were suspended in an unpolymerized type-I collagen solution and were exposed to continuous wave USWFs. The collagen solution was allowed to polymerize during the 15 min USWF treatment to maintain the USWF-induced banded pattern of cells within a 3-D collagen gel. Following a 24 h incubation period, endothelial cell sprouts were observed emerging from USWF-induced endothelial cell bands. The average length of these sprouts was ~100 μm. Sprouting was absent in sham samples where a rounded cell morphology was observed. The influence of acoustic exposure parameters on endothelial cell sprouting was investigated. These studies indicate that USWF technologies promote formation of capillary precursors in 3-D engineered tissue and thus, this technology has the potential to advance the field of tissue engineering.

9:20

4aBB5. Histological evaluation of continuous and pulsed high-intensity focused ultrasound induced vascular bioeffects in the presence or absence of contrast agents. Anna Tokarczyk, Ian Rivens, and Gail ter Haar (Joint Dept. of Phys., ICR: Royal Marsden NHS Foundation Trust, 15 Cotswold Rd., Sutton SM2 5NG, United Kingdom, anna.tokarczyk@icr.ac.uk)

A novel experimental model which uses an isolated, cannulated rat mesenteric artery (~ 400 μm diameter) has been used for the visualization of vessel behavior and damage during therapeutic ultrasound (US) exposures in the presence of contrast agents. The experimental setup includes a fluorescence microscope, ultrasound transducer, and a chamber in which the vessel, attached to micropipettes, can be perfused with a feeding buffer ± contrast agents and fluorescent dyes. A range of continuous and pulsed 1.7 MHz high intensity focused ultrasound (HIFU) exposures have been used at therapeutic and sub-therapeutic intensities (pressures). Vessel wall damage and leakage of intravascular buffer have been observed, predominantly in the presence of contrast agents. The observed lack of vascular constriction and expansion in response to phenylephrine and acetylcholine indicates smooth muscle and endothelial cell damage. This has been confirmed by histology (haematoxylin and eosin staining) and immunohistochemistry. Factor VIII and alpha-actin antibodies (which have receptors localised on endothelial, and smooth muscle cells respectively) have been used for accurate localisation of HIFU affected areas. This model will be also used to investigate the bio-effects induced by the exposure of vessels to diagnostic US exposures in the presence of contrast agents.

9:40

4aBB6. A pre-treatment planning strategy for high-intensity focused ultrasound treatments. Billy Andre (Dept. Mech., Boston Univ., 110 Cummington St., Boston, MA 02215, billy.andre@gmail.com), Phillip Jason White, Nathan McDannold, and Gregory L. Clement (Brigham and Women's Hospital, Boston, MA 02115)

High-intensity focused ultrasound (HIFU) therapy is a non-invasive treatment method for uterine fibroids patients. However, there exists focal distortion, due to inhomogeneous tissues, that can divert therapeutic dosage to non-targeted locations. A missing component of current HIFU protocols is a simulated pre-treatment planning, a procedure that can predict the efficiency of ultrasonic energy delivery pathways based on MRI scans using a simulated propagation model. To conduct this study, a 1.502-MHz transducer transmitted HIFU through five *ex vivo* bovine tissue specimens in an MR-compatible tank setup. A needle hydrophone (aperture = 0.2 mm) scanned the acoustic pressure field orthogonal to the axis of propagation for three different positions of the tissue specimens (0, 4, and 8 deg, about the rotation mounts). To quantify the level of distortion, the distortion index (DI = 1—ratio between local and total acoustic energy fields) were calculated. The data collected confirmed HIFU distortion due to tissue, where 48% of the focal acoustic intensity was redistributed away from the focal field. Focal distortion decreased on average by 7.2% (ranging from 5.6% to 12.3%, SD = 2.8%) upon optimizing the transducer-specimen orientation the default transducer orientation (0 deg). In addition, the propagation simulation model accurately predicted HIFU distortion from a specimen's MRI data.

10:00—10:30 Break

10:30

4aBB7. The harmonic motion imaging for focused ultrasound system for tumor detection and treatment: Simulation, *in vitro* and *in vivo* results. Elisa Konofagou, Caroline Maleke, and Yi Hou (Dept. of Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave., New York, NY 10023, ek2191@columbia.edu)

Harmonic motion imaging (HMI) for focused ultrasound (HMIFU) is a novel, all-ultrasound-based system that can simultaneously detect and localize tumors as well as subsequently generate and monitor their ablation based on their distinct stiffness. In this paper, we present the results of a fundamental simulation study that is then validated using *in vitro* and *in vivo* findings. The same HMI and HMIFU parameters as in the experimental studies were used, i.e., the high-intensity focused ultrasound (HIFU) frequency was 4.68 MHz and the modulation frequency equal to 25 Hz. An HIFU-simulator was used to predict the lesion and its resulting time-dependent displacement field. Using the same parameters, *in vitro* bovine liver experiments were performed to validate the size of the simulated lesions. A transgenic mouse model of invasive adenocarcinoma was used for *in vivo* feasibility using the same parameters. Good agreement between the simulated lesion maps and pathology findings of the *in vitro* liver experiments was found. The lesion formation was identified by a 30% decrease in displacement amplitude in vivo. Tumor cell death was also confirmed by histology. Based on these results, the HMIFU system may offer a cost-efficient and reliable alternative for real-time monitoring of thermal ablation.

Contributed Papers

10:50

4aBB8. Controllable plasma membrane poration by pulsing tandem microbubble. Georgy Sankin, Fang Yuan, and Pei Zhong (Dept. of Mech. Eng. and Mat. Sci., Duke Univ., Box 90300, Durham, NC 27708)

Understanding the dynamic interaction of cavitation bubbles with biological tissue is central to the effective and safe application of therapeutic ultrasound in clinical medicine. Coupled oscillation of two laser generated microbubbles (maximum radius = 28 μm) in constrained media and associated shear stresses are investigated experimentally. Bubble-bubble interaction in a microchannel of 25-μm height is observed using high-speed video cameras and μPIV technique. Two liquid micro-jets moving in opposite directions can be generated when the second bubble is produced at the maximum size of the first one. The interaction of these tandem microbubbles with single cell leads to controllable poration of adjacent cell membrane and dye uptake. Micro-PIV data are compared with cell viability at various bubble-cell distances and azimuthal orientations. This method provides a new approach for highly selective cell treatment *in situ* applicable to targeted microinjection of macromolecules and gene vectors in microfluidics devices. [This work was supported in part by NIH.]

11:05

4aBB9. Ultrasound induced mechanical induction of mesenchymal stem cells. Jia-Ling Ruan, Yak-Nam Wang, and Stuart B. Mitchell (1013 NE 40th St., Seattle, WA 98105)

Low-intensity pulsed ultrasound (LIPUS) has been used to accelerate fracture healing and tissue regeneration but the biological mechanism of these responses is not completely understood. Stem cell activity can be induced through biochemical and mechanical mechanisms. Despite the common use of LIPUS in fracture healing and tissue regeneration, there are only a few studies that examine the mechanical induction of stem cells with ultrasound. The purpose of this study is to determine the effects of ultrasound-generated mechanical stimulus on the behavior of human mesenchymal stem cells (hMSCs) *in vitro*. In our preliminary studies low-intensity pulsed ultrasound was used to induce mechanical strains on hMSCs *in vitro*. Amplitudes, pulse durations, and pulse repetition frequencies were varied such that different radiation pressures were generated on hMSCs in culture. Results indicated a significant increase in cell proliferation after 4 consecutive days of treatment as well as a significant difference in the cellular response between treatment parameters. Results suggest that LIPUS can be used to influence mechanical mediated stem cell behavior; however, more research is needed to fully elucidate the relationship between ultrasound and hMSC response.

11:20

4aBB10. Local tissue-sparing due to blood flow in medium-size vessels. Simon Woodford, Ian Rivens, Gregory Vilensky, Nader Saffari, and Gail ter Haar (Dept. of Phys., Inst. of Cancer Res., Sutton, Surrey SM2 5NG, United Kingdom)

High-intensity focused ultrasound (HIFU) is a noninvasive surgical technique that uses ultrasound energy deposition to thermally destroy selected tissue. However, the cooling effect of blood flow can lead to unwanted tissue-sparing during HIFU treatment, particularly in the tissue immediately adjacent to the vessel wall. Regions where large (clinically detectable) vessels are present will require an increase in intensity or treatment time. However, there is a range of vessel sizes that are thermally relevant yet clinically undetectable. The treatment protocol must be chosen such that the spared region around these vessels is minimized or eliminated completely. Using numerical simulations, we have determined preliminary recommendations for suitable treatment protocols.

11:35

4aBB11. Dynamics and flow field produced by coupled oscillation of tandem microbubble. Fang Yuan, Georgy Sankin, Jaclyn Lautz, and Pei Zhong (Dept. of Mech. Eng. and Mat. Sci., Duke Univ., Durham, NC 27708)

Coupled oscillations of two adjacent laser-induced microbubbles have been shown to produce unique asymmetric bubble deformation and microjet formation. The resultant microstreaming and shear stress can cause localized cell membrane poration with potential application in targeted drug and gene delivery. In this study, we investigate the bubble dynamics and flow field produced by laser-generated tandem microbubble in a microfluidic device. Flow field around the tandem microbubble is analyzed with respect to phase delay, inter-bubble distance, and size ratio between the two microbubbles. In addition, micropatterning technique is used to control the adhesion site and growth pattern of HeLa cells in relation to the tandem microbubble. Flow vorticity is observed to be a key parameter that correlates with the strength of tandem microbubble oscillation and resultant macromolecule uptake efficiency. [Work supported by NIH Grant Nos. R01DK052985, R21CA135221, and S10RR016802].

11:50

4aBB12. Ultrasound-mediated nail drug delivery system to treat fungal disorders. Danielle Abadi and Vesna Zderic (Dept. of Elec. and Comp. Eng., The George Washington Univ., 801 22nd St. NW, Washington, DC 20052, zderic@gwu.edu)

This physiotherapy device treats nail fungal disorders by improving drug delivery to the nail bed using low frequency ultrasound to increase the permeability of the nail. Applying therapeutic ultrasound to the nail will allow pits to form on the surface through which more of the topical drug can be delivered. Transducers with corresponding matching networks are constructed in-house from 300-kHz and 1-MHz piezoelectric crystals. A software interface allows patients to select which toes to treat and at what intensity level (high, medium, or low). The highest-intensity setting supplies 1.5 W/cm^2 for a duration of 30 s to prevent damage to tissue below the nail. The software interface sends information to the electrical driving system,

which is composed of an amplifier for each intensity level and a signal generator. A Franz diffusion cell setup in combination with a spectrophotometer is used for experimental testing of nail permeability by measuring the amount of drug delivered through the membrane. Animal nails are used as a human nail model, and both drug-mimicking dye and Penlac (a topical prescription drug) are used for testing.

THURSDAY MORNING, 22 APRIL 2010 LAUREL A/B, 8:00 TO 11:45 A.M.

Session 4aEA

Engineering Acoustics: Sound Projection and Transduction

Stephen C. Thompson, Chair
Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Chair's Introduction—8:00

Contributed Papers

8:05

4aEA1. On the concept of transient directivity: Pipes and horns. Daniel Tengelsen, Brian E. Anderson, and Timothy W. Leishman (Dept. of Phys. and Astron., Acoust. Res. Group, Brigham Young Univ., Provo, UT 84602)

Directivity is a convenient way to represent how sound radiates from some arbitrary object in steady-state. The steady-state condition is implied when time harmonicity is assumed. Because all physical systems do not begin in steady-state, this directivity measurement is only valid after the transient portion of the solution has decayed. Transient directivity—a measure of sound radiation versus angle at a given instant in time and before the system has reached steady-state—is presented. Understanding an object's transient and steady-state radiation characteristics may be important in understanding the sound radiation from sources that are transient in nature. Results of transient directivity will be presented for pipes and horns from both numerical models and experimental measurements.

8:20

4aEA2. Modeling of transducer arrays for direct digital-to-analog conversion of signals. Jose Amado, Nikita Tkachov, and Charles Thompson (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, jose_amado@student.uml.edu)

A binary weighted array of speakers will be used to reconstruct a decomposed sequence of delta functions. The idea of directly converting digital signals to an analog acoustic output was first proposed by J. L. Flanagan in 1979, in which he designed, fabricated, and tested digital transducers for 4-, 5-, and 6-bit PCM signals. Flanagan found that at 6-bit resolution, the system fell short of good quality, and that condenser transducers had a limited output sound level of about 85 dB. Simulations will be used to investigate experimentally developed models for transducers and apply the direct digital-to-analog approach.

8:35

4aEA3. A new loudspeaker design for the enhancemenet of sound image localization on flat display panels. Gabriel Pablo Nava, Keiji Hirata, and Yoshinari Shirai (NTT Commun. Sci. Labs., NTT Co., Hikaridai 2–4, Seika-cho, Kyoto 619-0237, Japan, pablo@cslab.kecl.ntt.co.jp)

In most audio-visual multimedia applications, conventional stereo loudspeakers have been used to implement auditory displays. However, a fundamental problem with this kind of displays is that only the listeners situated at the sweet spot and over the symmetrical axis of the loudspeaker array are able to accurately localize the sound images. Although a number of audio signal processing algorithms have been proposed to expand the listening area, relatively less study on new loudspeaker configurations has been explored. This paper introduces a simple, yet effective, loudspeaker design to enhance the localization of sound images over the surface of flat display panels. In contrast to previous approaches, expansion of the listening space is achieved by attachment of rigid barriers which physically modify the sound radiation pattern of the loudspeakers. Moreover, numerical simulations, experimental sound measurements, and subjective tests have been preformed to validate a prototype of the proposed loudspeaker design using a display panel of an immersive teleconferencing system. Finally, an example of an interactive application was implemented involving real-time speaker tracking with a microphone and video cameras.

8:50

4aEA4. Head-tracking interface using a Wii remote. Megha Sunny, Ayse Kalkan-Savoy, and Charles Thompson (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, megha_sunny@student.uml.edu)

In this work, we will examine the problem of detecting the angular motion of head in effort to build a head-tracking system to control sound. The head motion occurs in *X*, *Y*, and *Z* linear directions and three rotation angles, namely pitch, roll, and yaw. In the past, we had difficulties to detect the rotation angles, especially yaw. Our current work focuses on the detection of rotation angles using a software interface program. The distance between sound source and head, and angular rotation data is collected by the Wii-remote's built-in optical sensor and three-axis accelerometer. These real-time data will be used as input to our software interface. Results will be used in conjunction with head related transfer function to create the three dimensional sound source effects.

9:05

4aEA5. The influence of matching layer material loss on radiation impedance conversion in ultrasonic transducers. Minoru Toda (Measurement Specialties Inc., 135 Gedney Rd., Lawrenceville, NJ 08648, minoru.toda@measspec.com)

PZT based thickness mode ultrasonic transducers for both air and water /tissue typically have a quarter wavelength front matching layer with impedance Zm. It is widely known that the lower acoustic impedance Z_R of the propagation medium is converted to a higher impedance at the PZT surface by the relation $Z\text{max} = Zm^2/Z_R$ (quarter wavelength conditions). In this work, the converted impedance was accurately calculated using a transmission line model incorporating the mechanical quality factor Qm of the matching layer material. In a medical transducer, it was found that the peak value is 20% or 28% lower than $Z\text{max} = Zm^2/Z_R$ for $Qm = 15$ or 10, respectively. For air transducers, the peak value is one to two orders lower than Zmax for the same range of Qm. These Qm values are typically observed for the filled epoxy matching layer materials used in medical transducers and also for porous or air entrained materials used for air acoustic transducers. A simple impedance conversion equation for an air transducer has been proposed, making the design of air impedance matching layers easier and suggesting that neglecting material loss leads to serious errors.

9:20

4aEA6. Transitory response of an acoustic levitator. Sahir Shakir and Charles Thompson (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, sahir_shakir@student.uml.edu)

This work will examine the transitory response of an acoustic levitator. Simulations will be used to ascertain the relationship between viscous and inertial forces on the vertical conveyance of a solid projectile. The objective is to use standing waves of strong intensity in a cavity to elevate a solid projectile. The projectile is suspended by nonlinear acoustic forces, and by rate of change in the frequency, amplitude, or cavity length, the transitory response will be determined.

9:35

4aEA7. Two driving constructions of loudspeakers for low-frequency range by piezoelectric ultrasonic motors. Juro Ohga (Ohga's Acoust. Lab., 2-24-3, Tamanawa, Kamakura, 247-0071, Japan), Takehiko Adachi, Hiroki Saito, Ryosuke Suzuki, Gen Takeda, Hajime Kubota (Chiba Inst. of Technol., Narashino 275-0016, Japan), Hirokazu Negishi (MIX Acous. Lab., Shiba, Minato-ku, Tokyo 105-0014, Japan), Kazuaki Maeda (TOA Corp., Takarazuka 665-0043, Japan), and Ikuo Oohira (Oohira's Lab., Aobadai, Aoba-ku, Yokohama 227-0062, Japan)

The loudspeaker driven by piezoelectric ultrasonic motors is characterized by a precise very-low-frequency reproduction due to its high-driving mechanical impedance. It has a lot of merits comparing to the conventional electrodynamic loudspeakers. One of the reason will be that this loudspeaker is a power flow modulator, not a transducer. In this presentation, two sorts of ultrasonic motors are compared as driver elements of the loudspeakers. One is an ordinary revolution-type motor and the other is a reciprocal linear motion type actuator. The authors constructed and improved practical low-frequency-range loudspeakers by using continuous revolution of ultrasonic motors. Its final model uses combination of two motors with same axis, which drive two cone radiators moving oppositely. This model shows a satisfactorily large output sound pressure and stable reproduction. However, its complicated elements for connection of the motors and the cone radiators cause a mechanical weakness. The authors, therefore, propose another new, completely different construction to avoid this defect. It applies linear motion of two ultrasonic actuators. A moving piece driven by piezoelectric ultrasonic vibrators is connected directly to a cone radiator. Comparison at various viewpoints and practical performance of these two constructions are presented at the meeting.

9:50

4aEA8. An ultrasonic vibrator constructed from laminated Galfenol steel. Scott P. Porter, Stephen C. Thompson, and Richard J. Meyer, Jr. (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA, 16804)

An ultrasonic vibrator has been developed to serve as the drive mechanism for an electroacoustic transducer. This design explores the unique characteristics of Galfenol, a recently invented giant magnetostrictive material. In addition to possessing competitive strain capabilities, strong mechanical properties, and a high-magnetic permeability, Galfenol does not require a prestress mechanism and can be laminated to effectively mitigate eddy current losses. Designing the vibrator required the authors to carefully engineer the magnetic circuit so that proper bias fields could be established using a permanent magnet. This step will be demonstrated with one- and two-dimensional models. Drive coil considerations will also be discussed and the fabrication and assembly of the vibrator will be shown along with in-air measurements.

10:05—10:15 Break

10:15

4aEA9. A preliminary analog circuit model of a balanced-armature transducer utilized for energy harvesting. Holly A. Smith and Stephen C. Thompson (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, has202@psu.edu)

This research investigates a balanced-armature transducer's potential to harvest ambient vibrational energy into electrical energy that could then be used to power small devices or recharge batteries. Such a device is desirable due to its compact size and environmentally friendly operation. An analog circuit model of a balanced-armature transducer was created in a version of SPICE. To ensure the model's accuracy, the electrical impedance predicted by SPICE was compared to experimental measurements. The model was then adjusted for energy harvesting. [Work supported in part by the Office of Naval Research.]

10:30

4aEA10. Modified single crystals for high-power, broadband underwater projectors. Nevin P. Sherlock and Richard J. Meyer, Jr. (Appl. Res. Lab., P.O. Box 30, State College, PA 16804)

When operating high-power electromechanical devices, the performance is often limited by self-heating within the active material. This is especially true in high-power, broadband underwater projectors using the piezoelectric single crystal $Pb(Mg_{1/3}Nb_{2/3}O_3$–$PbTiO_3$ (PMN-PT). Although PMN-PT crystals show an excellent piezoelectric response ($d_{33} > 1500$ pC/N) and high-coupling coefficient ($k_{33} > 0.90$), device performance is limited by the high-mechanical losses ($Q_M < 100$) and low-temperature phase transformation ($T_{RT} = 95$ °C). This work describes the material property enhancements of two compositional modifications and compares the performance of these crystals in high-power, broadband transducers. One such modification is the addition of $Pb(In_{1/2}Nb_{1/2}O_3$ (PIN) or $PbZrO_3$ (PZ) to the binary PMN-PT composition. The greater thermal stability of ternary PIN-PMN-PT and PZ-PMN-PT crystals is shown by comparing the dielectric permittivity, piezoelectric coefficient, and coupling coefficient to unmodified PMN-PT as a function of temperature. Additionally, the mechanical losses of PMN-PT have been decreased by doping with $Mn^{3,4+}$ ions. Using a laser Doppler velocimeter, the losses are evaluated under increasing ac electric drive. Using these data, high-power, broadband projectors were constructed from these modified crystals, and the results are compared to a projector using unmodified PMN-PT. [Funded by ONR under N00014-07-1-0336.]

10:45

4aEA11. Modeling thermal mitigation and nonlinear behavior in high-power single crystal 1-3 composite transducers. Tara L. Tubbs and Richard J. Meyer, Jr. (Appl. Res. Lab., State College, PA 16804)

The desire for high-precision sonar systems has forced 1-3 composite transducers to the forefront of sonar design. Single crystal, which has improved mechanical and dielectric properties over PZT, provides a variety of advantages, and their implementation into a 1-3 composite design makes them a great candidate for sonar transducers. Driving these transducers at high levels, we can get a broad bandwidth and high-power in a smaller device. However, high power introduces overheating and nonlinear behavior in the material properties. Using finite element software GID/ATILA from ISEN along with COMSOL MULTIPHYSICS, it is possible to incorporate these phenomena and solve thermal mitigation problems. This allows for improved high-power single crystal 1-3 composite transducers.

11:00

4aEA12. Use of compressively stressed zinc oxide to increase microspeaker response. Lukas Baumgartel (Dept. of Phys., USC MEMS Res. Group, 3737 Watt Way, PHE 621, Los Angeles, CA 90089, lbaumgar@usc.edu) and Eun Sok Kim (USC MEMS Res. Group, Los Angeles, CA 90089)

A micromachined piezoelectric speaker was fabricated on a 5×5-mm^2, 1-μm-thick silicon nitride diaphragm. A 4×4-mm^2 zinc oxide (ZnO) piezoelectric transducer sits in the middle of the diaphragm, providing actuation. Two variations were fabricated: one with the compressively stressed ZnO covering the region between the transducer and diaphragm perimeter—causing wrinkling—and another with the ZnO removed in this region. In both variations, the stress gradient causes curvature in the active area, raising the resonant frequency to above 4 kHz. The displacement response is therefore approximately flat from 40 Hz to 4 kHz. The speakers are driven with a sinusoidal voltage, and the response is measured with a laser interferometer. The wrinkled device exhibits 11 times larger response and can be actuated by much smaller voltage, achieving lower THD while still having a larger deflection. The wrinkled device is driven at 2 $V_{0.t.p.}$ from 40 Hz to 4 kHz, demonstrating a response of 55 nm/$V_{0.t.p.}$ and an average THD of 5.1%. The unwrinkled device is driven at 15 $V_{0.t.p.}$ over the same range, yielding a response of 5 nm/$V_{0.t.p.}$ and an average THD of 8.6%. Measured sound output and displacement spectra match each other well.

11:15

4aEA13. Analysis and design of a MEMS (microelectromechanical system) directional microphone diaphragm with active Q control. Ronald N. Miles, Quang T. Su, Weili Cui, Dorel Homentcovschi (Dept. of Mech. Eng., Binghamton Univ., Binghamton, NY 13902-6000, miles@binghamton.edu), and N. Eva Wu (Binghamton Univ., Binghamton, NY 13902-6000)

The analysis and design of a MEMS directional microphone are described that incorporates electronic feedback to achieve active Q control. The microphone diaphragm consists of a 1 × 2-mm stiffened plate fabricated out of polycrystalline silicon that is supported on a central hinge. The sound pressure gradient incident on the diaphragm produces a rocking motion about the central hinge. Interdigitated comb fingers at each end of the diaphragm enable both capacitive sensing and electrostatic actuation. The diaphragm has been designed to have its dominant resonant mode have a frequency of approximately 1 kHz. By minimizing sources of passive damping, the thermal noise of the microphone has been shown to be lower than the noise floor of existing two-microphone systems used in directional hearing aids [Miles *et al.*, J. Acoust. Soc. Am. **125** (2009)]. However, this low-passive damping also results in an undesirable resonance within the audible frequency range. To minimize the adverse effects of this resonance, a simple analog electronic feedback system is designed that can result in acceptable performance in both the frequency and time domains. [Work funded by NIH Grant R01 DC009429.]

11:30

4aEA14. Response of a MEMS (microelectromechanical systems) directional microphone diaphragm with active Q control. Quang T. Su, Ronald N. Miles, Weili Cui (Dept. of Mech. Eng., Binghamton Univ., Binghamton, NY 13902-6000, quang.su@binghamton.edu), Mihir Shetye (Solteras, City of Industry, CA 91748), and N. Eva Wu (Binghamton Univ., Binghamton, NY 13902-6000)

Measured results are presented that demonstrate the use of proportional and derivative electronic feedback to improve the performance of a directional microphone. The microphone diaphragm consists of a 1 × 2 mm stiffened plate fabricated out of polycrystalline silicon that is supported on a central hinge. The sound pressure gradient incident on the diaphragm produces a rocking motion about the central hinge. Interdigitated comb fingers at each end of the diaphragm enable both capacitive sensing and electrostatic actuation. The sound pressure gradient near the diaphragm has been measured by numerically differentiating the pressure measured by a probe microphone at locations around the diaphragm. The sound-induced motion of the diaphragm was measured using a laser vibrometer. From these measurements, estimates of the mechanical parameters of the diaphragm were obtained. By applying a known quasi-static voltage across the interdigitated fingers and measuring the resulting diaphragm deflection, an estimate for the derivative of the capacitance with respect to the displacement is obtained for the comb fingers of the diaphragm. Using these experimentally determined parameters and a linearized dynamic model of the system, the measured response of the microphone system with feedback is accurately predicted. [Work funded by NIH Grant R01 DC009429.]

THURSDAY MORNING, 22 APRIL 2010 — DOVER C, 8:40 A.M. TO 12:00 NOON

Session 4aED

Education in Acoustics and ASA Committee on Diversity: Diversity Issues in Education in Acoustics

Juan I. Arvelo, Cochair
Johns Hopkins Univ., Applied Physics Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099

Preston S. Wilson, Cochair
Univ. of Texas at Austin, Dept. of Mechanical Engineering, 1 University Station, Austin, TX 78712-0292

Chair's Introduction—8:40

Invited Papers

8:45

4aED1. "Future faces of physics" and other initiatives to broaden participation in science. Catherine O'Riordan and Kendra Rand (American Inst. of Physics, One Physics Ellipse, College Park, MD 20740, coriorda@aip.org)

Together, Hispanic-Americans and African-Americans make up over 25% of the US population, but they earn only 7% of physics bachelor's degrees. In order to help broaden the participation of underrepresented groups in STEM fields, the American Institute of Physics has several programs to work with students as well as to reach the general public. To engage physics undergraduates on the challenging subject of diversity, the Society of Physics Students (SPS), a society of over 4000 undergraduate physics students that is part of the American Institute of Physics (AIP), has launched two new programs. SPS has developed a Physics Jeopardy-like game as part of a "Future Faces of Physics" kit that includes various materials that help engage students in conversations about diversity. These kits have been used at more than 20 regional and national meetings since 2008 with positive results. AIP has also worked with Research Corporation to launch a program that brings Nobel laureates to HBCU's and to SPS campus or regional events that include participation by Minority Serving Institutions. Another AIP program creates short TV science news segments and distributes them free of charge in Spanish. These programs and future directions will be summarized.

9:05

4aED2. What can we learn from statistics on acoustics? Rachel Ivie (Statistical Res. Ctr., American Inst. of Physics, One Physics Ellipse, College Park, MD 20740, rivie@aip.org)

This talk will present current statistics on acoustics degrees and employment in acoustics. The data come from the National Science Foundation's studies of degrees awarded and from the American Institute of Physics' surveys of members of our ten Member Societies (including ASA). Because acoustics encompasses many different disciplines, these statistics will be compared to data from larger fields such as engineering, physics, and life and earth sciences. Although the numbers are very small in acoustics, data will be shown on the participation of under-represented minorities and women where possible. Finally, implications for increasing diversity in acoustics will be discussed.

9:25

4aED3. What faculty say, and convey, matters: Interactions with underrepresented students in science, technology, engineering, and mathematics. Sharon Fries-Britt (Univ. of Maryland, 2203 Benjamin Bldg., College Park, MD 20742, sfries@umd.edu)

Join us for this invited presentation as we examine students perceptions of their interactions with faulty. The participants in this study were primarily undergraduate students; however, approximately one-quarter of the participants were graduate students and post-doctoral research assistants. The entire sample consisted of students from a variety of postsecondary institutions including public and private, predominantly White, historically Black and Hispanic serving institutions from across the United States. This research is part of a larger study conducted over a 5-year period (2004–2009) with the National Society of Black Physicists (NSBP) and the National Society of Hispanic Physicists (NSHP). The majority are physics majors; however, some students are pursuing dual degrees in other STEM disciplines such as math, astronomy, and engineering. The findings of this study indicate that their interactions with faculty in the classroom and in advising session are critical. When those interactions are positive students benefit tremendously; however, in many instances they are negative and the interactions can cause barriers to their engagement in learning process and in how supported students feel pursuing science. Join us as we discuss some of the challenges and opportunities that these students encounter in their interactions with faculty.

9:45

4aED4. Diversity in physics: Whose problem is this? What can I do? Theodore W. Hodapp (Dept. Education and Diversity, American Physical Society, One Physics Ellipse, College Park, MD 20740, hodapp@aps.org)

Physics has one of the lowest participation rates for underrepresented minorities and women of all science, technology, engineering, and mathematics (STEM) fields. Things are improving for women and, while still not representative of the population, the trends are encouraging. Underrepresented minorities, however, have not been as fortunate. I will describe the current status of participation in physics and discuss new initiatives planned to address the lack of involvement at all levels in the field. In particular, I will describe a new program by the American Physical Society (APS) that aims to significantly increase the number of minorities who receive Ph.D.'s in physics. Actions you can take within your community, university, or workplace will be discussed. I anticipate a lively discussion during the panel that follows to bring forward good ideas and possible actions we can collectively take to improve participation of *all* people in physics and the technical workforce. The APS has partnered with leaders in the community to address these issues, and we hope that members of the Acoustical Society will find ways to work collaboratively to attend to these problems. This is our responsibility.

10:05—10:30 Break

10:30

4aED5. Diversifying physics: Legal context, education action. Daryl E. Chubin (AAAS Capacity Ctr., 1200 New York Ave., NW, Washington, DC 20005)

Science education and career development are vital for (1) driving innovation and economic strength, (2) US leadership in producing research & development and the personnel responsible for its renewal, and (3) shaping federal investments in what universities do and K-12 schools teach. Yet the nation faces a demographic challenge: by 2042, the population will be "majority minority." Minorities represent less than 7% of the nation's STEM workforce and are grossly underrepresented in undergraduate and graduate degrees awarded in STEM fields. In addition, the US is flagging in STEM degree production compared to the nations of Europe and Asia. Regardless of one's political beliefs, the nation must prepare more of its citizens for careers in science and engineering. This lecture focuses on the legal climate for increasing participation of underrepresented groups (women and minorities) in physics education and careers. Review of student- and faculty-centered programs and practices that have been "effective" in research universities face another challenged: Can they be made "legally sustainable"? How to make progress "on the ground" in the face of what the law calls—for disciplines such as physics and specialties such as acoustics—a "pipeline problem" will be discussed.

10:50

4aED6. Vassar College-Bronx Institute acoustics workshop for low-income, ethnic minority, urban high school students. David T. Bradley (Dept. Phys. and Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604-0745, dabradley@vassar.edu) and Angela M. Kelly (Lehman College, City Univ. of New York, Bronx, NY 10468)

Recent studies continue to show that low percentages of women and members of underrepresented ethnic groups are pursuing careers in science, technology, engineering, and math fields. Acoustics is no exception to this trend. As national demographics change more rapidly, and globalization transforms the way we approach the scientific process, the need for diversity becomes even more essential. The inclusion of women and minorities not only changes the questions being asked but also changes how those questions are answered. For acoustics to grow as a discipline, barriers to participation and success of members from underrepresented groups must be eradicated, and we must address the leakage in the science pipeline for students and professionals at all stages in their careers. One key area of concern is the recruitment and retention of K-12 students. This presentation will detail the joint efforts of the Vassar College Physics and Astronomy Department and the Bronx Institute at Lehman College to establish a hands-on, inquiry based acoustic workshop series for urban, low-income, underrepresented ethnic minority students from a collection of high schools in the Bronx borough in New York City.

11:10

4aED7. Reinventing diversity. Howard J. Ross (Cook Ross, Inc., 8630 Fenton St., Ste. 824, Silver Spring, MD 20910, howardr@cookross.com)

Despite all of the efforts to find strategies to improve the way organizations are addressing diversity, inclusion, and cultural competency issues, some are still finding their goals unrealized. In response to this, we conducted extensive research within the field and focused on how to reinvent diversity for the 21st century. Our research points to the need for three major paradigm shifts: (1) A movement from the classic United States-based approach, which focuses too heavily on race and gender and an assimilation model of diversity, to one that incorporates a deep understanding of globalism and the impact of major changes in population demographics around the world, global business, and interactive communication and networking; (2) a shift from the "good person/bad person paradigm" of diversity, which has developed and permeated a corrective mindset about diversity; (3) a 'find them and fix them' approach, which escalates the "us vs. them" way that people approach the issue and makes it more rather than less difficult to address. We have to move away from the event-based way for we have approached diversity, a pattern that has given us many specific activities, but not enough emphasis on systems thinking and culture-based change, to one that is strategic, systemic, and culture based.

11:30—12:00 Panel Discussion

THURSDAY MORNING, 22 APRIL 2010 GRAND BALLROOM III/IV, 8:30 TO 9:45 A.M.

Session 4aPAa

Physical Acoustics: Nonlinear Systems

Matthew E. Poese, Chair
Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Contributed Papers

8:30

4aPAa1. Dynamic stabilization of the ultradamped inverted pendulum. Randy Carbo (Acoust., Penn State Univ., University Park, PA 16802, rmc258@psu.edu), Matthew E. Poese, and Robert W.M. Smith (Appl. Res. Lab., University Park, PA 16802)

The dynamic behavior of the damped inverted pendulum is analogous to the modes of fluid systems with nonzero viscosity where a denser fluid is situated above a lighter fluid (e.g., the Rayleigh–Taylor instability). Thermoacoustic devices have such configurations whenever a hot heat exchanger is situated below a cold heat exchanger. It has been shown that such systems can be stabilized by effectively modulating gravity and can be described by a damped Hill equation. A numerical solver was developed that uses a stroboscopic technique to determine whether solutions are stable (bounded) or unstable (unbounded). While it is sometimes assumed that the stability boundaries of undamped systems bound the damped cases, for large values of damping ($Q < 0.05$, where Q is the quality factor), the numerical solution predicts that damping can destabilize the system for certain regions of the parameter space. Results of an experiment performed by the authors using a physical pendulum and eddy current damping to verify this rather counter-intuitive result are described. [Research supported by the Office of Naval Research and ARL Exploratory and Foundational Research Program.]

8:45

4aPAa2. Molecular dynamics of nonlinear and nonequilibrium effects in sound propagation. Takeru Yano (Dept. of Mech. Eng., Osaka Univ., Suita, 565-0871, Japan, yano@mech.eng.osaka-u.ac.jp)

Nonlinear and nonequilibrium effects on the propagation process of large amplitude and high-frequency sound waves are studied with the method of molecular dynamics, where the whole physical phenomenon is described in the numerical solution of Newton's equation of motion for hundreds of thousands of gas molecules. The frequency of sound studied here is so high that the wavelength should be comparable with the mean free path of gas molecules, and hence the continuum theory cannot be applied to the resulting phenomenon. The wave profiles are obtained by averaging the molecular motions, and the most conspicuous nonlinear effect in the result is the mass and energy transports by a shock-like wave, for which the origin can be attributed to a nonequilibrium effect caused by the high-frequency sound. The nonequilibrium effects are quantitatively examined with the velocity distribution function of gas molecules.

9:00

4aPAa3. Application of the chaotic cavity transducer concept for imaging in non-reverberant media. Koen Van Den Abeele, Bart Van Damme, Steven Delrue (Wave Propagation and Signal Processing, K.U. Leuven Campus Kortrijk, E. Sabbelaan 53, B-8500 Kortrijk, Belgium, koen.vandenabeele@kuleuven-kortrijk.be), and Olivier Bou Matar (Ecole Centrale de Lille, Villeneuve dAscq 59652, France)

The concept of chaotic cavity transducer utilizes a combination of a piezoelectric ceramic disk glued on a cavity of chaotic shape on the hardware side, with the time reversal (or inverse filter) technique on the software side. Using a two step procedure of direct excitation-reception and time reversed excitation-reception, it is possible to focus energy anywhere inside a non-reverberating sample, thanks to the ergodicity of the chaotic cavity. Moreover, the same basic concept can be used to create a virtual phased array using a single channel device. Both experimental data and simulations will be provided to illustrate the concepts. The goal is to use the chaotic cavity transducer concept to enhance the localization of micro-damage coupled to nonlinear elastic wave spectroscopy methods.

9:15

4aPAa4. Generation of sound by flow instabilities in a low-speed helium jet. W. C. Kirkpatrick Alberts, II and David A. Ligon (US Army Res. Lab., ATTN: RDRL-CIE-S, 2800 Powder Mill Rd., Adelphi, MD 20783)

Schlieren images of low-speed helium flow from a nozzle of circular cross section reveal instability in the jet occurring approximately 1 cm from the nozzle orifice. Accompanying this instability is a pronounced whistle. In this presentation, both qualitative and quantitative analyses of the observed phenomenon will be discussed.

9:30

4aPAa5. Parametrically excited oscillations: Where does the energy come from? Murray Strasberg (David Taylor Model Basin, 9500 MacArthur Blvd., West Bethesda, MD 20817-5000, murray.strasberg@navy.mil)

Recent discussions of parametrically excited oscillations seem to have overlooked an obvious question. How is energy transferred to a parametrically excited system to replace the energy lost to dissipative elements? The conventional analysis converts the linear differential equation representing the parametrically excited system into a Mathieu equation which is a lnear differential equation with a periodically varying coefficient, such as a periodically varying stiffness or length. This does not explain the physical mechanism involved in the energy transfer. To indicate how the energy is transferred, several systems are examined here, viz., (1) the simple pendulum where work is done when the length of the pendulum is varied periodically at half the period of the pendulum to increase the amplitude of its oscillation, (2) the taught string where work is done while sinusoidally varying the tension so as to excite transverse vibration at half the frequency of the varying tension, and (3) shape oscillations of gas bubbles in water excited by spherical volume pulsation at twice the frequency of the shape oscillation. These and other examples lead to the suggestion that more is required to excite parametric oscillation than being a solution of a Mathieu equation.

THURSDAY MORNING, 22 APRIL 2010 GRAND BALLROOM III/IV, 10:00 A.M. TO 12:00 NOON

Session 4aPAb

Physical Acoustics: Shock Wave and High Strain Rate Probes of Materials

Albert Migliori, Chair
Los Alamos National Lab., MS E536, Los Alamos, NM 87545

Chair's Introduction—10:00

4a THU. AM

Invited Papers

10:05

4aPAb1. Materials science at extreme conditions. James Belak (Condensed Matter and Mater. Div., Lawrence Livermore Natl. Lab., P.O. Box 808, L-45, Livermore, CA 94550, belak@llnl.gov)

The response of a material to an extreme stress transient is sensitive to the loading path and the rate of loading. An extreme example is following a phase transformation, during which the microscopic structure of the material completely rebuilds itself. This microstructure, to a large degree, determines the further response of the material. Traditionally, microstructures have been measured by slicing up the pieces and looking inside with methods such as scanning electron micrsocopy and transmission electron micrsocopy. However, at extreme conditions, such as following a phase transformation, one cannot pick up the pieces and must resort to non-destructive probes such as x-ray scattering. Here, we review recent attempts to determine shocked microstructures using non-destructive diffuse and small-angle x-ray scattering and compare to large-scale molecular dynamics simulations. The talk will anticipate future experiments at emerging light sources such as the Linac Coherent Light Source. [This work was performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.]

10:35—10:45 Break

10:45

4aPAb2. High-pressure magnetically driven compression waves in condensed matter. Jean-Paul Davis (Sandia Natl. Labs., P.O. Box 5800, MS-1195, Albuquerque, NM 87185, jpdavis@sandia.gov)

The *Z* machine is a fast pulsed-power machine at Sandia National Laboratories designed to deliver a 100-ns rise-time, 26-MA pulse of electrical current to *Z*-pinch experiments for research in radiation effects and inertial confinement fusion. Since 1999, *Z* has also been used as a current source for magnetically driven, high-pressure, high-strain-rate experiments in condensed matter. In this mode, *Z* produces simultaneous planar ramp-wave loading, with rise times in the range of 300–800 ns and peak longitudinal stress in the range of 4–400 GPa, of multiple macroscopic material samples. Control of the current-pulse shape enables shockless propagation of these ramp waves through samples 1–2 mm thick to measure quasi-isentropic compression response, as well as shockless acceleration of copper flyer plates to at least 28 km/s for impact experiments to measure ultra-high-pressure (~3000 GPa) shock compression response. This presentation will give background on the relevant physics, describe the experimental technique, and show recent results from both types of experiments. [Sandia is a multiprogram laboratory operated by Sandia Corporation, a Lockheed Martin Company, for the United States Department of Energy's National Nuclear Security Administration under contract No. DE-AC04-94AL85000.]

11:15

4aPAb3. The influence of shock loading on material properties and dynamic damage evolution. Ellen K. Cerreta, George T. Gray, III, Darcie D. Koller, Curt A. Bronkhorst, Carl P. Trujillo, and Benjamin L. Hansen (T-3, Los Alamos Natl. Lab., MS B216, Los Alamos, NM 87545)

Many commercial and defense applications require structural metals for extreme environments. Specifically, automotive, aerospace, and infrastructure applications need materials with damage tolerance during dynamic loading. To this end, many studies have examined dynamic deformation and damage evolution. These studies have shown that kinetics of loading are critically important to damage evolution of bulk metals. Particularly, in dynamic loading environments in which a shock wave is imparted to the metal, kinetic and spatial effects based on shock wave shape play important roles in damage. These studies also show that depending on crystal structure, shock loading can alter the subsequent properties of a material significantly. However, while these phenomena are gaining acceptance in the dynamic damage community, the ability to predict these phenomena is limited. Here, the influence of dynamic loading across strain rates 103–106/s will be discussed. The role of test platforms and crystallography to examine the influence of kinetics will be tied to changes observed in deformation and damage evolution. It can be shown that isolating the influence of spatial and kinetic effects during dynamic loading is critical to understanding dynamic damage evolution and with this understanding, capabilities for predicting dynamic damage evolution can be advanced.

Contributed Paper

11:45

4aPAb4. Acoustical shock formation in highly nonlinear fluids. Cristian Pantea and Dipen N. Sinha (Mater. Phys. and Applications, MPA-11, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Weak shock formation is investigated for plane sound waves of finite amplitude in fluids with high-acoustical nonlinearity. The shock formation length is related to the sound speed of the fluid *c* and its nonlinear parameter *B*/*A*. The use of fluids with low-sound speed and high parameter of nonlinearity has the advantage that the shock formation can be achieved at much lower pressures. Also, the experiments can be done on a smaller scale because the shock formation length is relatively small. The experiments are performed using short pulsed sound beams produced by a planar transducer with a resonance frequency of 0.5–1 MHz. The propagation medium consists of either different types of fluorocarbon or methanol. Direct pressure measurements of the acoustic waves in the fluid were obtained using a high-frequency calibrated hydrophone. For comparison, a relatively smaller acoustical nonlinear material like water is also investigated. Experimental results related to second and higher harmonics generated in the fluid and their evolution in time along the propagation axis are compared with theoretical time-domain predictions of the Khokhlov–Zabolotskaya–Kuznetsov equation.

Session 4aPP

Psychological and Physiological Acoustics and Musical Acoustics: Music Processing: Neural Mechanisms and Hearing Impairment

Xiaoqin Wang, Chair
Johns Hopkins Univ., Dept. of Biomedical Engineering, 720 Rutland Ave., Traylor 410, Baltimore, MD 21205

Invited Papers

8:00

4aPP1. Individual differences reveal the basis of consonance. Josh H. McDermott (Ctr. for Neural Sci., New York Univ., 4 Washington Pl., New York, NY 10003, jhm@cns.nyu.edu) and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E. River Rd., Minneapolis, MN 55455)

Some combinations of musical notes are consonant (pleasant), while others are dissonant (unpleasant), a distinction central to music. Explanations of consonance in terms of acoustics, auditory neuroscience, and enculturation have been debated for centuries. These debates have remained largely unresolved, in part, because the various theories are difficult to distinguish with conventional methods. This talk will describe our recent studies applying the method of individual differences to this problem. We measured preferences for musical chords as well as nonmusical sounds that isolated particular acoustic factors, including the beating and harmonic relationships between frequency components. Listeners preferred stimuli without beats and with harmonic spectra, but across over 250 subjects, only the preference for harmonic spectra was consistently correlated with preferences for consonant over dissonant chords. Harmonicity preferences were also correlated with the number of years subjects had spent playing a musical instrument, suggesting that exposure to music amplifies preferences for harmonic frequencies because of their musical importance. Preferences for stimuli lacking beats, in contrast, were not correlated with musical experience. Harmonic frequency relations figure prominently in many aspects of hearing, and our results indicate that they also underlie the perception of consonance. [Work supported by NIH Grant R01DC05216.]

8:20

4aPP2. Music on more than one note: Pitch perception and neural coding of concurrent harmonic tones. Christophe Micheyl and Andrew, J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455-0344, cmicheyl@umn.edu)

The basis of Western music lies in the combination of simultaneous (harmony) and sequential (melody) harmonic complex tones or notes. Listeners must "hear out" simultaneous pitches and "track" pitch sequences over time. Surprisingly few psychoacoustic studies have studied pitch perception under such "natural" conditions of more than one note at a time. Similarly, the neural mechanisms that support these abilities remain poorly understood. Here, we will review psychoacoustical and neurophysiological findings, which concur to suggest an important role for frequency selectivity in the ability of the auditory system to segregate concurrent notes. In particular, the ability to accurately discriminate changes in the fundamental frequency or, subjectively, pitch of a "target" harmonic complex tone in the presence of a "masker" tone occupying the same spectral region seems to covary with the limits peripheral frequency selectivity. Given that harmonic complex tones are an important class of sound for both music and speech and that frequency selectivity is usually adversely affected by cochlear damage, such results may have important implications for the current understanding of auditory scene analysis of concurrent music and speech sounds and of the listening difficulties experienced by hearing-impaired individuals. [Work supported by NIH Grant R01 DC05216.]

8:40

4aPP3. Auditory and tactile integration in music meter perception. Juan Huang (Dept. of Neurosci. and Zanvyl Krieger Mind/Brain Inst., The Johns Hopkins Univ., Baltimore, MD 21218 and Dept. of Intelligence Sci., Key Lab. of Machine Percept., Peking Univ., Beijing 100871, China), Darik Gamble, Xiaoqin Wang, and Steven Hsiao (Dept. of Neurosci., Dept. of Biomedical Eng., and Zanvyl Krieger Mind/Brain Inst., The Johns Hopkins Univ., Baltimore, MD 21205)

Meter is a fundamental temporal structure of music. The perception of meter is typically inferred from the occurrence of accents in the music surface. Under normal listening conditions, listening to or playing music is usually accompanied by vibro-tactile input which we hypothesize contributes to meter perception. Previous studies have shown that beat perception can occur via purely tactile stimulation. Whether vibro-tactile stimulation can give rise to meter perception and how it interacts with auditory meter perception is unknown. Here we used accent occurrence and strength as cues to study meter perception in subjects performing auditory only, tactile only, and combined auditory-tactile psychophysical discrimination tasks. We find that subjects can perceive meter through purely auditory only and tactile only stimulation. Furthermore, when stimuli were ambiguous, tactile stimulation was found to enhance meter perception when subjects were given weak auditory amplitude-accented cues. Similarly, auditory stimulation enhanced meter perception when subjects were given weak tactile amplitude-accented cues. These results indicate that meter perception is processed cross-modally, and that auditory-tactile integration plays an important role in the neural representation of temporal structures in music.

9:00

4aPP4. Imaging and inducing disorders in musical perception and production. Psyche Loui (Dept. of Neurology, Beth Israel Deaconess Medical Ctr. & Harvard Med. School, 330 Brookline Ave., Palmer 127, Boston MA)

To perceive and produce music accurately, the brain must represent, categorize, plan, and execute pitched information in response to environmental stimuli. Convergent methods from psychophysics, neuroimaging, and noninvasive brain-stimulation with normal and tone-deaf (TD) subjects were employed to show that neural networks controlling pitch perception and production systems include bilateral frontotemporal networks. First, psychophysical data showed that the perception and production of pitch are uncorrelated in TD subjects, suggesting a disconnection between perception and production brain regions. This disconnection was extended in a diffusion tensor imaging study in TD and control subjects: tractography revealed that the arcuate fasciculus, which connects temporal and frontal lobes, is reduced in TD subjects, especially in its superior division in the right hemisphere. This disconnection highlights the importance of frontotemporal interactions in music processing. Finally, to reverse-engineer the perception-production network, transcranial direct current stimulation was applied over superior temporal and inferior frontal regions. Results showed diminished accuracy in pitch matching after stimulation compared to sham control. Taken together, results demonstrate that intact function and connectivity of a distributed cortical network, centered around bilateral superior temporal and inferior frontal regions, are required for efficient interactions with sounds in the environment. [Supported by NIDCD.]

9:20

4aPP5. Neural substrates of spontaneous musical improvisation. Charles J. Limb (Dept. of Otolaryngol.-Head and Neck Surgery, Johns Hopkins Hospital, 601 N. Caroline St., Baltimore, MD 21287)

To investigate the neural substrates that underlie spontaneous musical performance, functional MRI was used to study improvisation in professional jazz pianists. The purpose of the study was to identify the neural substrates that give rise to spontaneous musical creativity, defined as the immediate, on-line improvisation of novel melodic, harmonic, and rhythmic musical elements within a relevant musical context. It was hypothesized that spontaneous musical improvisation would be associated with discrete changes in prefrontal activity that provide a biological substrate for actions that are characterized by creative self-expression in the absence of conscious self-monitoring. By employing two paradigms that differed widely in musical complexity, it was found that improvisation (compared to production of over-learned musical sequences) was consistently characterized by a dissociated pattern of activity in the prefrontal cortex: extensive deactivation of dorsolateral prefrontal and lateral orbital regions with focal activation of the medial prefrontal (frontal polar) cortex. Such a pattern may reflect a combination of psychological processes required for spontaneous creative behaviors such as improvisation, in which internally motivated, stimulus-independent behaviors unfold in the absence of central processes that typically mediate self-monitoring.

9:40

4aPP6. Musical experience impacts hearing speech in noise. Nina Kraus (Auditory Neurosci. Lab., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, www.brainvolts.northwestern.edu)

Musical experience profoundly impacts how sound is transcribed by the nervous system. This influence is likely mediated by cognitive processes such as attention and memory through the corticofugal system [Tzounopoulos and Kraus, Neuron **62**, 463–469 (2009)]. Hearing in noise is difficult for everyone but especially for children with developmental dyslexia and older adults. We have identified objective neural signatures—from the human auditory brainstem—that reflect hearing in noise [Chandrasekaran *et al.*, Neuron **64**, 311–319 (2009) and Hornickel *et al.*, Proc. Natl. Acad. Sci. U.S.A. **31**, 13027 (2009)]. Musicians develop the ability to hear relevant signals embedded in a network of melodies and harmonies. This ability transfers to hearing a target speaker's voice in background noise. We are beginning to understand the biological basis for this perceptual advantage [Parbery-Clark *et al.*, J. Neurosci **29**, 14100–14107 (2009)]. Sensory processing of speech and music is tightly coupled with the cognitive abilities that underlie language and musical expertise; this knowledge can be used to advantage in the consideration of educational and remediation strategies. [Work supported by NSF SGER 0842376.]

10:00—10:15 Break

10:15

4aPP7. Measuring and predicting the quality of nonlinearly distorted music and speech as perceived by hearing-impaired people. Chin-Tuan Tan (Dept. of Otolaryngol., School of Medicine, New York Univ., 550 First Ave., NBV 5E5, New York, NY 10016), Brian C. J. Moore (Univ. of Cambridge, Cambridge CB2 3EB, United Kingdom), and Mario Svirsky (School of Medicine, New York Univ., New York, NY 10016)

The goals of this study were to characterize and model the perception of nonlinearly distorted speech and music by hearing-impaired listeners. Hearing-impaired listeners were asked to rate the perceived quality of speech and music that had been subjected to various forms of nonlinear distortion, some of which are inherent to certain hearing aid designs including (1) hard and soft, symmetrical and asymmetrical clipping; (2) center clipping; (3) "full-range" distortion, produced by raising the absolute magnitude of the instantaneous amplitude of the signal to a power (8800;1), while preserving the signal of the amplitude; (4) automatic gain control (AGC); (5) output limiting. Stimuli were subjected to frequency-dependent amplification as prescribed by the "Cambridge formula" before presentation via Sennheiser HD580 earphones. The pattern of the rating was reasonably consistent across subjects with only two of ten subjects not making consistent ratings. The mean ratings were not lower with increasing amount of soft or center clipping or when the compression ratios of the AGC and output limiting were increased. The deleterious effects produced by these nonlinear distortions may have been offset by the beneficial effects of improving audibility and compensating for loudness recruitment. [Work supported by Deafness Research Foundation.]

10:35

4aPP8. Music through hearing aids: Perception and modeling. Kelly Fitz and Martin F. McKinney (Starkey Labs., 6600 Washington Ave. S, Eden Prairie, MN 55344, firstname_lastname@starkey.com)

Historically, the primary focus of hearing aid development has been on improving speech perception for those with hearing loss. Modern-day hearing-aid wearers, however, face many different types of acoustic signals, such as music, that require different types of processing. Music signals differ from speech signals in a variety of fundamental ways, and relevant perceptual information is conveyed via different signal attributes in the two types of signals. The research described here is an effort to improve music perception in listeners with hearing impairment. First, methods have been developed to quantitatively measure deficits in music perception for impaired and aided listeners. Second, specific perceptual features have been evaluated as to their relative importance in the successful perception of music and that information has been used to guide signal processing development. Finally, the relevant perceptual features have been modeled, and the models have been used to evaluate and compare signal processing algorithms designed to improve music perception through hearing aids. An overview of our research will be presented along with key recent results.

Contributed Papers

10:55

4aPP9. A novel physiological mechanism for pitch encoding in monkey primary auditory cortex. Yonatan I. Fishman (Dept. of Neurology, Albert Einstein Coll. of Med., 1300 Morris Park Ave., Bronx, NY 10461, yonatan.fishman@einstein.yu.edu) and Mitchell Steinschneider (Albert Einstein Coll. of Med., Bronx, NY 10461)

Pitch is a fundamental perceptual attribute of hearing. While auditory cortex is implicated in pitch perception, how pitch is represented at the cortical level remains unclear. The present study examines a novel hypothesis for how the pitch of pure tones and of harmonic complex tones, with or without the fundamental frequency, is encoded in primary auditory cortex (A1): pitch is represented non-topographically by the temporal distribution of population activity in A1. Sounds of lower pitch evoke a greater proportion of sustained multiunit activity (MUA), relative to initial onset MUA, than sounds of higher pitch, such that the temporal distribution of MUA systematically varies with pitch. Pure tones and harmonic complexes with the same pitch evoke a similar proportion of sustained MUA. The temporal distribution of MUA is largely invariant to changes in stimulus parameters (e .g., level and relative phase) that leave the perceived pitch unchanged. Timing of AEP components recorded in superficial layers of A1 parallels similar pitch-related changes in AEPs recorded in humans. Coding of perceptual qualities based on the time course of neural activity has been proposed in other sensory modalities (e.g., olfaction) and offers a novel alternative to topographic representations of pitch at the cortical level.

11:10

4aPP10. Neural correlates of sensory consonance and dissonance in primate primary auditory cortex. Mitchell Steinschneider (Dept. of Neurosci., Albert Einstein Coll. of Med., 1300 Morris Park Ave., Bronx, NY 10461, mitchell.steinschneider@einstein.yu.edu) and Yonatan I. Fishman (Albert Einstein Coll. of Med., Bronx, NY 10461)

The perception of consonance and dissonance of isolated chords (sensory consonance/dissonance) is fundamental to music appreciation. Consonant chords are composed of tones related to each other by simple frequency ratios (e.g., perfect fifth 3:2), whereas dissonant chords are composed of tones related to each other by complex ratios (e.g., minor second 256:243). Dissonance is thought to be due to the perception of beats (modulation frequencies < 20 Hz) or roughness (modulation frequencies from 20–250 Hz), which occur when two or more components of a complex sound are separated from one another in frequency by less than the width of an auditory filter (i.e., critical bandwidth). These unresolved frequency components interact in the auditory periphery to produce amplitude-modulated (AM) fluctuations in the composite waveform envelope. We demonstrate that the magnitude of neuronal phase-locking to these AM fluctuations in primary auditory cortex of awake monkeys correlates with the perceived consonance/dissonance of musical chords and parallels human perception of roughness. This correlation is displayed by population activity, as measured by auditory evoked potentials (AEPs), current source density, and multiunit activity. We further demonstrate that phase-locking of AEPs in Heschl's gyrus of humans is similar to that seen in the monkey.

11:25

4aPP11. A biomimetic multi-resolution spectrotemporal model for musical timbre recognition. Kailash Patil and Mounya Elhilali (Dept. of Elec. & Comput. Eng., 3400 N. Charles St., Barton 105, Baltimore, MD 21218)

Timbre is usually defined as that property of sound that is neither pitch nor loudness that helps in identifying sounds. Such a definition of *what it is not* is in itself problematic and unsatisfactory. The complexity of characterizing a timbre space stems from the intricate interactions between spectrotemporal dynamics of sound, which overcasts the simple description of individual acoustic dimensions as usually captured by techniques such as multidimensional scaling. By contrast, sound encoding in the mammalian auditory system, and particularly its sensitivity to spectrotemporal modulations, offers a rich feature space to explore perceptual representations of timbre. In the present work, the problem of musical timbre modeling was casted as a generative model based on a reduced set of multi-resolution spectrotemporal features inspired from encoding of complex sound in the auditory cortex. A probabilistic system based on Gaussian mixtures was built to perform a timbre recognition task on a data set consisting of 12 instruments. The system was able to achieve an accuracy of 98%, hence corroborating the claim that physiologically inspired features are effective in properly outlining a timbre space. These results could have significant implications in terms of better understanding the neural correlates of timbre perception.

Session 4aSA

Structural Acoustics and Vibration: Applications of Structural Acoustics and Vibration I

Robert M. Koch, Chair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841-1708

Contributed Papers

9:00

4aSA1. Comparison of techniques of acoustically interrogating a fluid-filled pipe. Curtis F. Osterhoudt, Christopher Dudley, and Dipen N. Sinha (MPA-11, Los Alamos Natl. Lab., Los Alamos, NM 87545, cfo@lanl.gov)

Through-transmission of sound in a fluid-filled pipe was experimentally investigated, with comparison to a one dimensional computational model. Pipe curvature is shown to be a contributor to deviations from the model, as is dynamical interaction between the finitely yielding pipe walls and the fluid contents. Various techniques of acoustically interrogating the system are considered. These include the different information which may be extracted via continuous-wave acoustical excitation, swept-frequency interferometry, and tone- and chirp-burst excitations. Each of these techniques has their advantages and disadvantages, especially in situations where short-time measurements must be made, and a version of the gain-bandwidth product must be taken into account. Some algorithms for automatically extracting fluid sound-speed are discussed, and it is shown that no one dominates under all circumstances. [This work was supported by Chevron USA.]

9:15

4aSA2. Sonar-induced strains and stresses in an absorbing solid elastic sphere. Kenneth G. Foote (Woods Hole Oceanograph. Inst., Woods Hole, MA, 02543), David T. I. Francis (Univ. of Birmingham, Edgbaston, Birmingham B15 2TT, United Kingdom), Mario Zampolli, and Tim van Zon (TNO Defense, Security and Safety, 2509 JG The Hague, The Netherlands)

The interaction of harmonic plane and spherical pressure waves with an absorbing solid elastic sphere is modeled. The analytic, boundary-element-method (BEM) and finite-element-method (FEM) solutions for the internal displacement, strain, and stress fields are evaluated numerically for 50-mm-diameter solid spheres of poly(methyl methacrylate) (PMMA) and high-density polyethylene (HDPE) in an immersion medium of mass density 1000 kg/m^3 and sound speed 1500 m/s. The mass density and longitudinal and transverse sound speeds of the PMMA sphere are 1191 kg/m^3, 2690 m/s, and 1340 m/s, respectively. The corresponding properties of the HDPE sphere are 957 kg/m^3, 2430 m/s, and 950 m/s. The two materials are assumed to have a hysteresis-type absorption, hence with constant product of absorption coefficient and wavelength. The respective values of this product for longitudinal and shear waves for PMMA are assumed to be 0.19 and 0.29 dB, and for HDPE, 0.40 and 1.20 dB. Each of two frequencies is considered, 10 and 100 kHz, for which the wavenumber-radius product is $\pi/3$ and $10\pi/3$, respectively. Results for the three solution methods are compared. [Work partly supported by NOPP through ONR Award No. N000140710992.]

9:30

4aSA3. On the cancellation of acoustic waves scattered from an elastic sphere. Matthew D. Guild (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, mdguild@arlut.utexas.edu), Andrea Alù, and Michael R. Haberman (The Univ. of Texas at Austin, Austin, TX 78713-8029)

Recent research has suggested the possibility of creating non-absorptive elastic covers that eliminate the acoustic field scattered from an elastic object, also known as acoustic cloaks [Norris, Proc. R. Soc. London, Ser. A **464**, 2411 (2008)]. The work presented here employs the scattering cancellation technique [Alù and Engheta, Phys. Rev. E **72**, 016623 (2005)] to investigate the effectiveness of a single isotropic elastic layer to cloak an elastic sphere. The presentation discusses the benchmarked analytical and finite element scattering models which were employed to explore the design space of the cloaking layer. Parametric studies showing the influence of cloak stiffness and geometry on the frequency dependent scattering cross section are then presented. These case studies clearly illustrate the fundamental physical behavior leading to the observed reduction in scattering cross section at design frequencies. Finally, material selection and the creation of composite materials required to produce a functional scattering cancellation layer are presented and discussed.

9:45

4aSA4. Effective mass density of a random configuration of cylinders. Francine Luppé (LOA-GOA, FRE CNRS 3102, Univ. of Le Havre, pl.R. Schuman, 76610 Le Havre, France, francine.luppe@univ-lehavre.fr), Jean-Marc Conoir (Univ. Paris 6, 75005 Paris, France), and Pascal Pareige (Univ. of Le Havre, 76610 Le Havre, France)

The dynamic effective mass density of a random distribution of $n0$ cylinders/m^2 in an ideal fluid is looked for. The Fikioris and Waterman approach is used to obtain the reflection coefficient of the random half-space at the plane interface with the ideal fluid. This coefficient is expanded into powers of $n0$, using Linton and Martin's expansion of the wavenumber of the coherent wave. The reflection coefficient is then compared to that obtained when a homogeneous viscous fluid replaces the random medium. When the two reflection coefficients are equal, the random fluid is acoustically equivalent to the viscous one, which is thus considered as the effective fluid. The coherent wave in the random medium is described as the acoustic mode in the effective fluid, with the shear viscosity of the latter being set to zero. Equating the two reflection coefficients provides an effective mass density that depends on frequency and on the incidence angle, except at low frequency. The angle dependence is discussed.

10:00

4aSA5. Coherent backscattering enhancement in cavities. Simple shape cavity revisited. Stefan Catheline, Thomas Gallot, Philippe Roux, Guillemette Ribay, and Julien de Rosny (Laboratoire de Gophysique Interne et Tectonophysique (LGIT), CNRS Universit de Grenoble, France, stefan.catheline@ujf-grenoble.fr)

Coherent backscattering effect (CBE) is classically introduced in disordered, random, or chaotic media. In this work, the attention is focused on simple parallepipedic cavities since, contrarily to a widespread idea, CBE can also be observed for a pure-tone source in a one-dimensional (1-D) cavity. This approach is of two-fold interest. First, analytical computations predict a dimensional dependence of the coherent backscattering enhancement according to a $^{(3/2)}d$ law, d being the dimensionality of the cavity, that have not yet been compared to experiments. Second, it opens a new ballistic interpretations for which each multiply reverberated path is associated with more (rectangle and parallelepipedic cavities) or less (1D cavity) than one single reciprocal counterpart. This paper is the first of two, the second paper dealing with some impacts of symmetry on CBE.

10:15—10:30 Break

10:30

4aSA6. Coherent backscattering enhancement in cavities. Highlight of the role of symmetry. Thomas Gallot, Stefan Catheline, and Philippe Roux (Laboratoire de Gophysique Interne et Tectonophysique (LGIT), CNRS & Universit de Grenoble, France, thomas.gallot@obs.ujf-grenoble.fr)

Through experiments and simulations, some consequences of symmetry on coherent backscattering enhancement (CBE) in cavities are reported here. First, CBE outside the source is observed (a) on a single symmetric point in a one dimensional cavity, in a tw0-dimensional (2-D) circular membrane, and in a 2-D symmetric chaotic cavity; (b) on three points in a rectangle; and (c) seven in a parallelepiped. Second, existence of enhanced intensity lines and plans in 2-D or three-dimensional cavities is demonstrated. Third, we show how antisymmetry is responsible for the existence of a coherent backscattering decrement with a dimensional dependence of (12)d law, *d* being the dimensionality of the cavity.

10:45

4aSA7. How does the scattering from an empty cylinder change with the level of filling? Duncan P. Williams (Dstl Physical Sci., Porton Down, Salisbury SP4 0JQ, United Kingdom), Richard V. Craster (Univ. of Alberta, Edmonton T6R 2Y9, Canada), and Samuel D. M. Adams (Imperial College, London SW7 2BZ, United Kingdom)

The acoustic scattering from elastic objects features in many tasks stretching from the use of sonar to search for underwater objects such as submarines and mines to the inspection of shipping containers and cargo screening. The properties of canonical objects, such as cylindrical and spherical shells, which are completely filled with fluid, or immersed in fluid, have been widely studied, but typically stop short of studying shells containing different levels of filling. Not knowing the extent of the effect of filling on the scattering from different objects can limit how well one can discriminate between similar objects or inspect the interior of objects and, for example, find contraband or other suspicious materials. This paper looks at the two-dimensional response of a partially filled elastic cylinder. A computational method to model non-uniform domains is introduced and the use of perfectly matched layers is discussed. Results are shown for combinations of cylinders and different levels of filling. In particular, we show how the response of the cylinder depends on the level of the filling. The results are used to comment on the usefulness, or otherwise, of the response to estimate the nature and level of filling based on short- or long-range observations.

11:00

4aSA8. Modeling of a two-transducer through-wall ultrasonic communication system. Sebastian Roa Prada, Kyle R. Wilt, Henry A. Scarton, Gary J. Saulnier, Jonathan D. Ashdown, Pankaj K. Das, and Andrew J. Gavens (Dept. of Mech. Eng., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, sebastian-roa27@hotmail.com)

Ultrasound at 1 MHz is used as a carrier of information across solid walls without penetration. A communication channel is created by bonding two transducers on either side of a solid wall. The outside transducer transmits a continuous wave generating an acoustic field. The inside electronics are powered by harvesting the electrical output of the inside transducer. Digitized bits, representing the value of an analog signals measured on the inside (such as temperature), are used to alternate the electrical load of the inside transducer between two finite values. Changes in the electrical load of the inside transducer modify its acoustical impedance as seen by the incident waves coming from the wall, which in turn modulates the amplitude of the reflected signal. This modulated wave is detected at the electrical terminals of the outside transducer, where it is then demodulated to recover the data. The mechanical components of the system are modeled in connection to the electronic circuits by means of electro-mechanical analogies. Simulation of the communication system is performed using the electric circuit simulation package PSPICE. Simulation, finite element solutions, and experimental results are presented and discussed. Digital data communication rates exceeding 50 000 bits/s are achieved.

11:15

4aSA9. Relationships between structural energy density, power flow, and their influence on acoustic intensity. Jeff M. Fisher, Dan A. Manwill, Jon D. Blotter, Scott D. Sommerfeldt, and Kent L. Gee (Dept. of Mech. Eng., Brigham Young Univ., 435 CTB, Prov, uT jefffish@nm.byu.edu)

Power flow in structures has been a topic of research for the past few decades. Many different methods for determining structural power flow magnitude and direction have been developed and proven. Structural energy density methods have likewise been developed. In this paper, a study of both structural energy density and power flow in a plate and their influence on the acoustic field is presented. Structural energy density and power flow were determined using equations typically used with accelerometer arrays but adapted for use with a scanning laser doppler vibrometer. Experiments were performed from 25 to 100 Hz by exciting the plate at single frequencies but using different excitation points and methods at each frequency. In some cases two sources of excitation were used to alter the power flow response, giving a broader basis for concluding relationships. Both speaker and shakers were used as sources. The acoustic intensity in the room was calculated at approximately 3 in. from the vibrating plate. Relationships between structural energy density, power flow, and the acoustic field variables are presented.

11:30

4aSA10. Real-time active control of structural energy density and structural power flow. Daniel A. Manwill (dmanwill@byu.net), Jeff M. Fisher (Dept. of Mech. Eng., Brigham Young Univ., 435 CTB, Provo, UT 84602), Scott D. Sommerfeldt, Kent L. Gee, and Jonathan D. Blotter (Brigham Young Univ., Provo, UT 84602)

Structural energy density and structural power flow have long been used as metrics in the active control of vibrating structures. The greater portion of this previous work has focused on frequency-domain methods which incorporate assumptions about the relative contributions of near-field and far-field energy components. This paper describes the implementation of filtered-x-based time-domain control schemes which utilize 9- and 13-accelerometer arrays to estimate and control structural energy density and structural power flow, respectively. Experiments were performed on a clamped steel plate excited and controlled by various combinations of loudspeakers and electrodynamic shakers in a frequency range from 25 to 100 Hz. Analog circuitry was used to estimate spatial derivatives and reduce channel count. The development of control laws incorporating the effects of the analog circuitry is presented. Control attentuation results are given, sensor placement is discussed, and implementation challenges are addressed. [This work is supported by NSF Grant 0826554.]

Session 4aSC

Speech Communication: Speech Perception and Perceptual Adjustment (Poster Session)

Cynthia Clopper, Chair
Ohio State Univ., Dept. of Linguistics, 1712 Neil Ave., Columbus, OH 43210

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors or odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors or even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

4aSC1. Perceptual and production training of allophones and phonemes in Spanish. Wendy Herd (Dept. Linguist., Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66044, wenherd@ku.edu)

American English learners of Spanish often do not acquire native-like pronunciation of intervocalic /d/, tap, and trill in words like CODO "elbow," CORO "choir," and CORRO "I run." The trill proves difficult because it does not exist in English. Although the tap exists as an allophone of /t/ and /d/ in English, students of Spanish must learn to process it as a phoneme rather than an allophone. Similarly, learners have difficulty acquiring the spirantization of voiced stops, where intervocalic /d/ is produced as a voiced dental fricative or approximant. This study investigates whether American English learners of Spanish can be trained to perceive and produce the intervocalic /d/, tap, and trill contrasts in Spanish. Both perceptual and production training methods were used. Past research has reported that perceptual training alone improves both perception and production, and that production training alone improves both as well; however, the production training studies have not been limited to production as trainees have been able to listen to the training stimuli. This study systematically controls both training modalities and introduces a third training methodology that includes both perception and production to discover whether perceptual, production, or combination training is most effective. [Research supported by NSF.]

4aSC2. Production and perception of Taiwan Mandarin syllable contraction. Grace Chen-Hsiu Kuo (Dept. of Linguist., Phonet. Lab., UCLA, Los Angeles CA 90095, gracekuo@humnet.ucla.edu)

Taiwan Mandarin syllable contraction is an optional lenition process which involves the elision of the intervocalic segments and the merger of the tonal elements of two syllables. Here it is shown that syllable contraction is gradient and non-neutralizing. In a production experiment, 20 subjects read a list of minimal sentence pairs, containing disyllabic contractable words and matched monosyllabic lexical words, at two speech rates (88 and 144 beats/min), three times each. Degree of contraction was measured as the depth of the intensity trough between two syllables [Mermelstein 1975], with a trough depth (TrD) of zero meaning fully contracted. 8% of disyllables were somewhat contracted (TrD between 0 and 2 dB) while 29% were fully contracted (TrD = 0 dB). Fully contracted disyllables were compared on several other acoustic measures to their monosyllabic counterparts and were found to differ most strongly in duration. In a perception experiment, items from the production experiment were presented to 35 listeners for forced-choice identification as disyllabic or monosyllabic words. Accuracy was generally high; reaction times were slower for lexical monosyllables, which were sometimes labeled as contracted. Thus, even when fully contracted syllables were produced, they remained acoustically and perceptually distinct from monosyllables.

4aSC3. Prosodic perception and production of English- and Chinese-native speakers. Amanda Rodriguez and Chang Liu (Dept. of Commun. Sci. and Disorder, The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712)

Our previous studies have shown that English-native adult speakers demonstrated categorical perception in tonal identification of speech and nonspeech sounds as typically as Chinese-native adult speakers. The purpose of this study was to investigate whether prosodic perception of English speech sounds was different between English- and Chinese-native listeners. F0 contour was manipulated from falling to rising patterns for the target word embedded in a short sentence. Listener's task was to identify the prosody of each sentence, either question or statement. Preliminary results suggested that both groups of listeners showed typical categorical perception, while the two groups had significant difference in categorical boundary. The difference in categorical boundary for prosodic perception between the two groups of listeners was similar to our previous findings in tonal perception, likely due to the difference in the listener's language background. English sentences with statement and question will be recorded for the two groups of listeners. The relationship between prosodic perception and perception will be discussed. [Work supported by The University of Texas at Austin, Undergraduate Research Fellowship.]

4aSC4. Influence of perceptual training of syllable codas for English consonants on sentences. Teresa Lopez-Soto (Dept. English Lang., Univ. of Seville, c/ Palos de la Frontera, s/n 41004 Sevilla, Spain, teresals@us.es) and Diane Kewley-Port (Indiana Univ., Bloomington, IN 47405)

This study extends our previous report suggesting that moderate amounts of speech perception training were associated with improved speech production. This study also examined perception in sentences. A new group of 8 Spanish speakers (<10 year residence in US) trained on 13 English final consonants in syllables using SPATS software. On days 1 and 7 the group participated in perception and production tasks with the 13 codas (pre- and post-tests). Perception tests included coda identification in both syllable stimuli and in 18 sentences extracted from the IEEE corpus (90 keywords). Results showed the following. (1)With 5-h training of the 13 codas in syllable stimuli, perception improved significantly (>10%) in the syllable post-test, with smaller improvement in the sentence post-test (>7%). (2) Although average improvement was small for codas tested in sentences, several improved substantially (>20%), while a few decreased considerably after perception training. (3) Absolute values in both post-tests show similar performance, near 90%, although interesting differences for specific consonants were noted. The results showed that perception of sounds when measured in different linguistic contexts (syllables versus sentences) rendered different results. The study lays ground for investigating how cross-language perception of individual sounds is influenced by the phonetic context.

4aSC5. Cue weighting and variability in perception and production. Jiwon Hwang (Dept. of Linguist., Stony Brook Univ., Stony Brook, NY 11794-4376)

The Korean single liquid phoneme shows an allophonic variation: lateral [l] occurs in coda and tap [r] in onset. Intervocalically, they may appear contrastive as tap or geminate lateral [ll] ([iri] wolf vs [illi] reason), differing in duration and laterality. Kim [(2007)] demonstrated that Korean listeners identified a shortened geminate lateral as geminate /ll/ rather than tap, despite the fact that the duration of edited stimuli was matched for tap. The current study examines whether the weighting of laterality cues over duration cues for [ll] vs [r] is motivated by the native language acoustics. Korean speakers produced 24 Korean words [(C)V_V], containing [ll] or [r] in two

speech modes (in a carrier sentence vs in isolation). The duration of [ll] was significantly longer than tap, but it varied greatly depending on the speech mode while tap did not. The intensity of tap was significantly lower than [ll] generally, but it showed a greater variability. However, F3 at the offset of the preceding vowel was lower for tap than for [ll] regardless of the speech mode. The consistent use of F3 in production supports the perceptual cue weighting pattern where Korean listeners rely more on spectral cues than duration for intervocalic [ll]-[r] contrast.

4aSC6. The production and perception of English consonant sequences by Japanese-speaking learners of English. Mieko Sperbeck and Winifred Strange (Dept. of Linguist., City Univ. of New York, the Graduate Ctr., 365 Fifth Ave., New York, NY 10016)

This study reports the second part of a study investigating vowel insertion phenomena among Japanese speakers. A previous study [Sperbeck (2009)] that measured categorial discrimination demonstrated that some contrastive CCV versus CəCV sequences were hard for Japanese listeners (72% correct overall). The current study explored difficulties in production and how production correlated with perception. Nonsense words were constructed as the stimuli. They were of the form/CC(C)ani/ and /CəC(C)ani/, where CC(C) combinations were /sp, sk, pl, kl, bl, gl, spl, skl/. A delayed imitation task was used to assess production. Participants heard a native speaker's productions (e.g., Say blani now) twice, produced the target word in isolation (e.g., blani), and then produced it in the carrier sentence (e.g., I said blani now). The latter was scored in this study. Two phonetically trained native English speakers perceptually transcribed the productions. Results showed that the overall percent correct was 66% (SE = 3.17) among Japanese speakers. There was a significant correlation between perception and production performance (rho = + 0.715. $p<0.01$). However, the major error type was vowel deletion, rather than vowel epenthesis in producing the CəCV tokens. The relationship between perception and production among L2 learners will be discussed.

4aSC7. The discrimination, perception, and production of two German /r/ allophones by two groups of American English speakers. Dilara Tepeli (Dept. of Communicative Disord., Univ. of Wisconsin-Madison, 1975 Willow Dr., Madison, WI 53706)

The German /r/ sound is one of the most difficult sounds for American English (AE) speakers learning German as a foreign language. Part of this difficulty may be due to its rich phonetic variation. The standard German /r/ variant [R] and dialectal variant [R′] are achieved by varying the tongue constriction degree while keeping place of articulation constant [Schiller and Mooshammer (1995)]. The close articulatory proximity of these allophones provides an opportunity for testing the relationship between perception and production in L2 sound acquisition. The aim of this study is to investigate how well experienced AE speakers and naive AE speakers can discriminate and produce the difference between the uvular fricative [R] versus the uvular trill [R′]. Two groups of AE subjects who participated in an imitation study were prompted to produce single words beginning with either [R] or [R′]. Subjects also participated in a discrimination and categorization test. Preliminary results suggest that inexperienced AE can discriminate [R] versus [R′] well. They often perceive the sounds as /h/ and are more successful at producing [R′] than [R]. Experienced speakers also discriminate the two sounds well, perceive both sounds as the German /r/ and struggle more with producing [R′] than [R].

4aSC8. Auditory feedback shifts in one formant cause multi-formant responses. Shira Katseff (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720-2650, skatseff@berkeley.edu), John F. Houde (Univ. of California at San Francisco, San Francisco, CA 94143-0444), and Keith Johnson (Univ. of California, Berkeley, Berkeley, CA 94720-2650)

Talkers are known to compensate for experimentally-induced shifts in auditory feedback. In a typical experiment, talkers might hear their F1 feedback shifted (so that [ɛ] sounds like [æ], for example), and compensate by lowering F1 in their subsequent speech. Typically, compensation is assumed to directly oppose the action of the feedback shift and is measured in terms of the shifted parameter. In this study, we instead find that sensitivity to altered auditory feedback is multidimensional: subjects respond to altered F1 feedback by changing their F2 production and vice versa. In particular, subjects whose [i] is heard as [ɪ], a shift primarily in F1, compensated by producing a higher F2, while subjects whose central vowel [ʌ] was heard as [ɛ] or [o], a shift primarily in F2, compensated by producing a higher or lower F1. We argue that it is insufficient to consider auditory sensitivity in terms of a single formant and suggest that this method of altering auditory feedback is a practical tool for investigating the psychological reality of formants and their combinations.

4aSC9. Perception of vocal imitations and identification of the imitated sounds. Guillaume Lemaitre, Arnaud Dessein (IRCAM, 1 place Stravinsky, 75004 Paris, France), Karine Aura (Universit de Toulouse le Mirail, 31058 Toulouse, France), and Patrick Susini (IRCAM, 75004 Paris, France)

We report two studies investigating how vocal imitations enable the recognition of the imitated sounds. First, we asked couples of participants to listen to series of everyday sounds. One of the participants ("the speaker") had then to describe a selected sound to the other one (the "listener"), so that he could "guess" the selected sound. The results showed that, spontaneously, the speakers used, among other para-linguistic cues, large numbers of vocal imitations. Moreover, they suggested that the identification performances were increased when vocal imitations were used, compared to only verbal descriptions. Second, we sampled 28 sounds across an experimental taxonomy of kitchen sounds and required laypersons to vocally imitate these sounds. Another group of participants was then required to categorize these vocal imitations, according to what they thought was imitated. A hierarchical cluster analysis showed that, overall, the categories of vocal imitations fitted well with the categories of imitated sound sources. By using finer analysis techniques, we also showed that some imitations inconsistently clustered. On the other hand, the consistent clusters of imitations were perfectly predicted by a few acoustical descriptors. We therefore conclude that vocal imitations of sounds contain enough information for the recognition of the imitated sounds.

4aSC10. The perception and acoustic features of Korean ditropic sentences. Seung-yun Yang, Ji Sook Ahn, and Diana Van Lancker Sidtis (Dept. of Communicative Sci. & Disord., NYU, 665 Broadway, Ste. 900, New York, NY 10003)

Ditropic sentences are utterances that convey either a literal or an idiomatic meaning (e.g., It broke the ice). This study investigated listener's ability to discriminate between literal or idiomatic meanings and examined the acoustic features contributing to this distinction. Ten ditropically ambiguous Korean sentences were audio-recorded by four native speakers of Korean. Each utterance was produced twice with either a literal or idiomatic meaning. Fifteen native Korean subjects listened to a randomized presentation of these utterances singly and in pairs without other context and identified each as literal or idiomatic. Listeners successfully discriminated the intended idiomatic or literal meanings (singletons = 70.65%, pairs = 75 .67%). These results were consistent with those of Van Lancker and Canter [(1981)] for English ditropic sentences. Each utterance was acoustically analyzed in terms of means and variations in fundamental frequency, duration, and intensity. Analyses of variance revealed significantly longer durations and greater variation in syllable duration for literal than idiomatic sentences, whereas idiomatic sentences were characterized by significantly greater variation in intensity than literal sentences. Some prosodic cues for Korean differed from those found previously for English [Van Lancker *et al.* (1981)] and French [Abdelli-Baruh *et al.* (2007)]. These results further understanding of use of prosody in sentential linguistic contrasts.

4aSC11. Discrimination and identification of synthetic [da]-[ga] sounds by adults and children 4–6 years of age. Kelly Richardson (Dept. of Communicative Disord. and Sci., Univ. at Buffalo, 3435 Main St., Buffalo, NY 14214, kcr2@buffalo.edu) and Joan Sussman (Univ. at Buffalo, Buffalo, NY 14214)

Identification of sounds along the alveolar-to-velar contrast has been shown to be different for children born with clefts of the palate compared to other children [e.g., Whitehill *et al.* (2003)]. However, little information concerning discrimination abilities of children, in general, is known. The current study used a seven-step continuum of synthetic consonant-vowel syllables changing from "da" to "ga" by the 40 ms third formant frequency transitions [Sensimetrics Corporation (1995)]. Each listener heard 180 trials of a "change no-change" discrimination paradigm with two, three, four, and

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six-step comparisons to the endpoint, stimulus number 1 "da." Listeners were asked to verbally respond as to whether there was a "change" or "no-change" in the sounds presented. Results revealed that as the stimulus contrast increased, adults became better at discriminating the comparisons, whereas children's performance remained poor for all comparisons. Adults also showed less variability in their responses as the contrasts grew larger. In the identification task, young children displayed a much smaller velar category compared to the adult group. Furthermore, children were significantly poorer than adults at identifying the endpoint stimuli, as well as within-category exemplars. Results are compared to perception of other places of articulation by children and adults.

4aSC12. Misperceptions of foreign language phonology. Julia Yarmolinskaya, Colin Wilson, and Brenda Rapp (Dept. Cognit. Sci., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218)

For many years it has been assumed that when hearing familiar sounds in unfamiliar combinations, listeners will perceive the sounds accurately. In the recent years, this assumption has been challenged [Halle *et al.* (1998) and Berent *et al.* (2006)]. The present study investigates what listeners actually hear when presented with familiar consonants in unfamiliar combinations. A group of monolingual native English speakers was asked to transcribe Russian words containing only consonants attested in English but presented in a two-, three-, and four-consonant combinations (CC, CCC, and CCCC clusters) which are not legal in English. The results indicate that the accuracy of response and the nature of errors depend on the type of cluster. CC clusters (e.g., /pn/) were most often misperceived as containing a vowel between the two consonants, and perception accuracy was well-explained by phonological principles, such as sonority. On the other hand, CCC (e.g., /vzb/) and CCCC (e.g., /fstr/) cluster transcriptions contained relatively few vowel insertions, but many deletions and substitutions and their accuracy were better explained by acoustic factors, such as voicing of the consonants. These results suggest that speech perception is influenced by both phonological and acoustic factors.

4aSC13. Effects of listener experience with foreign accent on perception of accentedness and speaker age. Paul Rodrigues (Dept. of Linguist., Indiana Univ., 1021 E, Third St., Bloomington, IN 47405, prrodrig@indiana.edu) and Kyoko Nagao (Nemours A.I. duPont Hospital for Children, Wilmington, DE 19803)

The current study examined the effects of foreign accent and listener experience on the perception of a speaker's age and native language. Ten audio stimuli were prepared from the recording of five Arabic speakers and five English speakers (18–79 years old) from the Speech Accent Archive [Weinberger (2009)] for the perception experiment. Thirty native speakers of English participated in the perception experiment through Amazon's Mechanical Turk website, estimated the speaker age, rated the speaker's accentedness, and estimated the native language of the speaker. The listeners were divided into two groups based on their experience with foreign accented English (experienced and inexperienced groups). Higher correlation was found between perceived age (PA) and actual chronological age (CA) for the native (English) stimuli than for the Arabic-accented stimuli in both listener groups. The correlation between PA and CA was higher in the experienced listeners than in the inexperienced listeners. Accentedness rating suggests that the inexperienced listeners tend to rate both native and non-native speakers neutrally on the scale. The results suggest that experiences with any foreign accented speech facilitates identification of age from speech and help to form the ability to perceive differences in the degree of foreign accented speech.

4aSC14. Word segmentation of American English /s/ in semi-spontaneous speech. Dahee Kim (Dept. of Linguist., The Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210-1298, daheekim@ling.osu.edu), Christine Szostak, Colin Widmer, and Mark A. Pitt (The Ohio State Univ., Columbus, OH 43210-1287)

To comprehend spoken language, listeners need to find words from a continuous stream of speech sounds. Little work has explored whether there are reliable acoustic cues to word boundaries in conversational speech, which is highly reduced and under-articulated, potentially creating ambiguities at word boundaries. Segmentation may be even more difficult when the same segment repeats at a word boundary, ending the preceding word and beginning the following word (e.g., gas station). Segmentation in this environment was investigated by examining the production and perception of fricative /s/ in semi-spontaneous speech. Twenty talkers produced sentences containing ambiguous two-word sequences with /s/ between the two words. All sequences are interpretable in three ways (e.g., grow snails, gross snails, and gross nails) depending on how the frication is segmented. Acoustic analyses of the production data examined whether there are acoustic cues distinguishing the three versions of the ambiguous sequences. Listening experiments using the talkers' productions as stimuli evaluated the degree of ambiguity in the tokens and identified acoustic cues that listeners use to segment the two words. Results will be discussed in the context of theories of speech perception and word segmentation.

4aSC15. Perception of prosodic boundaries in spontaneous speech with and without silent pauses. Yoonsook Mo and Jennifer Cole (Dept. of Linguist., Beckman Inst., Univ. of Illinois, Urbana-Champaign, 1420 N. Mathews Ave., Urbana, IL 61801, ymo@illinois.edu)

In speech comprehension, listeners attend to variation in multiple acoustic parameters encoding prosodic structure. Given the multiplicity of acoustic cues, we ask whether prosody perception is dependent on any individual cue or whether acoustic redundancy encoding prosody supports robust prosody perception in the absence of an individual cue. The present paper reports on a study of boundary perception in spontaneous speech with and without silent pause as a boundary cue. Prior studies show that in read speech, silent pause is important for boundary perception, while in spontaneous speech, listeners can detect boundaries without pauses. Our study tests the role of pause in boundary perception with two versions of 36 short speech excerpts from the Buckeye Corpus: one with pauses intact and another with all pauses truncated to 20 ms. In real-time transcription tasks based only on auditory impression, boundary locations were marked by 74 subjects for the intact stimuli and by an additional 15 subjects for truncated excerpts. Inter-transcriber agreement was comparable across the intact and truncated conditions. Paired-sample t-tests show significantly higher rates of boundary perception for intact stimuli indicating that silent pause is an important but not necessary cue to boundary perception and cue redundancy allows for robust perception.

4aSC16. Dichotic digit listening in Mandarin and English by Mandarin-speaking adults. Shu-Yu Liu (School of Speech Lang. Pathol. and Audiol., Chun Shan Medical Univ., Chien-Kuo North Rd., Taichung 402, Taiwan, audio@csmu.edu.tw) and Jia-Shiou Liao (Chun Shan Medical Univ., Taichung 402, Taiwan)

This study examines Mandarin speakers' performance on Mandarin and English dichotic digit recognition tests (DDTs). The 60 right-handed subjects, whose primary language was Taiwan Mandarin, had started English as their second language no later than in seventh grade and continued with English through their freshman year in Taiwan. All the subjects were tested on their ability to pronounce and use English digits before participating in the experiments. Subjects took Mandarin one- and two-pair DDTs, and English one- and two-pair DDTs, in free-recall paradigms. In each DDT, subjects were asked to report the digits orally in any order no matter in what order they had heard the numbers in each ear. The scores were calculated from the number of digits the participants responded correctly to in each ear on one- or two-pair tests; they were then statistically analyzed. The English one- and two-pair recognition tests showed a significant right-ear advantage (REA) but not the Mandarin ones. The scores on the Mandarin one- or two-pair DDTs are higher than those on the English ones, suggesting REA or the influence of the non-native language on the subjects' performance.

4aSC17. Lexical recognition memory across dialects. Cynthia G. Clopper (Dept. of Linguist., Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu) and Terrin N. Tamati (Indiana Univ., Bloomington, IN 47405)

Implicit recognition memory for spoken words is more accurate when words are repeated by the same or a similar talker than when they are repeated by a different talker. The current study explored implicit recognition memory for words repeated by the same talker, by a different talker from the same dialect, and by a different talker from a different dialect in a word recognition task in noise. Repetitions produced by the same talker facilitated word recognition performance. However, for target words originally produced by talkers from the Northern dialect of American English, repetitions produced by a talker from the Midland dialect of American English inhibited word recognition performance. No repetition effect was observed for repetitions produced by a different talker from the same dialect or for words repeated by Northern talkers that were originally produced by Midland talkers. These results suggest an asymmetry in how indexical information is stored and activated in lexical processing. The same talker repetition effect was observed for both dialects, but one variety (the Northern dialect of American English) inhibited the repetition effect across dialects and the other variety (the Midland dialect of American English) did not.

4aSC18. Perceptual similarity of unfamiliar regional dialects. Terrin N. Tamati (Dept. of Linguist., Indiana Univ., Bloomington, IN 47405, ttamati@indiana.edu)

Linguistic experience has been shown to influence the perception of regional dialect variation. Recent studies have found that the amount and type of experience with regional dialect variation affect performance in identifying or categorizing regional dialects. Experience also shapes the perceived similarity among regional varieties. To examine the effect of familiarity on the perceived similarity of regional dialects, a paired comparison perceptual similarity rating task was carried out with a group of unfamiliar regional dialects. Native speakers of American English made explicit judgments about the similarity of unfamiliar talkers from the United Kingdom and Ireland based on the regional dialect. Results show that listeners judged the regional dialects of pairs of talkers from the same dialect region as more similar than those of pairs of talkers from different dialect regions. A multidimensional scaling analysis revealed two dimensions of perceptual dialect similarity, both reflecting the geographic location of the cities of origins of the talkers (north versus south and east versus west). Thus, despite being unfamiliar with the regional dialects in the study, listeners were able to use dialect-specific differences in the acoustic signal to make judgments on the perceptual similarity of talkers based on regional dialect.

4aSC19. Identification of the place of articulation of trilingual postvocalic nasals and stops by native speakers of American English, Korean and Japanese. Takeshi Nozawa (College of Economics, Lang. Edu. Cntr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu, Shiga 525-8577, Japan) and Sang Yee Cheon (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

Native speakers of American English and Korean produced postvocalic nasals and stops in /CVC/ frames in which the syllable final segment was a consonant with airstream blocking, and native speakers of Japanese produced more nasal /N/ and obstruent /Q/ in /CVNCV/ and /CVQCV/ frames. In Japanese stimuli, the consonant after the nasal or the obstruent was always a stop. Their utterances were recorded and edited to be used as stimuli for the experiment. The release burst of the English stimuli and the second syllable of Japanese stimuli were deleted. Native speakers of these three languages were recruited as listeners. They identified the place of articulation of the syllable-final nasals and stops of the three languages. As predicted, Japanese listeners performed most poorly because there are no phonemic contrasts between postvocalic nasals or stops in Japanese. Korean listeners outperformed the other two groups of listeners in identifying the place of articulation. Postvocalic stops in Korean are not released, so the Korean listeners may not depend on the release burst to identify the place of articulation of a syllable final stop. However, they made more voicing errors than American listeners probably because voiced stops in Korean cannot occur in a postvocalic position.

4aSC20. Why [spa] not [psa]? On the perceptual salience of initial /s/-stop and stop-/s/ sequences. Asimina Syrika (Dept. of Communicative Disord., Univ. of Wisconsin-Madison, Goodnight Hall, 1975 Willow Dr., Madison, WI 53706, syrika@wisc.edu), Jan Edwards, Marios Fourakis, Eun Jong Kong (Univ. of Wisconsin-Madison, Madison, WI 53706), Benjamin Munson (Univ. of Minnesota, Minneapolis, MN 55455), and Mary E. Beckman (Ohio State Univ., Columbus, OH 43210)

Initial /s/-stop clusters occur frequently in the world's languages, but initial stop-/s/ clusters are relatively infrequent. Furthermore, there appear to be no languages that contain initial stop-/s/ clusters, but not /s/-stop clusters, while the reverse is not true [Morelli, (1999) and (2003)]. This study aims at uncovering a perceptual explanation for these patterns by examining the salience of initial /s/-stop and stop-/s/ clusters in Greek, where both sequences are common. Twenty naïve Greek adult listeners identified syllables beginning with /sp/, /st/, /sk/, /ps/, /ts/, or /ks/, in two vowel contexts, /a/ and /i/, in real words spoken by ten Greek adult native speakers. The syllables were mixed with parts of Greek multitalker babble using SNRs of -6, 0, and $+6$ dB and presented to listeners for identification. Results showed significantly poorer identification for the /ps/ and /ks/ clusters than the /ts/ and /s/-stop clusters, particularly in the -6 and 0 SNRS. There was also a significant interaction with vowel, such that /sk/ and /ts/ were identified more accurately before /a/, whereas /ks/ was identified more accurately before /i/. The implications of these findings for phonological acquisition and speech perception are considered. [Work supported by NIDCD 02932 and NSF Grant 0729140.]

4aSC21. The perceptual acquisition of Korean fricatives by first language Mandarin listeners. Jeffrey J. Holliday (Dept. of Linguist., Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, jeffh@ling.ohio-state.edu)

Despite numerous studies, it remains unclear how naive "foreign language" listeners become proficient L2 listeners, particularly in regard to difficult L2 contrasts. In an earlier study in which nine English-speaking L2 learners of Korean of varying proficiency and history of exposure to Korean identified Korean tense /s*/ versus non-tense /s/, listeners showed varying perceptual strategies. The results suggested that learners gradually learn to perceive differences in L2 contrasts by re-weighting useful cues and learning to ignore the "inefficient" cues that are initially relied on when the members of the L2 contrast are assimilated to L1 categories. This paper will report on the results of the same task (identification of CV sequences excised from real words), testing 30 L1 Mandarin speakers who have been in an intensive Korean language program in Seoul for about 2 months. The results of the present study will show whether there are as many inter-listener differences when the level and type of L2 exposure are much more controlled. In addition, because the acoustic cues relevant to the Mandarin sibilant fricative distinctions differ from those used in English the results will show to what extent the choice of perceptual strategy depends on acoustic properties of L1 contrasts.

4aSC22. Cross-linguistic perception of velar and alveolar obstruents: A perceptual and psychoacoustic study. Timothy Arbisi-Kelm (Dept. of Commun. Sci. and Disord., Augustana College, 639 38th St., Rock Island, IL 61201, timothyarbisi-kelm@augustana.edu), Jan Edwards (Univ. of Wisconsin-Madison, Madison, WI 53706), and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN 55455)

It is well-known that velar stop consonants coarticulate more with the following vowel than stops at other places of articulation. The fine phonetic detail of this coarticulation is highly language-specific. For example, /k/ in Greek is more front before front vowels and more back before back vowels relative to /k/ in English [Arbisi-Kelm *et al.* (2008)]. The purpose of this study was to investigate how these cross-linguistic differences in production influence perception of place of articulation for lingual stops. The stimuli were word-initial consonant-vowel (CV) sequences excised from words produced by 2- to 5-year-old children and adults. The listeners were 20 adult native English speakers (tested in Minneapolis, USA) and Greek speakers (tested in Thessaloniki, Greece) who listened to these sequences combined across ages and languages in a visual analog scaling task [Urberg-Carlson *et al.* (2008)]. Listeners rated how alveolar or velar each sequence was by clicking on a double-headed arrow anchored with language-specific orthographic representations of the target consonants. Results showed that the two groups of adults perceived the sounds differently, as would be expected. We will report on the relationship between listeners' perception and psychoacoustic properties of the stop bursts. [Work supported by NIDCD 02932 and NSF BCS072914 and BCS0729277.]

4aSC23. Individual differences in use of English fricative perceptual cues. Elizabeth Casserly (Dept. of Linguist., Indiana Univ., Memorial Hall 322, Bloomington, IN 47405, casserly@indiana.edu)

This study examines individual variation as a potential explanation for contradictory and inconsistent reports of English speakers' use of acoustic cues in identification of the voiceless sibilants [s] and [ʃ]. While there is widespread agreement that the spectral shape of turbulent noise is key for

identification of these two categories, some studies find that formant transitions to and from the noise also influence identification [Whalen, J. Acoust. Soc. Am. **69**, 275–282 (1981)], while others do not [Harris, Lang. Speech **1**, 1–7 (1958)]. Similarly, the majority of studies investigating the effect of vowel context on fricative perception show that the presence of round vowels biases listeners toward perception of [s] [Kunisaki and Fujisaki, Ann. Bull. RILP **11**, 85–91 (1977)], but others show precisely the opposite effect, where round vowels favor [ʃ] responses [Nittrouer and Studdert-Kennedy, J. Speech Hear. Res. **30**, 319–329 (1987)]. In this study, 30 native English speakers participated in a labeling experiment that fully crossed all three factors—spectral noise shape, formant transitions, and vocalic context—for each subject. Every pattern of cue use found in the literature is also found in one or more of the individuals, which may explain why averaged results vary so widely across reports.

4aSC24. Learning and generalization of novel contrasts across speakers. Kyuwon Moon (kyuwon@stanford.edu) and Meghan Sumner (Dept. of Linguist., Stanford Univ., Stanford, CA 94305-2150)

This paper examines how listeners use a learned contrast when encountering novel speakers. Do speakers reset to their native perceptual biases or apply a learned contrast to new speakers? In two experiments, participants took a minimal pair decision pre-test (MPD), a training session in which a native Korean speaker contrasts stop release (e.g., [bEt] = BET, [bEt'] = BED) without any V/C duration differences, a post-test (identical to pre-test), and a generalization-test [identical to pre-test, but with a speaker of a different L1 (Arabic)]. In Experiment 1, the only difference between the post-test and gen-test was the L1 of the speakers. We found that a learned phonetic contrast generalizes across speakers of different L1s with equally strong effects for post-test and generalization-test independent of order of presentation. In Experiment 2, the post-test was identical to that in Experiment 1, but the learned contrast was paired with vowel durations consistent with native English. Listeners were slower and less accurate in both the generalization-test and post-test when the generalization-test was presented before the post-test. The resulting asymmetry between the two experiments suggests that listeners use learned contrasts, but quickly reset to native patterns when native cues are present.

4aSC25. Perceptual learning of talker-idiosyncratic phonetic cues. Alexandra Jesse (Max-Planck-Institut für Psycholinguistik, Postbus 310, 6500 AH Nijmegen, The Netherlands, alexandra.jesse@mpi.nl) and Rochelle S. Newman (Univ. of Maryland, College Park, MD 20742)

A number of recent studies have explored "perceptual learning," in which listeners use lexical knowledge to learn about a talker's idiosyncratic phoneme pronunciations and adjust their perception of other tokens from that talker accordingly. In a typical perceptual learning study, listeners might hear an item that is ambiguous between "crocotile" and "crocodile" during exposure. Since only crocodile is a word, listeners would learn (following several examples) that this talker has long VOTs, and subsequently at test show a shift in their categorization of a /d/-/t/ VOT continuum by the same talker. The present study explored perceptual learning through cues rather than through lexical knowledge. We used a phonetic contrast (s-th) in which there are both primary (spectral) and secondary (amplitude/duration) cues to phonetic identity. Listeners heard tokens of minimal s-th word pairs in which either the primary or secondary cue was ambiguous, but the alternative cue was unambiguous and thus disambiguated the phonetic identity of the word. We tested whether listeners use the unambiguous cue to learn about the speaker's production of the ambiguous cue (even though doing so was unnecessary for lexical identification) which would then influence later perceptual identification of a series based only on that cue.

4aSC26. Talker-specific accent: Can speech alignment reveal idiolectic influences during the perception of accented speech? Rachel M. Miller, Kauyumari Sanchez, Lawrence D. Rosenblum, James W. Dias, and Neal Dykmans (Dept. of Psych., Univ. of California, 900 Univ. Ave., Riverside, CA 92521, rmiller.ucr.grad@gmail.com)

Listeners use talker-specific (idiolectic) information to help them perceive and remember speech [e.g., Goldinger, J. Exp. Psychol. Learn. **22**, 1166–1183 (1998)]. However, recent research has shown that idiolectic information is not as helpful when listeners hear accented speech [e.g., Sidaras *et al.*, J. Acoust. Soc. Am. **125**, 5 (2009)]. It could be that listeners fail to encode idiolectic information when perceiving accented speech. To examine whether idiolectic is still encoded, experiments tested if subjects would display speech alignment to specific accented models. Speech alignment is the tendency to imitate another talker and can occur when shadowing heard speech [e.g., Goldinger, Psychol. Rev. **105**, 251 (1998)]. Native English subjects were asked to shadow a Chinese- or Spanish-accented model producing English words. Raters then judged whether the shadowed tokens were more similar in pronunciation to those of a shadowed model or of a different models with the same accent. In a second experiment, raters judged whether shadowed tokens were more similar in accent to those of (unshadowed) models with the same or a different accent. Preliminary results reveal that subjects align to the shadowed model, suggesting that idiolect is still encoded. Subjects also show moderate alignment to accent.

4aSC27. Initial acoustic-phonetic processing of competing verbal stimuli examined using dichotic verbal transformations. Peter W. Lenz, James A. Bashford, Jr., and Richard M. Warren (Dept. of Psych., Univ. of Wisconsin-Milwaukee, P.O. Box 413, Garland 224, Milwaukee, WI 53201, plenz @uwm.edu)

Initial studies with dichotic verbal transformations (VTs) of repeating words employed a cross-ear asynchrony of half the word's duration and listeners called out the independent perceptual changes at each ear as they occurred. In contrast to monaural and diotic VTs, the dichotic "immediate response" procedure is extremely difficult and tiring, requiring concurrent monitoring while remembering the word previously heard on each side. The present study employs a much less demanding task—a cued report procedure: the listener calls out what is heard on each side when prompted by a periodic light flash. When the asynchronous stimuli were statements of the same word, the transition rates on each side were the same as when presented monaurally, in contrast with the decreased rates reported with the earlier procedure. With different words on each side, the transition rates were diminished by an amount depending on the extent of their phonetic differences—rates were not influenced by either semantic relations or differences in the neighborhood density of the competitors. It is suggested that dichotic verbal transformations provide access to aspects of the acoustic-phonetic front end of speech analysis that may be obscured by subsequent levels of processing. [Work supported by NIH.]

4aSC28. PRESTO: Perceptually robust English sentence test: Open-set—Design, philosophy, and preliminary findings. Hanyong Park (Speech Res. Lab., Univ. of Indiana, 1101 E. 10th St., Bloomington, IN 40405, hanypark@indiana.edu), Robert Felty (Univ. of Colorado, Boulder, CO), Kelly Lormore (Indiana Univ. Cochlear Implant Program, Indianapolis, IN), and David B. Pisoni (Univ. of Indiana, Bloomington, IN)

Traditional clinical word recognition tests (e.g., SPIN and HINT) do not reflect variability in real world environments. These tests are often constructed with predictable short sentences spoken by a small number of talkers without dialect variation; thus, individual differences are difficult to uncover. To compensate for these shortcomings, PRESTO uses materials taken from the TIMIT database in order to incorporate variation in talkers, dialects, and number of words in a sentence. Familiarity and lexical frequency, syntactic structure, and semantic content are also considered. To provide normative data, normal-hearing young listeners performed an open-set identification task with the PRESTO materials in open field with six-talker babble mixed at different signal-to-noise ratios and also with no additional background noise. The HINT sentences were also used with the same listeners to assess the validity of PRESTO. Results indicate that listeners perform near ceiling for both the PRESTO and HINT materials in ideal listening conditions. However, under degraded listening conditions, the PRESTO shows much greater variability. Preliminary results from a clinical population also indicate that the more challenging PRESTO lists reveal individual differences among patients which are not apparent from scores on the HINT. [NIDCD T32-DC00012.]

Session 4aSP

Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics: Maximum Entropy and Bayesian Signal Processing I

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Chair's Introduction—7:35

Invited Papers

7:40

4aSP1. Bayesian approach to model-based signal processing: An overview. James V Candy (Lawrence Livermore Natl. Lab., P.O. Box 808, L-151, Livermore, CA 94551)

Although available for a long time with the advent of high-speed/high-throughput computing, the development of Bayesian processing techniques has evolved recently in acoustical signal processing. Bayesian signal processing is concerned with the estimation of the underlying probability distribution of a random signal in order to perform statistical inferences such as the conditional mean estimation. Knowledge of this distribution provides all of the essential information available required for problem solution. The usual limitations of nonlinear approximations and non-gaussian processes prevalent in classical algorithms (e.g., Kalman filters) are no longer a restriction to perform Bayesian inference. This approach enables the next generation of processors called particle filters that are sequential Monte Carlo methods providing an estimate of the underlying discrete probability distribution. In this overview, Bayesian signal processing is presented from a probabilistic perspective starting with Bayes rule and evolving to the development of a bootstrap particle filter perhaps one of the most common and simplest constructs available. The relationship of Bayesian processing to the concept of maximum entropy is discussed. Maximum entropy and its applicability in Bayesian processing is also mentioned briefly.

8:00

4aSP2. Defining uncertainty with maximum entropy method. David P. Knobles, Jason Sagers, and Robert Koch (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

The maximum entropy (ME) method was strongly defended and advocated by E. T. Jaynes as a means to define uncertainty. Here, the ME method is applied to the estimation of ocean waveguide parameter probability distributions from measured acoustic data. An ME analysis produces a canonical distribution, well known from equilibrium statistical mechanics, which is the distribution that maximizes the entropy subject to constraints that reflect selected features of the measured data and a model. The ME method gives the most conservative distribution based only on the measured data and observed features. A Bayesian approach also has the goal of defining uncertainty, but starts from the specification of the likelihood function and the model priors. Data noise is naturally handled in the specification of the likelihood function. The discussion introduces simple examples, showing basic relationships between the constraints in ME and the maximum likelihood estimation. In special cases the form of the likelihood function used in Bayesian conditionalization can be derived from the ME approach. The form of the cost function is an important consideration in comparing ME and Bayesian methods of inferences. In general ME and Bayesian inferences lead to different results. [Work supported by ONR Code 321 OA.]

8:20

4aSP3. Automatic signal detection in noise using entropy. Christine Erbe (JASCO Appl. Sci., Brisbane Technol. Park, 1 Clunies Ross Ct, Eight Mile Plains, QLD 4113, Australia, christine@jasco.com)

Automatic detection of signals in noise is a common problem in many areas of acoustics. In the field of passive acoustic monitoring of marine mammals, the signals to be detected are vocalizations. The noise originates from natural (wind, waves, and rain) and man-made sources (e.g., shipping, construction, and seismic surveys). Signal characteristics vary broadly: frequency ranges from a few Hz to 200 kHz, duration from milliseconds to seconds to hours. Noise characteristics vary by similar orders of magnitude. While specific automatic detectors have been designed to successfully find specific calls in specific environments, the challenge is to find a large variety of calls in a large variety of noise. An exploitable difference between calls and noise is that most noise is a result of stochastic processes (wind, waves, rain, cavitating propellers + seismics generate gas bubbles underwater of varying size + resonance frequency), while many animal signals are a result of deterministic processes (vibrating strings & cavities of predetermined/fixed size). Shannon entropy was computed for power spectrum density functions of underwater recordings. Noise yielded high signal low entropy. Results are presented from passive acoustic surveys of marine mammals. The benefits and limitations of entropy applied to automatic signal detection are discussed.

8:40

4aSP4. The likelihood ratio and Bayesian signal processing. R. Lee Culver and Colin W. Jemmott (Appl. Res. Lab and Grad. Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Over the past several years, we have been developing an architecture for classifying source depth associated with passive sonar signals. The classifiers utilize the statistics of a signal parameter which are estimated using knowledge of the environment and an acoustic propagation program. We have applied the likelihood ratio (LR) test to classify source depth using signal statistics from the SWellEx-96 and 1996 Strait of Gibraltar sea tests. More recently Bissinger developed a Hellinger distance classifier, and Jemmott is developing a histogram (discrete Bayesian) filter for this purpose. In this talk, we examine the relationship between the LR test and a processor that makes use of Bayes rule. We consider some of the fundamentals. It is useful to understand the underlying assumptions of the LR, the likelihood function, and how they are related to a Bayesian processor which makes use of prior information and computes a posterior probability distribution function. Under what conditions do the two processors produce the same answer? When would the Bayesian processor be a better choice? We compare the processors and apply them to the SWellEx-96 data. [Work supported by the Office of Naval Research Undersea Signal Processing.]

9:00

4aSP5. Bayesian bounds on passive sonar accuracy from binary performance metrics. John R. Buck (Dept. ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., N. Dartmouth, MA 02747, johnbuck@ieee.org)

A passive sonar algorithm measures the pressure field at a sensor array then estimates the sound source location from these observations and an acoustic propagation model. Passive sonar performance is usually characterized by the mean squared error (MSE) between the estimated and true source locations. Consequently, passive sonar performance bounds typically provide lower bounds on the achievable MSE for a given array and environment. Such MSE bounds can be misleading in environments with strong sidelobes, as the optimal estimator may choose an unlikely location to balance the error among several highly likely locations. An alternative algorithm would be to partition the search space, then use the array observations to choose which block contains the source, but not its exact location in the block. The resulting binary performance metric is now the probability of choosing the incorrect block or error probability (P_e). Information theory allows us to formulate Bayesian bounds on the minimum achievable P_e for a given array, propagation environment, and search space partition. These bounds quantify the trade-off among SNR, P_e, and location estimation accuracy for passive sonar. [Work supported by ONR Code 321US.]

9:20

4aSP6. Using Bayesian inference for acoustic array design. Paul M. Goggans and Chung-Yong Chan (Dept. of Elec. Eng., Univ. of Mississippi, Anderson Hall, Rm. 302B, University, MS 38677)

Because inference and design are both generalized inverse problems, the tool and methods developed for Bayesian parameter estimation and model comparison can be adapted and used for the solution of design problems. As an example, this paper presents the use of the Bayesian inference framework for the automated design of linear transducer arrays [P. M. Goggans and C.-Y. Chan, "Antenna array design as inference," AIP Conf. Proc. **1073**, 294–300 (2008)]. Commonly, automated array design is cast as an optimization problem and solved using numerical optimization techniques. Here, the design of linear arrays is cast as an inference problem and solved using numerical Bayesian inference techniques. Compared to optimization-based methods, the inference-based method presented here has the advantage of being able to automatically determine the number of array elements required to satisfy design requirements and specifications. In addition, array design cast as inference can incorporate, as prior information, design requirements such as a minimum spacing between two adjacent elements, a maximum aperture width, and a necessary operating frequency bandwidth. Sample results are presented to demonstrate the application of the Bayesian inference framework in the automated design of linear arrays.

9:40

4aSP7. Recursive Bayesian state estimation for passive sonar localization. Colin W. Jemmott and R. Lee Culver (Penn State Appl. Res. Lab and Grad. Program in Acoust., P.O. Box 30, State College, PA 16804, cwj112@psu.edu)

A model-based recursive Bayesian signal processing framework is shown to localize a moving source emitting a low-frequency tonal signal in a shallow water environment. Source motion maps spatial variation in transmission loss into amplitude modulation of the signal received on a passive horizontal array. Acoustic propagation modeling predicts this variability, which is used to estimate source range, depth, range rate, and acoustic level. Uncertainty in transmission loss resulting from uncertainty in environmental parameters is predicted using Monte Carlo modal propagation modeling. Monte Carlo marginalization over environmental uncertainty provides robustness against data-model mismatch. The maximum entropy method is used to construct a probability density function (pdf) of transmission loss at each range depth location based on the Monte Carlo results. The resulting pdfs belong to the exponential family and result in an implementable recursive Bayesian processor. The physics of acoustic modeling determine the form of the processor through the transmission loss pdfs and are an intimate part of the localization technique. This processor is distinct from Bayesian matched field processing in that it neither relies on a vertical array nor computes modal amplitudes from received data. Results using SWellEx-96 will be shown. [Work supported by ONR Undersea Signal Processing.]

10:00—10:15 Break

10:15

4aSP8. Bayesian geoacoustic inversion. Stan E. Dosso and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

This paper describes a general Bayesian approach to estimating seabed geoacoustic parameters from ocean acoustic data, which is also applicable to other inverse problems. Within a Bayesian formulation, the complete solution is given by the posterior probability density (PPD), which includes both data and prior information. Properties of the PPD, such as optimal parameter estimates, variances /covariances, correlations, and marginal probability distributions, are computed numerically for nonlinear problems using Markov-chain Monte Carlo methods. However, in many practical cases, both an appropriate model parametrization and the data error distribution are unknown and must be estimated as part of the inversion. These problems are linked, since the resolving power of the data is affected by the data uncertainties. Model selection is carried out by evaluating Bayesian evidence (parametrization likelihood given the data), or a point estimate thereof such as the Bayesian information criterion, which provides the simplest parametrization consistent with the data. The error covariance matrix (including off-diagonal terms, as needed) is estimated from residual analysis under the assumption of a simple, physically reasonable distribution form, such as a Gaussian or Laplace distribution. The validity of the above assumptions and estimates is examined *a posteriori* using both qualitative and quantitative statistical tests.

10:35

4aSP9. Particle filtering for sequential multipath arrival time and amplitude estimation. Rashi Jain and Zoi-Heleni Michalopoulou (Dept. of Math. Sci., New Jersey Inst. of Technol., Newark, NJ 07102)

Accurately estimating arrival times from acoustic time series in the ocean is essential for successful source and array element localization and estimation of the geometry of the sound propagation environment and environmental parameters such as sound speed in the water column and sediments. We have developed a sequential Monte Carlo method that characterizes multipath arrivals as moving targets, tracking them at spatially separated receiving phones. We focus on switching models that are suitable for unknown and varying numbers of arrivals at different phones. We also present approaches that efficiently and effectively extract amplitude information from received time series; such information can be then employed for sediment characterization. Our methods are applied to Haro Strait Primer and Shallow Water 06 data; their performance is evaluated through comparisons to conventional approaches. [Work supported by ONR.]

Contributed Papers

10:55

4aSP10. Sequential Bayesian strategies in geoacoustic inverse problems. 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 considers sequential Bayesian strategies for geoacoustic inverse problems which are difficult to solve simultaneously due to computational constraints. Bayesian inference provides a powerful approach to learning problems such as this since sequential inversions of multiple data sets [with the posterior probability density (PPD) of one inversion applied as prior information in the subsequent inversion] are equivalent to simultaneous inversion of all data. However, passing PPDs forward as priors has its own challenges when the PPD is sampled numerically for nonlinear inverse problems, particularly when the model parameter space is of high dimensionality and the data information content is high. In such cases, approximations are required to efficiently carry PPD information forward to subsequent inversions. The approach developed here represents numerically sampled PPDs in terms of discretized marginal probability distributions for principal components of the parameters, which minimizes the loss of information in representing inter-parameter correlations. The sequential Bayesian approach is applied to seabed reflectivity inversion with multiple data sets representing travel-time data and frequency-domain reflection coefficient data for a series of increasing penetration depths. Data information content is quantified by accounting for potential error biases as well as data error covariances. [Work supported by the Office of Naval Research.]

11:10

4aSP11. Three-dimensional source tracking in an uncertain environment via Bayesian marginalization. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Box 115, 3191 Horten, Norway) and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

This paper develops a non-linear Bayesian marginalization approach for three-dimensional source tracking in shallow water with uncertain environmental properties, with application to horizontal line array (HLA) data. The algorithm integrates the posterior probability density via a combination of Metropolis–Hastings sampling over environmental and bearing model parameters and Gibbs sampling over source range and depth, with a priori track constraints on source velocity. Two-dimensional marginal distributions for source range/depth and range/bearing are derived. The Viterbi algorithm is applied to obtain the most probable track, with uncertainties estimated from the marginal distributions. The algorithm is applied to simulated data in continental shelf environment and to towed-source and ship-noise data recorded on a HLA deployed on the seafloor in an experiment conducted in the Barents Sea.

11:25

4aSP12. Computation of normalizing constants in geoacoustic Bayesian inference. Jan Dettmer and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

This paper considers approaches to computing normalizing constants (Z) in Bayesian inference problems. Bayes' theorem combines the likelihood function, model prior, and Z to form the posterior probability density (PPD). Z (also known as evidence) is difficult to compute for general problems and a common approach is to avoid its computation entirely by calculating an unnormalized estimate of the PPD which is sufficient for moment estimates. However, estimating the normalized PPD, including Z, allows for moment estimates as well as quantifying the likelihood of the model parametrization. This is commonly referred to as model selection and poses a natural way to quantifying the most appropriate model parametrization for a given data set (Bayesian razor). Several approaches for computing Z have been developed in the statistics community, some of which are applied here to the geoacoustic inference problem. Annealed importance sampling follows an annealing approach and computes weighted averages along cooling trajectories. Nested sampling uses a likelihood constraint to move from the prior mass to the posterior. Both methods also give parameter estimates which are compared to Metropolis–Hastings results. [Work supported by the Office of Naval Research.]

Session 4aUW

Underwater Acoustics: Propagation and Scattering in Heteorogeneous Waveguides

Jon Collis, Chair
Colorado School of Mines, Dept. of Mathematical and Computer Science, 1500 Illinois St., Golden, CO 80401

Contributed Papers

8:00

4aUW1. Improving the parabolic equation solution for problems involving poro-elastic media. Adam M. Metzler, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180), Michael D. Collins, Ralph N. Baer (Naval Res. Lab., Washington, DC 20375), and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Parabolic equation solutions for elastic media have undergone several modifications recently that increase their capabilities and accuracy. These include formulation in different dependent variables, approaches for handling range dependence such as coordinate rotations and single scattering, and treatment of media anisotropy. These advances are being extended to problems with heterogeneous and range-dependent poro-elastic media, which provide useful models of some shallow-water sediments. Other parabolic equation solutions for poro-elastic media [Collins *et al.*, J. Acoust. Soc. Am. **98**, 1645–1656 (1995)] are prior to recent progress. Vertical dependence is treated by applying heterogeneous depth operators from the equation of motion. Horizontal dependence is treated by incorporating single-scattering approaches. [Work supported by the ONR.]

8:15

4aUW2. Seismo-acoustic propagation near low-shear speed poroelastic ocean sediments using a hybrid parabolic equation solution. Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Accurate and efficient parabolic equation solutions exist for complex propagation environments featuring elastic and porous elastic sediment types. An area of concern has been low-shear wave speed sediments that become singular as their shear modulus tends toward zero. A historic approach for treating sediments of this type has been to assume that it is a fluid, and effects due to elasticity are negligible. This approach is limited in accuracy unless shear is accounted for. In this presentation, the ocean bottom sediment interface layer is treated as a porous elastic layer in which poroelastic momentum equations are solved and combined with an existing elastic parabolic equation implementation. Appropriate boundary conditions are enforced at the fluid-poroelastic and poroelastic-elastic interfaces. The new solution is tested on problems with a low-shear ocean bottom interface layer.

8:30

4aUW3. Improving the parabolic equation solution for problems involving sloping fluid-solid interfaces. Michael D. Collins (Naval Res. Lab., Washington, DC 20375, collins@noddy.nrl.navy.mil) and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, New York 12180)

Several approaches are being investigated for improving the elastic parabolic equation for problems involving sloping fluid-solid interfaces. Approaches based on single scattering and energy conservation provide accurate solutions for problems involving sloping fluid-fluid and solid-solid interfaces, but the mixed-media problem has proven to be more challenging. The energy-conservation approach has been applied previously by deriving a linear equivalent to the nonlinear expression for energy flux. One of the approaches that are currently being investigated is based on going back to the nonlinear expression. Although it would not be practical to solve the full nonlinear scattering problem, promising results have been obtained for the fluid-fluid case with this approach by correcting the amplitude at only one grid point near the interface. With this approach, the nonlinear problem reduces to the evaluation of a quadratic function. Another approach that is being investigated is based on an alternative formulation that involves the vertical displacement and a quantity that is proportional to the normal stress on a horizontal interface. In these variables, the interface conditions across a horizontal interface are first order, and this may facilitate the extension of the single-scattering solution to the mixed-media problem. [Work supported by the Office of Naval Research.]

8:45

4aUW4. Dependence of the structure of the shallow convergence zone on deep ocean bathymetry. Stephen D. Lynch (slynch@mpl.ucsd.edu), Gerald L. D'Spain (Marine Physical Lab., Scripps Inst. Oc., San Diego, CA), Kevin Heaney (OASIS, Lexington, VA), Arthur B. Baggeroer (MIT, Cambridge, MA), Peter Worcester (Scripps Inst. Oc., La Jolla, CA), and James Mercer (APL/UW, Seattle, WA)

During an experiment in the northern Philippine Sea in 2009, low-frequency tones were transmitted from a shallow (15- and 60-m) source deployed from R/V Melville keeping station to a shallow (250-m) horizontal receiver array towed by R/V Kilo Moana approximately one convergence zone (CZ) away. Recordings were made during events in which the receiver ship maintained constant range in the convergence zone and during events in which the receiver ship transited radially through the CZ. The shallow CZ exhibits strong dependence on the bathymetry mid-way between the source and receiver array. In fact, the variability of the structure of the first CZ in this environment is significantly more strongly affected by the heterogeneous character of the bottom than water column fluctuations. Numerical modeling with a parabolic equation code is used to support the conclusions from the data analysis.

9:00

4aUW5. Range dependence in the level set method for underwater acoustics. Sheri L. Martinelli (Div. of Appl. Mathematics, Brown Univ., 182 George St., Providence, RI 02912)

The level set method due to Osher and Sethian [J. Comput. Phys. **79**, 12–49 (1988)] provides a way to obtain fixed grid solutions to the high-frequency wave equation. Instead of tracing rays from the source, the level set method embeds the wavefront implicitly in the phase space and propagates it according to the velocity field determined by the local ray direction, thus avoiding the complications involved in the spatial reconstruction of wavefronts from diverging rays. A level set method has been developed and implemented as a fixed-grid alternative to ray tracing to solve for the acoustic phase. One of the issues that arises with the increased dimensionality of posing the propagation problem in the level set framework is that the presence of reflecting boundaries produces a discontinuity in the phase space corresponding to a sudden change in propagation direction. When a reflecting boundary is range-dependent, further complications arise. To improve algorithm performance, specialized methods are applied to the level set equations that combine upwinding with higher-order spatial interpolation that avoid the generation of spurious oscillations that occur with most traditional finite difference methods. [Work supported by ONR 333 and the SMART Program.]

9:15

4aUW6. Two dimensional finite element propagation and reverberation modeling in shallow water waveguides. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, misakson@arlut.utexas.edu)

Finite element propagation models do not rely on approximations of the scattering at the interfaces and therefore provide excellent benchmark solutions to rough interface waveguide propagation and reverberation studies. In this study, two dimensional finite element solutions for reverberation and propagation are calculated for waveguides that have rough interfaces at the air/water boundary and the water/sediment boundary. The effects of upward and downward refracting sound speed profiles are also considered. [Work sponsored by Office of Naval Research, Ocean Acoustics.]

9:30

4aUW7. An algorithm to predict surface loss. Cathy Ann Clark (NUWCDIVNPT, B1320, R457, 1176 Howell St., Newport, RI 02841)

A semi-empirical surface loss algorithm is presented which is comprised of a rough scattering component derived from theory and a term which represents low-frequency, low angle loss from other mechanisms such as absorption and bubbles, based on a fit to measured data. A prediction of surface duct propagation using the semi-empirical algorithm and a current Navy standard propagation model is compared to measured data.

9:45

4aUW8. Boundary flattening transforms for acoustic propagation under rough and periodic sea surfaces. Roger Oba (Acoust. Div. Naval Res. Lab., Washington, DC 20375, roger.oba@nrl.navy.mil)

A conformal transform is presented that maps an acoustic domain with a one-dimensional, rough sea surface onto a domain with a flat top. The nonperturbative transform presented here broadly generalizes that of Dozier [J. Acoust. Soc. Am. **75**, 1415–1432] to include many wavelengths of the surface variation. A two-dimensional, flat-top domain permits the direct application of a parabolic equation model acoustic propagation model for the Helmholtz equation using a modified sound speed. Once the field is computed, the inverse transform permits the acoustic field interpolation in terms of the original coordinates. The mapping is derived from techniques in the classical theory of flow around an airfoil. Forward scatter test cases with periodic and rough sea surfaces provide verification of the method using a parabolic equation model. The periodic surface case demonstrates scattering from steep grazing angles to shallower ones. An extension to scattering to irregular cylinders is outlined following the scheme of DiPerna and Stanton [J. Acoust. Soc. Am. **96**, 3064–3076, (1995)]. [This research is sponsored by the Office of Naval Research.]

10:00—10:15 Break

10:15

4aUW9. Three-dimensional scattering from pressure-release rough surfaces. Sumedh M. Joshi and Marcia J. Isakson (Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758, sumedhj@mail.utexas.edu)

In order to compare a variety of three-dimensional (3-D) rough surface scattering theories, the scattering of a spherical wave incident on a pressure-release rough surface is modeled. Random surface realizations are computed from a spatial roughness power spectrum measured as part of the EVA sea test conducted in 2006. Scattering from these surfaces is computed using boundary and finite element methods. A singularity removal technique is applied to solve the Helmholtz–Kirchhoff boundary integral equation in 3-D. This integral solution is compared with 3-D finite elements and the 3-D Kirchhoff approximation, to determine the range of validity of the models.

10:30

4aUW10. Boundary roughness and the effect of internal waves on signal coherence for shallow water transmission. Harry, A. DeFerrari (Div. of Appl. Marine Phys., RSMAS - Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149)

Broadband acoustic propagation experiments at three shallow sites allow for comparison of coherency of individual surface-reflected bottom-reflected modes of propagation. There appears to be a dependence of the correlation parameters of times and length on frequency and mode number that cannot be attributed to internal waves alone and likely depends on bottom and surface roughness. At low frequencies, f<100 Hz, during periods of quiescence internal waves, all modes of propagation have equally long coherence parameters. The coherency decrease equally for all modes as internal wave activity increases. For higher frequencies, 200 Hz $<f<$1000 Hz, the coherence parameters depend on mode number, with the lower order modes always more coherent than successive higher order modes. At still higher frequencies, $f>$1000 Hz, identifiable modes are not always observed; instead there is a continuum of arriving pulse energy with very low coherency even with minimal internal waves. Apparently, the randomizing effect of internal waves depends on bottom and surface roughness and frequency. At low frequencies, the boundaries appear flat and internal waves have a minimal effect. At the highest frequencies, phase coherence is already degraded by boundary roughness so that the slightest of internal wave activity completely randomizes the signals.

10:45

4aUW11. Horizontal Lloyd mirrors arising from propagation through straight and curved nonlinear internal wave fronts. Kara G. McMahon, Laurel K. Reilly-Raska (Dept. Math. Scis., Rensselaer Poly. Inst., Troy, NY 12180, mcmahk3@rpi.edu), James F. Lynch, Timothy F. Duda (Woods Hole Ocean. Inst., Woods Hole, MA 02543), and William, L. Siegmann (Rensselaer Poly. Inst., Troy, NY 12180)

Experimental observations and theoretical studies show that nonlinear internal waves (NIWs) occur widely in shallow water and cause acoustic propagation effects including mode coupling and ducting. Horizontal ducting results when acoustic modes interact with NIW fronts that comprise waveguide boundaries. For small grazing angles between a mode trajectory and a front, an interference pattern may arise that is hypothesized [Lynch *et al.*, J. Ocean Eng. **31**, 33–48 (2006)] to be a horizontal Lloyd mirror. We examine acoustic formulations for this feature and benchmark calculations for the acoustic intensity with those from the adiabatic mode parabolic equation. Results using different waveguide features are compared, including continuous-gradient and jump sound-speed profiles of varying strengths. We focus on differences in the location of the source relative to the NIW as well as the frontal curvature. The curvature influences both incidence angles and reflection characteristics. For sources oriented inside the front, as curvature increases the areas with interference patterns shrink, while sources beyond the front cause patterns to expand. [Work supported by ONR.]

11:00

4aUW12. Blind deconvolution of remote-source signals from ocean acoustic array recordings. Shima H. Abadi (shimah@umich.edu), David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, ML 48109), and Daniel Rouseff (Univ. of Washington, Seattle, WA 98105)

Reconstructing the signal originally broadcast from a remote source in an unknown multipath environment is a task commonly known as blind deconvolution. At frequencies of several kilohertz and above, multipath shallow-ocean sound propagation may be adequately described by ray acoustics. This presentation describes results from the application of ray-based artificial time reversal (ATR) to underwater sound propagation measurements. The receiving array was vertical and it recorded signals with center frequencies and bandwidths of a few kHz at source-receiver ranges up to 3 km in a water depth of approximately 60 m. Ray-based ATR uses a simple-beam-former-determined ray-arrival direction to construct a frequency-dependent phase correction at the receiving array that allows the Green's function of the sound channel and the original source waveform to be separately estimated. Here, the correlation coefficient between the original signal and the ATR-reconstructed signal is presented as a function of range and signal-to-noise ratio. In addition, the effect of reducing the number elements of the receiving array and the use of a coherent combination of reconstructed results for various ray arrival directions on cross correlation coefficient will be discussed. [Work supported by ONR.]

11:15

4aUW13. Experimental studies of underwater acoustic communication in Trondheim fjord. Guosong Zhang (Dept. of Electr. and Telecomm., Norwegian Univ. of Sci. and Tech., O.S. Bragstads plass 2B, NO-7491, Norway, guosong.zhang@iet.ntnu.no), Hefeng Dong, and Jens M. Hovem (Norwegian Univ. of Sci. and Tech., NO-7491, Norway)

Direct-sequence spread-spectrum signal was used for communication tests over underwater channels in Trondheim fjord. Differential binary phase shift keying was utilized between two adjacent symbols. To the receiver, a method uses a time updated channel impulse response estimation to recover differential phase modulated information, and it takes the estimation from the previous symbol as the match filter. The effectiveness of this method is ensured by the coherence between two consecutive symbols over time varying channels. This method is insensitive to multipath patterns, and it does not require time synchronization as precise as the convention de-spread method does. In our experiments, good performance was achieved, even in low SNR tests. The performance loss at high SNR in the experiments was caused by long time delay spread. Late-arriving paths from the previous symbol were buried in the current symbol during time-windowing process, and the late-arriving paths might decrease the magnitude of the differential phase information. In this situation, it is prone to cause errors.

11:30

4aUW14. Frequency dispersion of parametric array signal in shallow water. Igor Esipov (N. Andreyev Acoust. Inst., 4, Shvernik str., 117036 Moscow, Russia, igor.esipov@mail.ru), Sergey Tarasov, Vasily Voronin (Inst. of Technol. in Taganrog, Russia), and Oleg Popov (N. Andreyev Acoust. Inst.)

Results of experimental test of parametric array application for marine shallow water waveguide excitation by sweep frequency modulated signal are discussed. Parametrical sound signal is forming in shallow water environment, which is stimulated by intensity modulated high frequency power acoustical pump. As a result the end-fire parametric array is forming there, which excites sharp directional signal radiation at the modulation frequency. Such a low-frequency signal, generated in the virtual end-fire array by parametrical means, will propagate in shallow water waveguide independently from the pump radiation. Shallow water signal propagation obeys to waveguide dispersion. Sweep modulated signal compression is experimentally shown for single mode signal propagation in shallow water. Acoustical signal of 2-ms duration is generated in frequency band of 7–15 kHz by parametric array. This signal is transmitted in single lobe of 2 deg width along the path of 5.6 km long in water layer from 2.5- to 3-m depth. The directivity of the signal transmitted was constant in the whole frequency range. It was shown the single mode excitation of the shallow water waveguide takes place under this circumstance. [Work supported by ISTC, Project No. 3770.]

Meeting of 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/SC 1, 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 Dr., 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 Dr., 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 Blvd., 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 Ave., 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 Ct., 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 NE 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
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200 Dixie Ave., Kalamazoo, MI 49001

V. Nedzelnitsky
U.S. Technical Advisor (TA) for IEC/TC 29 Electroacoustics
National Institite of Standards and Technology (NIST), 100 Bureau Dr., Gaithersburg, MD 20899–8221

The reports of the Chairs of these TAGs will not be presented at any other S Committee meetings.

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:

ASC S12, Noise	**22 April 2010**	**9:15 a.m. to 10:30 a.m.**
ASC S12, Noise	**22 April 2010**	**9:15 a.m. to 10:30 a.m.**
ASC S2, Mechanical Vibration and Shock	**22 April 2010**	**11:00 a.m. to 12:00 noon**
ASC S1, Acoustics	**22 April 2010**	**1:45 p.m. to 2:25 p.m.**
ASC S3, Bioacoustics	**22 April 2010**	**3:00 p.m. to 4:15 p.m.**
ASC S3/SC1, Animal Bioacoustics	**22 April 2010**	**4:30 p.m. to 5:30 p.m.**

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 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 S2/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 S3

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, Wellesly, 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 8:00 a.m. on Thursday, 22 April 2010.

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.

THURSDAY MORNING, 22 APRIL 2010 KENT A/B, 11:00 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 W. Campbell Road, P.O. Box 669, Sidney, OH 45365-0669

C. F. Gaumond, Vice Chair, ASC S2
Naval Research Laboratory, Code 7142, 4555 Overlook Ave., 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 Noise, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 8:00 a.m. on Thursday, 22 April 2010.

Scope of S2: Standards, specifications, methods of measurements and test, and terminology in the field of mechanical vibration and shock, condition 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 4pAA

Architectural Acoustics and Engineering Acoustics: Rooms for Reproduced Sound II

K. Anthony Hoover, Chair
McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362

Invited Papers

1:00

4pAA1. Measuring and defining the sound quality of audio systems. Peter Mapp (Peter Mapp Assoc., Colchester, United Kingdom)

"High quality" is a term often employed as a requirement for the performance of an audio or sound system, yet what does it mean and can it be quantified? In particular, bass performance is an important aspect but again is extremely hard to define. Frequency response assessment alone is not enough to characterize sound quality as this does not provide information with regard to the temporal or transient characteristics of the room or system. The paper will look at a number of techniques that potentially can be used to help assess sound system performance including a new measurement method intended to quantify the transient and temporal characteristic of the bass sound in a room, whether this is via a sound system or an assessment of the room itself.

1:20

4pAA2. But how does it look? Deb Britton (K2 Audio, LLC, 4900 Pearl East Circle, Ste. 201E, Boulder, CO 80301, deb@k2audio.com)

As acousticians and audio professionals, we are constantly striving for the "optimum sound solution." Quite often our optimum solution is at odds with the room's architecture or the desired aesthetic. This paper will present some examples of spaces where high-quality sound, even coverage, speech intelligibility and visual aesthetics were all achieved.

1:40

4pAA3. Sweetness spotting. Sam Ortallono (zSam Ortallono MediaTech Inst., 3324 Walnut Bend Ln., Houston, TX 77042)

This paper investigates the traditional listening position for the mixing of stereo music in a near field-monitoring situation. Common sense and current practice reinforce the ideal listening post to be at the apex of an equilateral triangle formed by the two speakers and the observer's head. This experiment will put that notion to the test. Using white noise pink noise, sine waves, and program material we will compare listening positions to the left and right as well as closer and farther. Readings from several mixing studios at multiple facilities will provide the numbers. Results will be correlated with room size, shape, and room treatment. Will the data uphold the longstanding wisdom on phase coherency and stereo image or will the results surprise us all?

2:00

4pAA4. Critical listening research on reproduced sound in diffuse environments. William L. Martens (Faculty of Architecture, Design and Planning, Univ. of Sydney, New South Wales 2006, Australia, bill@arch.usyd.edu.au) and Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774)

In critical listening rooms designed to monitor precisely small signal processing changes, what can be heard is influenced by the room's modal response and the degree to which reflections are controlled by absorptive, reflective, and diffusive boundary surfaces. In exploring the problem of room acoustics affecting the results of such important listening activities, it is informative to explore the boundary conditions. For example, the acoustics industry utilizes absorption and reverberation chambers to explore related auditory phenomena. Recently, a diffusion chamber was created with broadband diffusion on walls and ceiling. Modal frequencies were also controlled with dedicated plate resonators effective down to 50 Hz. Impulse response measurements in this diffuse chamber revealed a dense reflection pattern 30 dB or more below the direct sound. The initial perception in this room was precise sonic images, a comfortable sense of ambiance and an apparent transparency to the subtleties of reproduced imagery that support decisions of sound engineers. Related experiments being carried out in other controlled listening rooms will be reviewed to place the current work in context.

2:20

4pAA5. How reverberation degrades and aids source perception in auditory scenes. Barbara G. Shinn-Cunningham (Boston Univ. Hearing Res. Ctr., 677 Beacon St., Boston, MA 02215)

Reverberant energy (or reverberation) influences perception of source content (in addition to influencing perception of the environment and of source location). Moreover, reverberation can affect perception in different ways depending on whether a source is played in quiet or in a scene containing competing sounds. Reverberation smears spectro-temporal structure, which can interfere with estimation of duration, pitch contour, and other basic attributes. In the extreme, such spectro-temporal smearing (self-masking) can interfere with extracting the meaning of signals like speech or music. However, if the direct sound reaching a listener is nearly inaudible, reverberation adds energy, improving source detectability. While these in-quiet effects contribute to perception of sources in a complex acoustic scene, additional factors also come into play. For instance, if two otherwise similar sources have different levels of reverberation, the reverberation can make them more perceptually distinct, making it easier to focus attention on one and understand it. On the other hand, the same spectro-temporal smearing that can hurt perception of a source in quiet can also disrupt source segregation, making it difficult to selectively analyze a desired source. Examples of these different effects of reverberation on perception, taken from recent experiments, will be presented and discussed.

Contributed Papers

2:40

4pAA6. Optimal loudspeaker placement for sound field reconstruction in geometrically constrained environments. Joshua Atkins and James West (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., 3400 North Charles St., Baltimore, MD 21212)

The design of loudspeaker arrays for holographic sound reproduction systems involves sampling a given aperture by a set of loudspeakers. For full sphere apertures it can be proven that there are only five equidistant sampling methods (given by the five regular polyhedrons) which only allows for arrays with up to 20 elements. For larger arrays, non-equidistant design methods such as t-design and Gaussian sampling have been proposed. However, in realistic environments it is generally not possible to place loudspeakers on the floor or on walls at certain locations. Furthermore, it is not desirable to have the loudspeakers constrained to spherical radii. This work shows a method for loudspeaker array design that allows for consideration of these special cases. The method relies on the decomposition of the sound field into a spatially orthonormal basis of spherical harmonics. A heuristic method is used to minimize the eigenvalue spread of the spherical harmonic matrix. The loudspeaker weights are found by calculating the generalized inverse of this matrix. Constraints on the speaker positions, highest reproduction mode, and number of loudspeakers are imposed in the minimization. The results of a design for a 16 speaker array in our acoustics laboratory are discussed.

2:55

4pAA7. Acoustics in a home music listening room. Sergio Beristain (Acoust. Lab. E.S.I.M.E., P.O. Box 12-1022, Narvarte, 03020 Mexico Distrito Federal, Mexico, sberista@hotmail.com)

A room was designed with the sole purpose of listen all kinds of recorded music, with a high preference to classical and romantic music. It was decided to locate the room in the basement level of a large single family home, where the external noise control or the control of the sound going out were not the big issues, so that the internal acoustics became the main one. The acoustics was adjusted for clear and balanced sound, and the high-end audio system to be installed can handle monophonic, stereophonic, and surround signals, but stereo sources are paramount. The room is 4 m wide, 7 m long, and 2.8 m high, with the lower part of the walls used for the equipment and music source storage, while the floor, ceiling, and higher part of the walls are the places where the non-proprietary acoustic materials and the speakers are going to be located.

3:10

4pAA8. Acoustic calibration of the exterior effects room at the NASA Langley Research Center. Kenneth J. Faller, II, Stephen A. Rizzi, Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, kenneth.j.faller@nasa.gov), William L. Chapin, Fahri Surucu (AuSIM, Inc., Mountain View, CA 94043), and Aric R. Aumann (Analytical Services and Mater., Inc., Hampton, VA 23666)

The exterior effects room (EER) at the NASA Langley Research Center is a 39-seat auditorium built for psychoacoustic studies of aircraft community noise. The original reproduction system employed monaural playback and hence lacked sound localization capability. In an effort to more closely recreate field test conditions, a significant upgrade was undertaken to allow simulation of a three-dimensional (3-D) audio and visual environment. The 3-D audio system consists of 27 full-range satellite speakers and four subwoofers, driven by a real-time audio server running a derivation of vector base amplitude panning. The audio server is part of a larger simulation system, which controls the audio and visual presentation of recorded and synthesized aircraft flyovers. The focus of this work is on the calibration of the 3-D audio system, including gains used in the amplitude panning algorithm, speaker equalization, and absolute gain control. Because the speakers are installed in an irregularly shaped room, the speaker equalization includes time delay and gain compensation due to different mounting distances from the focal point, filtering for color compensation due to different installations (half space, corner, and baffled/unbaffled), and crossover filtering.

Session 4pAB

Animal Bioacoustics, Signal Processing in Acoustics, and Noise: Topical Meeting on Signal Processing of Subtle and Complex Acoustic Signals in Animal Communication II: Automated Classification of Animal Acoustic Signals II

Ann E. Bowles, Cochair
Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109

Sean K. Lehman, Cochair
Lawrence Livermore National Lab., Livermore, CA 94550

Chair's Introduction—1:10

Contributed Papers

1:15

4pAB1. Classification of marine mammal vocalizations using an automatic aural classifier. Paul C. Hines and Carolyn M. Ward (Defence R&D Canada, P.O. Box, 1012 Dartmouth, Nava ScotiaS B2Y 3Z7, Canada, paul.hines@drdcrddc.gc.ca)

Passive sonar systems are often used to detect marine mammal vocalizations in order to localize and track them. Unfortunately, transients generated by sources other than marine mammals can also trigger passive sonar systems which leads to a large number of false alarms. Furthermore, even in the case of a successful detection, classifying the genus is often required and this typically requires expertise not readily available on the vessel. Perceptual signal features—similar to those employed in the human auditory system—have been used to reduce false alarms in active sonar by automatically discriminating between target and clutter echoes. This contributes to improved sonar performance [Young and Hines, J. Acoust. Soc. Am. **122**, 1502–1517 (2007)]. Many of the features were inspired by research directed at discriminating the timbre of different musical instruments (a passive classification problem) which suggests it might be applied to classify marine mammal vocalizations. To test this hypothesis, the automatic aural classifier was trained and tested on a set of marine mammal vocalizations from a variety of species. This paper will provide an overview of the aural classifier's architecture, describe the preparation of the data set, including the attempt to provide ground-truth confirmation of the data, and present some preliminary results.

1:30

4pAB2. Processing burst pulse waveforms from odontocetes. Paul Hursky, Michael B. Porter (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037), Tyler Olmstead, Marie Roch (San Diego State Univ., San Diego, CA 92182-7720), Sean Wiggins, and John Hildebrand (Univ. of California in San Diego, La Jolla, CA 92093)

A number of marine mammals produce an interesting vocalization called a burst pulse, which consists of a series of clicks whose repetition rate increases to the point that the entire waveform sounds more like a continuous buzz. These animals also produce individual clicks at a much less frequent rate, sometimes with a consistent inter-click interval, sometimes without. However, because these animals are often observed in groups that number in the hundreds at low rates, it is difficult to associate individual clicks with individual animals or to group multipath arrivals. A burst pulse waveform avoids these association ambiguities—the clicks making up a burst pulse are so frequent, and so consistent in amplitude and inter-click interval that they are easily associated with a single animal, and any multipath structure is consistently revealed. We will present results of processing a variety of burst pulses observed on array sensors from a series of experiments off the coast of California and discuss algorithms for detecting burst pulse waveforms and for extracting classification features from these waveforms.

1:45

4pAB3. Probabilistic bioacoustic signal extraction within spectrograms. T. Scott Brandes (Signal Innovations Group, Inc., 1009 Slater Rd., Ste. 200, Res. Triangle Park, NC 27703)

A methodology for probabilistic bioacoustic signal extraction within spectrograms of natural sound recordings is proposed. Probabilistic models of signal attributes are described for generating a host of likelihoods used to estimate the probability that individual pixels within a spectrogram represent part of a bioacoustic signal. The pixel probabilities result in a transformation of the spectrogram into a probability map of bioacoustic signal presence. It is shown that these probability maps create a dramatic increase in the bioacoustic signal-to-noise ratio within the spectrogram. These probability maps along with threshold filtering provide a means for image segmentation of the spectrogram, creating blocks of pixels that represent bioacoustic signals, facilitating feature and signal extraction. This methodology is applied to natural sound recordings of three quality types in a wide range of signal-to-noise ratios. In each instance, the probability mapping greatly increases the signal-to-noise ratio and, when applied as a threshold filter, is far more selective with pixel inclusion than threshold filtering applied based on sound level. Suggested applications include automated call recognition of birds, frogs, and insects from field recordings within a wide range of ambient noise.

2:00

4pAB4. Recognition of killer whale individuals from multiple stereotyped call types. Nicole Nichols, Les Atlas (Dept. of Elec. Eng., Univ. of Washington, Box 352500, Seattle, WA 98195, nmn3@u.washington.edu), Ann Bowles (Hubbs-SeaWorld Res. Inst., San Diego, CA 92109), and Marie Roch (San Diego State Univ., San Diego, CA 92182)

Recognition of marine mammal individuals will be an essential tool for obtaining demographic information of populations as habitat monitoring is increasingly being conducted with passive acoustics. Our previous research established the potential to identify killer whale individuals with passive acoustic recordings with up to 78% accuracy. The data for this result consisted of 75 vocalizations from four adult (two male, two female) killer whales of Icelandic origin in residence at SeaWorld San Diego. The stereotyped vocalizations used were all SD1a or SD1b, and caller identification was known to a high degree of certainty allowing for quantifiable validation. Classification was performed with 12 mel-frequency cepstral coefficients, and 12 delta and 12 delta-delta cepstral coefficients as inputs to a Gaussian mixture model. We initially chose to use one call type as the SD1 call was frequently produced by all four individuals and it eliminated potential ambiguity from identifying call type instead of individual. In the research we now present, we extend our prior methods to be used with multiple call types. Because individuals in the wild will not have a known frequency of call type usage, we propose a normalization method for training a classifier that is unbiased with respect to call-type frequency.

2:15

4pAB5. Autonomous underwater glider based embedded real-time marine mammal detection and classification. Tyler J. Olmstead, Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182, tyler.j.olmstead@gmail.com), Paul Hursky, Michael B. Porter (Heat, Light, and Sound Res. Inc., La Jolla, CA 92037), Holger Klinck, David K. Mellinger (Oregon State Univ., Newport, OR 97368), Tyler Helble, Sean S. Wiggins, Gerald L. D'Spain, and John A. Hildebrand (Univ. of California in San Diego, La Jolla, CA 92093)

Autonomous marine vehicles offer the potential to provide low-cost data suitable for passive acoustic monitoring applications of marine mammals. Due to their extremely low-power consumption and long range, gliders are an attractive option for long-term deployments. Challenges related to power availability, payload size, and weight have previously restricted the viability of marine mammal monitoring. As an example, the wide bandwidth of odontocete echolocation clicks requires a high sampling rate and poses challenges with respect to limitations in power, size, and weight of the deployed system. Recent developments in commercial off-the-shelf hardware driven by the mobile phone industry's need for multimedia-rich smart phones have resulted in low-power architectures capable of performing computationally demanding signal processing and stochastic recognition tasks in real time. We describe our work on a small form-factor, light-weight package used to perform real-time passive acoustic detection and classification. The system detects echolocation clicks using Teager energy. Echolocation clicks are then classified using cepstral features processed by a Gaussian mixture model. [Work sponsored by ONR.]

2:30

4pAB6. Analysis of cetacea vocalizations using ocean bottom seismic array observations, western Woodlark Basin, Papua New Guinea. Scott D. Frank (Dept. of Mathematics, Marist College, 3399 North Rd., Poughkeepsie, NY 12601, scott.frank@marist.edu) and Aaron N. Ferris (Weston Geophysical Corp., 181 Bedford. St., Lexington, MA 02420)

Over a 6-month period in 1999 an array of seven ocean bottom seismometers and seven bottom mounted hydrophones covered a 30-km^2 area at 2–3-km water depth in the western Woodlark Basin. The array's main task was the detection of microearthquakes associated with nearby active tectonics. However, it also made fortuitous broadband recordings of Cetacea vocalizations that we associate with finback whales. Each of these signals has a 10–15-s duration and they exhibit a repetitive pattern with 10–15-min period. The signals are well recorded across both the seismometers and hydrophones, which were sampled at 64 and 125 Hz, respectively. Time-varying spectral analysis demonstrates that the signals are frequency modulated in the 20–30-Hz band with most energy occurring at $\sim$22 Hz. We employ array-based methods (e.g., optimal beam forming techniques) and acoustic transmission loss simulations to determine range and bearing of the acoustic sources. Preliminary analysis indicates that the acoustic sources originate near the array, and secondary signals may represent back-scattered energy from the short-wavelength, high-topographic features associated the active tectonics with in the basin.

2:45

4pAB7. Echolocation behavior of pairs of flying *Eptesicus fuscus* recorded with a Telemike microphone. Mary E. Bates (Dept. of Psych., Brown Univ., 89 Waterman St., Providence, RI 02912, mary_bates@brown.edu), Yu Watanabe, Yuto Furusawa, Emyo Fujioka, Shizuko Hiryu, Hiroshi Riquimaroux (Doshisha Univ., Kyotanabe, Japan), Jeffrey M. Knowles, and James A. Simmons (Brown Univ., Providence, RI 02912)

Four big brown bats (Eptesicus fuscus) were flown singly and in pairs in a room containing a sparse array of vertically hanging plastic chains as obstacles. Each bat carried a lightweight radio telemetry microphone (Telemike) that recorded their emitted echolocation sounds without artifacts from Doppler shifts, directional effects, and atmospheric attenuation. The broadcasts of both bats were also recorded with two stationary ultrasonic microphones located at the far end of the flight room. The echolocation broadcasts of bats flying singly were compared to those emitted when the bats were flown together. The principal change was shifting of harmonic frequencies very slightly ($<$5 kHz) away from each other and from frequencies used when flying alone. In contrast, the duration of emissions was more stable between single and double bat flights. Changes in ending frequency have been associated with a jamming avoidance response in big brown bats and could indicate attempts to avoid interference while flying with conspecifics in an enclosed space. [Work supported by ONR and NSF.]

3:00

4pAB8. Paradoxical reference frequency shifts during paired flights in Japanese horseshoe bats evaluated with a Telemike. Yuto Furusawa, Shizuko Hiryu, and Hiroshi Riquimaroux (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani Tatara, Kyotanabe 610-0321, Japan)

In order to examine vocal behaviors for the echolocating bats to improve extraction of jammed echolocation signals, group flight experiments were conducted where two Japanese horseshoe bats were flown simultaneously in a flight chamber. Horseshoe bats are known to conduct Doppler-shift compensation, keeping their echo CF2 frequency constant to which their auditory system is sharply tuned. It was hypothesized that bats shifted their reference frequencies apart to avoid overlapping the reference frequencies. Echolocation sounds during sequential flights were recorded with an onboard wireless microphone, Telemike. Reference frequencies were investigated by measuring Doppler-shift compensated CF2 frequencies of the bats' own echoes. Recorded data show that the bats kept their echo CF2 frequency constant with a standard deviation (SD) of 50–80 Hz, indicating that they exhibited Doppler-shift compensation with the same accuracy as that used for flying alone (SD = 83.2 Hz). However, the bats did not shift their reference frequency toward increasing the frequency difference even when the initial difference in the reference frequencies between individuals was less than 100 Hz. Rather, the bats tended to make their reference frequencies closer during group flight compared to those for single flights. These findings suggest that multiple bats may squeeze their echo CF2 frequencies into a narrow frequency range. In addition, horseshoe bats may adapt to acoustic interference without expanding the frequency difference between individuals. [Work supported by ONR grant.]

3:15

4pAB9. Miniaturization of insect-inspired acoustic sensors. Erick Ogam (Laboratoire de Mcanique et d'Acoustique, CNRS, UPR7051, 31 chemin Joseph Aiguier, 13402 Marseille Cedex 20, France), Franck Ruffier (Institut des Sci. du Mouvement, 13009 Marseille France), Armand Wirgin (CNRS, 13402 Marseille Cedex 20, France), and Andrew Oduor (Maseno Univ., Maseno, Kenya)

Insects can navigate and follow sound sources with precision in complex environments using efficient principles based on particular and appropriate sensory-motor processing. It is precisely these principles, little understood at this point, that we want to understand, model, and validate by rebuilding them on a micro-flying device. To achieve this goal, a new bio-mimetic acoustic sensor inspired by the cricket's auditory system has been developed and tested in an anechoic chamber. The performance of the processing algorithm, the aperture of the auditory system, and interferences caused by the geometry of the sensor itself were first evaluated: synthetic signals were generated by a boundary element model taking into account the three-dimensional geometry of the sensor to characterize its performance in free space. A processing based on cross-correlation for localizing and tracking an acoustic source was also studied to assess whether the observed limitations were inherent to the geometry of the sensor or due to limitations of the cricket inspired processing algorithm. The sensor worked well with several synthetic chirped cricket songs. [Work supported by CNRS PIR Neuroinformatique grant SonoBot.]

3:30—3:45 Break

3:45—4:30 Demonstrations and Discussions

Session 4pAO

Acoustical Oceanography and Underwater Acoustics: Impact of Shallow Water Acoustic Propagation by Linear Internal Waves and Neutrally Buoyant Intrusions

Dajun Tang, Cochair
Univ. of Washington, Applied Physics Lab., 1013 NE 40th St., Seattle, WA 98105

David L. Bradley, Cochair
Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804–0030

Chair's Introduction—1:00

Invited Papers

1:05

4pAO1. Three-dimensional finestructure and vertical microstructure in thermohaline intrusions. Michael C. Gregg (522 Henderson Hall, Univ. Washington, 1013 NE 40th St., Seattle, WA 98105), Andrei Y. Shcherbina, Matthew H. Alford, and Ramsey R. Harcourt (Univ. Washington, Seattle, WA 98105)

Thermohaline intrusions have intrigued physical oceanographers since the first XBT section was published by Spilhaus in 1939. Characterized by vertical gradients of temperature and salinity having compensating density effects, intrusions are meters to tens of meters thick and extend laterally for hundreds of meters to many kilometers. Boundaries often have TS staircases produced by salt fingering or diffusive layering. Large diapycnal fluxes have been inferred from sections showing intrusions sloping across isopycnals, but these inferences are questionable, as isopycnal movements can give the illusion of diapycnal motion. To address these issues, in 2007 we used a depth-cycling towed body, a neutrally buoyant Lagrangian float, and microstructure profilers to map and follow intrusions in the subtropical front north of Hawaii. All intrusions were connected to the front, i.e., none was isolated lenses, some as long, thin folds, and others resembled tongues. About 10 m thick, the sheets protruded 10 km into the warm, saline side of the front, were coherent laterally for 10–30 km, and lasted at least 1 week, evolving at both inertial (23 h) and sub-inertial ($\sim$10 day) time scales. They appeared to be formed by advective deformation of the front driven by sub-mesoscale two-dimensional motions.

1:25

4pAO2. Internal waves, intrusions, and coastal zone acoustics. Timothy F. Duda, Andone C. Lavery (Dept. AOPE, MS 11, Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, tduda@whoi.edu), and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Ubiquitous linear internal waves are known to dominate fluctuations of long-range low-frequency sound in the deep ocean. In the coastal zone, however, variations in low-frequency-sound propagation caused by large episodic nonlinear internal waves are often dominant and have been studied in detail. Apart from this, sound-speed anomalies from smaller but pervasive linear internal waves can also influence propagation, as do density-compensated temperature/salinity intrusions found in areas near fronts. The physical nature of each of these two phenomena and existing models of their effects on propagation will be reviewed. The possibility of using high-frequency acoustics to remotely examine and map intrusions via double-diffusive microstructure scattering will also be examined. The anisotropy of the coastal internal wave field will be discussed, and subsequent acoustic propagation anisotropy will be examined using data from the multi-PI Shallow-Water 2006 experiment in the Mid-Atlantic Bight. [Research supported by the Office of Naval Research.]

Contributed Papers

1:45

4pAO3. Analysis of horizontal coherence during the transverse acoustic variability experiment. Peter C. Mignerey and David J. Goldstein (Acoust. Div. 7120, Naval Res. Lab., Washington, DC 20375)

The transverse acoustic variability experiment took place in the northern limit of the East China Sea in 65–80 m of water. 300-Hz and 500-Hz cw acoustic signals were transmitted over distances of 33 and 20 km to a broadside HLA where they were received at 22–27 dB SNR. Fourteen estimates of transverse mutual-coherence functions show high correlation with scale lengths on the order of 1000–1700 m. A towed CTD chain provided simultaneous measurements of the sound-speed fluctuation spectrum due to internal waves in an effort to determine whether the observed shallow-water coherence is understandable within the framework of a normal-mode formulation of path-integral theory. An important property of the mutual-coherence function is its behavior at small hydrophone separations. Here the phase-structure function is expected to follow a power law. Measured logarithmic phase-structure function have slopes that vary between 0.6 and 1.1, which disagree with path-integral solutions that predict slope 2. The discrepancy between observation and data is not currently understood. [Work supported by the Office of Naval Research.]

2:00

4pAO4. Performance of the transverse acoustic variability experiment horizontal line array. David J. Goldstein and Peter C. Mignerey (Acoust. Div. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, david.goldstein@nrl.navy.mil)

The transverse acoustic variability experiment was designed to study the impact of shallow-water environmental fluctuations on acoustic coherence in the horizontal plane, transverse to the direction of propagation. The experiment was conducted during August 2008 in the East China Sea. A criti-

cal component to the success of the experiment was a two-segment 96-channel horizontal line array, moored on the sea floor at an approximate depth of 80 m. An analysis of the performance of this array is presented, including pulse arrival time statistics, beam patterns, gains, and signal-to-noise ratio. Methods used to determine the relative positions of array hydrophones are also discussed. [Work supported by the Office of Naval Research. We thank the staff at KORDI and the crews aboard the R/V Eardo and Sunjin for their assistance and bountiful hospitality.]

2:15

4pAO5. Mode-2 internal waves impact on acoustic signal properties. Marshall Orr (Acoust. Div., The Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375)

The properties of mode-2 internal waves repeatedly observed on the New Jersey Shelf (USA) with high-frequency acoustic flow visualization systems will be overviewed. The impact of the mode-2 internal waves on the range/depth distribution of acoustic signal properties will be discussed.

2:30

4pAO6. Effects of anisotropic internal waves on acoustic propagation within the East China Sea. Chad M. Smith (cms561@psu.edu) and David L. Bradley (Graduate Program in Acoust., Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030)

Using environmental and acoustic data from the TAVEX 2008 Korean /U.S. cooperative experiment, the anisotropic spatial properties of internal waves recorded during CTD tows and their connection to temporal variations in recorded acoustic data from a horizontal line array will be discussed. Structure of internal waves, wave travel speed and trajectory, as well as computational modeling of the acoustic field and its comparison with recorded data will also be covered. [Work supported by the Office of Naval Research Contract No. N00014-08-1-0455.]

2:45

4pAO7. Adjoint study of water column variability and propagation in the transverse acoustic variability experiment (TAVEX 2008). Michelle M. Kingsland and David L. Bradley (Appl. Res. Lab., Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16801, mbm224@psu.edu)

Internal wave and other water column activity in the conductivity, temperature, and depth (CTD) data collected in the East China Sea as a part of TAVEX 2008 are used as a basis for analyzing acoustic propagation at 500 Hz between a fixed source and fixed horizontal line array placed on the seafloor. Because the CTD data were not taken coincident with the acoustic propagation path (in general), adjoint modeling methods will be used to study acoustic data and prediction misfit. Usefulness of the adjoint method for these experimental conditions will be discussed. [Work supported by ONR.]

3:00

4pAO8. Periodicity of intensity fluctuations in shallow water in the presence of solitary internal waves. T. C. Yang (Naval Res. Lab., Washington, DC 20375, tc.yang@nrl.navy.mil)

This paper investigates the intensity variation with time of a broadband signal in the presence of moving solitary internal waves. The autocorrelation of the received impulse responses summed over a vertical line array of receivers (the so called Q function) is measured from the SWARM95 data using broadband pulses received on the NRL array at a range ~42 km from the source. The signals are m-sequences centered at 400 Hz with a bandwidth of 100 Hz. The peak of the Q function shows a periodicity of 0.042 cycles/min, with a value changing with time depending on the position of the internal wave packets. Coupled mode equation is used to calculate mode coupling induced by the solitary internal waves and explain the observed periodicity of the intensity fluctuations. The results suggest that the positions and some parameters of the solitary waves could be remotely estimated from underwater sensors. [Work supported by the Office of Naval Research.]

3:15—3:30 Break

Invited Papers

3:30

4pAO9. Observed space-time scales of internal waves and finescale intrusions on the New Jersey Shelf and their likely acoustic implications. John Colosi (Naval Postgrad. School, Monterey, CA 93943), Tim Duda, Ying-Tsong Lin, Jim Lynch, and Arthur Newhall (Woods Hole Oceanograph. Inst., Woods Hole MA 02543)

It has been known for some time that ocean density surfaces possess multi-scale variations in temperature (T) and salinity (S), termed spice (hot and salty, cold and fresh) by Munk in 1981. Compensating T and S variations along a surface of constant density have reinforcing sound speed anomalies and thus have important acoustic implications. With the advent of new technologies to observe small scale, rapidly changing ocean thermohaline structure, spice variations, and their acoustic effects are just starting to be understood. Better known are the impacts of internal waves which vertically advect ocean sound speed structure. This talk will present analysis of moored T and S observations on the New Jersey Shelf region during the SW06 experiment in which the sound speed effects of internal waves and spice can be approximately separated. These results will be contrasted with similar measurements made in the deep waters of the Philippine Sea. Using recent results from coupled mode theory, some discussion of the acoustic implications of the internal wave and spice structure will be presented. In particular, the phase randomizing effects of linear internal waves and spice will diminish the mean acoustical influence of intense but localized nonlinear internal waves.

3:50

4pAO10. Characterizing water column variability and its impact on underwater acoustic propagation. Kyle M. Becker (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030) and Megan S. Ballard (Appl. Res. Labs., Univ. Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

During the Shallow Water Experiment 2006 (SW06), a low-frequency acoustic source was towed out and back along radials from a receiver array located at the origin. Measurements were made along three radials, separated by 45 deg, with each track repeatedly traversed over a period of 6–10 h. Concurrently with the acoustic measurements, the water column sound speed field was estimated both along the source track and at the receiver location. The objective was to fully characterize the environment in the water column over the acoustic path during transmissions. In this way, variability in the acoustic field measured along the repeated tracks could be associated with variability in the water column. The variability in the water column, measured independently at the source and at the receiver, is examined and related to variability observed in the acoustic fields. Spatial and temporal variability is characterized on transect-to-transect, radial-to-radial, and day-to-day basis. Over 3 days, variability in the water column could be attributed to thermohaline intrusions, linear and non-linear internal waves, as well as mesoscale variability. This talk will focus on the requirements for characterizing these features and understanding their impact on acoustic propagation and prediction. [Work supported by ONR.]

Contributed Papers

4:10

4pAO11. Reverberation clutter from combined internal wave refraction and bottom backscatter. Dajun Tang and Frank Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

A hypothesis is proposed that one mechanism for clutter observed in shallow water reverberation measurements is due to the combined effect of forward scatter and subsequent backscatter. The forward scatter refracts part of the sound from low grazing angle to high grazing angle, then being backscattered by bottom roughness. The refracted sound impinges onto the bottom at higher grazing angle, resulting in higher backscatter because of elevated bottom scattering cross section as compared to that at lower grazing angles. The resultant reverberation will stand out as a target-like clutter. An example is presented as a non-linear internal wave propagates in a shallow water channel, resembling conditions found on the New Jersey Shelf. The effect on reverberation of the internal wave as clutter is investigated using a time-domain numerical model. The model uses parabolic equation for the two-way propagation and first order perturbation approximation for bottom backscatter. [Work supported by ONR.]

4:25

4pAO12. Acoustic propagation effects from saline intrusions across the shelfbreak front off the New Jersey coast in summer. Arthur E. Newhall, Ying-Tsong Lin, James F. Lynch, Timothy F. Duda, Glen G. Gawarkiewicz (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and John A. Colosi (Naval Postgrad. School, Monterey, CA 93943)

In the summer of 2006, the multi-task, interdisciplinary Shallow Water 2006 (SW06) experiment was conducted on the continental shelf and shelf-break regions off the New Jersey coast. During SW06 the shelfbreak front variability was clearly characterized by combined measurements of conductivity-temperature-depth casts, long-period (5 weeks) and quick sampling oceanographic sensor moorings, and surveys with a Scanfish, a ship-towed undulating vehicle carrying multiple sensors. This data set indicates that the slope water penetrated onto the shelf via neutrally buoyant intrusions. Due to the higher temperature and salinity, these frontal intrusions, seen as thin layers near the seasonal pycnocline, increased the local sound speed and accounted for variations in acoustic propagation. In this paper, we will use field data and PE numerical acoustic simulations to study the impact of these frontal intrusions. Specific examples on the creation of vertical acoustic double ducting and horizontal ducting formed by these intrusions and their modulation by linear internal waves are discussed. [Work is supported by the Office of Naval Research.]

4:40

4pAO13. Propagation of low-frequency sound through densely sampled internal waves. Megan S. Ballard (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758, meganb@arlut.utexas.edu) and Kyle M. Becker (Appl. Res. Lab., Penn State, State College, PA 16804)

It is well known that nonlinear internal gravity waves can have a significant effect on acoustic propagation. In particular, horizontal refraction and ducting of sound can occur when the acoustic propagation path is aligned with fronts of internal waves. Additionally, more complicated effects are associated with internal waves that are curved and/or truncated. In this talk, observations of internal waves measured on the New Jersey shelf area of the north Atlantic and their effects on the acoustic field are presented. During the experiment, a low-frequency source broadcasting continuous tone was towed repeatedly out and back along radials with respect to a VLA. The ship's track was oriented parallel to the shelf break so that the acoustic propagation path was roughly aligned with internal waves propagating up the shelf. Internal waves were measured at the location of the receivers by temperature sensors on the VLA, at the location of the source by a towed CTD chain, as well as by a cluster of 16 environmental moorings located adjacent to experiment site. The high-spatial sampling of the environment allows for identification of range-varying features of internal waves.

4:55

4pAO14. Horizontal reflection of a low-frequency sound signal from a moving nonlinear internal wave front. M. Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716), J. F. Lynch, Y.-T. Lin (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and B. G. Katsnelson (Voronezh Univ., Voronezh 394006, Russia)

Simultaneous measurements of acoustical and internal waves are reported while an internal wave approaches an acoustic track during the SW06 experiment. The incoming internal wave packet acts as a moving layer reflecting and refracting acoustic waves in the horizontal plane. The mechanism of this interaction is shown in received acoustic data on a vertical hydrophone array using a modal approach. It is shown that the wave front of internal waves behaves as selective filter, depending on the mode number and frequency of the broadband signal. In other words the reflection coefficient in horizontal plane depends on mode number and frequency. Experimental data analysis shows good agreement with the theory. [Work supported by ONR 321OA.]

5:10

4pAO15. Acoustic wave propagation and scattering in turbulence and internal waves. Tokuo Yamamoto N (Div. of Appled Marine Phys., RSMAS, Univ. of Miami, Miami, FL 33149)

Acoustic wave propagation and scattering in turbulence and internal waves in Kanmon Strait and the outer New Jersey shelf are measured in a bistatic source-receiver configuration at frequency 5500 Hz. While internal wave dominates on the open New Jersey shelf, the turbulence dominates in the Kanmon Strait channel. Initially strong internal wave scattering is reduced rapidly due to strong mixing in the shallow strait. On the flat and open continental shelf, internal wave and warm water intrusion dominate while turbulence is near absent. The parabolic equation code predictions agree with data very well. The bistatic scattering data are processed for bistatic angles and the effects of Doppler shift on the mean flow and the broadening of peek spectra on the turbulent fluctuations. The mean flow and turbulence fluctuation obtained from the Doppler shift agreed with the two parameters obtained by the reciprocal acoustic transmission.

5:25

4pAO16. Using ocean ambient noise cross-correlations for noise-based acoustic tomography. Charlotte Leroy, Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405), Philippe Roux (LGIT, Grenoble, France), William Kuperman, William Hodgkiss, and Shane Walker (UC San Diego, La Jolla, CA)

Recent theoretical and experimental studies have demonstrated that an estimate of the Green's function between two hydrophones can be extracted passively from the cross-correlation of ambient noise recorded at these two points. Hence monitoring the temporal evolution of these estimated Green's functions can provide a means for noise-based acoustic tomography using a distributed sensor network. However, obtaining unbiased Green's function estimates requires a sufficiently spatially and temporally diffuse ambient noise field. Broadband ambient noise ([200 Hz–20 kHz]) was recorded continuously for 3 days during the SWAMSI09 experiment (next to Panama City, FL) using two moored vertical line arrays (VLAs) spanning the 13-m water column and separated by 150 m. The feasibility of noise-based acous-

tic tomography was assessed in this dynamic coastal environment over the whole recording period. Furthermore, coherent array processing of the computed ocean noise cross-correlations between all pairwise combinations of hydrophones was used to separate acoustic variations between the VLAs caused by genuine environmental fluctuations–such as internal waves–from the apparent variations in the same coherent arrivals caused when the ambient noise field becomes strongly directional, e.g., due to an isolated ship passing in the vicinity of the VLAs.

THURSDAY AFTERNOON, 22 APRIL 2010 GALENA, 12:55 TO 6:00 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Ultrasonically Activated Agents

Oliver Kripfgans, Chair
Univ. of Michigan, Radiology Dept., Zina Pitcher Pl., Ann Arbor, MN 48109-0553

Chair's Introduction—12:55

Invited Papers

1:00

4pBB1. Cavitation and vaporization threshold of perfluorocarbon droplets. Kelly Schad and Kullervo Hynynen (Dept. Medical Biophys., Univ. Toronto, 2075 Bayview Ave., Toronto, ON, Canada)

Focused ultrasound therapy can be enhanced with microbubbles by thermal and cavitation effects. However, localization of treatment becomes difficult as bioeffects can occur outside of the target region. Spatial control of gas bubbles can be achieved with acoustic vaporization of perfluorocarbon droplets. This study was undertaken to determine the acoustic parameters for bubble production by droplet vaporization and how it depends on the acoustic conditions and droplet physical parameters. Droplets of varying sizes were sonicated *in vitro* with a focused transducer and varying frequencies and exposures. Simultaneous measurements of the vaporization and inertial cavitation thresholds were performed. The results show that droplets cannot be vaporized at low frequency without inertial cavitation occurring. However, the vaporization threshold decreased with increasing frequency, exposure, and droplet size. In summary, we have demonstrated that droplet vaporization is feasible for clinically relevant sized droplets and acoustic exposures.

1:20

4pBB2. Applications of acoustic droplet vaporization in diagnostic and therapeutic ultrasound. Mario L. Fabiilli, Oliver D. Kripfgans, Man Zhang, Kevin J. Haworth, Andrea H. Lo, Paul L. Carson, and J. Brian Fowlkes (Dept. of Radiology, Univ. of Michigan, 3225 Med. Sci. I, 1301 Catherine St., Ann Arbor, MI 48109, mfabiill@umich.edu)

Micron- and nano-sized colloids are being studied in diagnostic and therapeutic applications of ultrasound (US). Unlike clinically utilized microbubbles, emulsions possess unique physiochemical properties that could translate into distinct, clinical benefits beyond conventional contrast agents. Droplets, composed of a superheated liquid, can be phase transitioned into bubbles using US, a process known as acoustic droplet vaporization (ADV). Droplets, transpulmonary in size, transition into bubbles upon ADV and can reach diameters that occlude capillaries and arrest blood flow in the vascular bed. Examples of ADV in diagnostic and therapeutic applications will be presented. First, ADV has been used in phase aberration correction in transcranial US imaging. Second, ADV-generated microbubbles can reduce and occlude renal perfusion *in vivo*. Third, the effects of thermal therapy using high-intensity focused ultrasound (HIFU) has been enhanced and controlled more effectively using ADV. Fourth, ADV has been utilized as a release mechanism for therapeutic agents that are incorporated into the emulsion. In all applications, the physiochemical properties of the droplets coupled with the spatial and temporal control afforded by ADV-generated microbubbles are crucial to the success of each ADV development. [This work was supported in part by NIH Grant 5R01EB000281.]

1:40

4pBB3. Dodecaperfluoropentane emulsion for oxygen delivery. Evan Unger, Terry Matsunaga, Russel Witte, Ragnar Olafsson (Dept. of Radiology, Univ. of Arizona, 1501 N. Campbell Ave., Tucson, AZ 85724), Arthur Kerschen, Melissa Dolezal, and Jenny Johnson (NuvOx Pharma, LLC, 1635 E. 18th St., Tucson, AZ 85719)

Dodecaperfluoropentane emulsion (DDFPe) was prepared from DDFP and surfactant PEG-Telomer-B and studied for particle sizing, storage stability, acoustically *in vitro* at 37 °C, and for oxygen transport in porcine model of hemorrhagic shock and as radiation sensitizer in murine model of tumor hypoxia. DDFPe had mean particle size of about 200 nm stable on storage at room temperature for over 2 years. Compared to microbubbles of perfluorobutane (PFB), more than 100-fold higher DDFPe was needed for comparable acoustic backscatter compared to PFB. Acoustic reflectivity increased with higher MI ultrasound but was still much less for DDFPe than for PFB. DDFPe was highly effective oxygen therapeutic in models of hemorrhagic shock and tumor hypoxia. A dose of as little as 0.6 cc/kg of DDFPe was sufficient to prevent death from hemorrhagic shock. In hypoxic tumors a similar dose reversed tumor hypoxia and restored sensitivity to radiation therapy. No ultrasound was used in these experiments for *in vivo* oxygen delivery. Compared to liquid perfluorocarbons which have been studied as oxygen therapeutics, DDFPe uses a much lower dose, about 1%. DDFPe is a promising oxygen transport therapeutic. Additional work is underway to try to develop DDFPe for this application.

2:00

4pBB4. Phase change nanodroplet for sensitizer for thermal and cavitational effects of ultrasound. Ken-ichi Kawabata, Rei Asami, Hideki Yoshikawa, Takashi Azuma (Life Sci. Res. Ctr., Central Res. Lab., Hitachi, Ltd., Higashi-Koigakubo, Kokubunji, Tokyo 185-8601, Japan, kenichi.kawabata.ap@hitachi.com), and Shin-ichiro Umemura (Tohoku Univ., Aramaki, Aoba-ku, Sendai 980-8579, Japan)

Although HIFU therapy is very low invasive method for tumor treatment, its time inefficiency and the absence of focus detecting mechanism of current systems limit the number of applicable cases. For an improved HIFU tumor treatment system, we propose to utilize a non-echogenic liquid nanodroplet which turns into highly echogenic microbubbles upon non-therapeutic ultrasound pulses. Such a nanodroplet would give echographic information on the focus of HIFU when exposed to the pulse and accelerate HIFU treatment by enhancing the ultrasonic energy deposition as microbubble. As such a nanoparticle system, we developed a phase change nanodroplet (PCND) which consists of superheated perfluorocarbon nanodroplet coated with PEGylated phospholipids. Significance of PCND to enhance ultrasonically induced therapeutic effects was investigated on mice, rats, and rabbits. At frequencies of 2 and 3 MHz, it was found that the presence of PCND (and generated microbubbbles) halved the intensity threshold for inducing thermal damage. At frequency around 1 MHz, it was found that the PCND halved the threshold for inducing damage in murine tumor, and the damage was induced both by thermal and cavitational effects of ultrasound. [This work was in part entrusted by the New Energy and Industrial Technology Development Organization, Japan.]

2:20

4pBB5. Relationship between cavitation, rapid loss of echogenicity, and drug release from echogenic liposomes. Kirthi Radhakrishnan (Dept. of Biomedical Engg., Univ. of Cincinnati, 231 Albert Sabin Way, MSB 6155, Cincinnati, OH 45267, radhakki@mail.uc.edu), Jonathan A. Kopechek, Kevin J. Haworth (Univ. of Cincinnati, Cincinnati, OH), Shaoling Huang, David D. McPherson (Univ. of Texas Health Sci. Ctr., Houston, TX), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH)

Echogenic liposomes (ELIPs), encapsulating air and drug, are being developed for use as vesicles for ultrasound-mediated drug release. Both calcein and a thrombolytic drug have been encapsulated in ELIP and released with 6-MHz color Doppler ultrasound in an *in vitro* flow system. To elucidate the role of stable or inertial cavitation in loss of echogenicity and drug release, ELIPs were insonified with 6-MHz pulsed Doppler ultrasound within the same flow system. A 10-MHz focused passive cavitation detector (PCD) was placed confocally with the pulsed Doppler sample volume. B-mode images and PCD signals were recorded during ELIP infusion (5 ml/min) and exposed to pulsed Doppler over a range of peak rarefactional pressure amplitudes (0–1.4 MPa) at two pulse repetition frequencies (1250 and 2500 Hz). The mean digital intensities were calculated on each image within two regions of interest placed upstream and downstream of the Doppler sample volume. The thresholds for rapid loss of echogenicity were ascertained from six B-mode images acquired over 30 s. These thresholds were compared to stable and inertial cavitation thresholds. The implications of these data on the mechanism of rapid liberation of gas and drug release will be discussed. [Work supported by the NIH R01 HL059586 and NIH R01 HL7400]

2:40

4pBB6. Bubble evolution in acoustic droplet vaporization. Adnan Qamar, Z. Z. Wong (Dept. of Biomedical Eng., Univ. of Michigan, 1101 Beal Ave., Ann Arbor, MI 48109), J. Brian Fowlkes, and Joseph L. Bull (Univ. of Michigan, Ann Arbor, MI 48109, joebull@umich.edu)

We present an overview of our recent theoretical and computational work on acoustic droplet vaporization within a confined tube and compare to our experimental data. This work is motivated by a developmental gas embolotherapy technique that involves injecting superheated transvascular liquid droplets and subsequently vaporizing the droplets with ultrasound to selectively form vascular microbubbles. The theoretical model describes the rapid phase transition from a superheated dodecafluoropentane (DDFP) droplet to the vapor phase via nucleation within the DDFP droplet. The theoretical results are compared to results from our computational models and high-speed camera experiments, and close agreement between the experimental bubble evolution and the prediction of the model is noted. Even though we consider bubbles that are small compared to the tube diameter, it is demonstrated that the tube affects the bubble evolution and resulting flow fields. [This work is supported by NIH Grant R01EB006476.]

3:00—3:30 Break

Contributed Papers

3:30

4pBB7. Ultrasound-mediated tumor chemotherapy with drug-loaded phase-shift nanoemulsions. Natalya Rapoport, Kweon-Ho Nam, Douglas A. Christensen, and Anne M. Kennedy (Dept. of Bioengineering, Univ. of Utah, 72 S. Central Campus Dr., Rm. 2646, Salt Lake City, UT 84112, natasha.rapoport@utah.edu)

This paper describes droplet-to-bubble transition in block copolymer stabilized perfluoropentane nanoemulsions used for ultrasound-mediated tumor chemotherapy. Three physical factors that trigger droplet-to-bubble transition in liquid emulsions and gels have been evaluated, namely, heat, ultrasound, and injections through fine-gauge needles. Among those listed, ultrasound irradiation was found the most efficient factor. Tumor accumulation of nanodroplets after systemic injections to tumor bearing mice was confirmed by ultrasound imaging. Efficient ultrasound-triggered drug release from tumor-accumulated nanodroplets was confirmed by dramatic regression of ovarian, breast, and orthotopic pancreatic tumors treated by systemic injections of drug-loaded nanoemulsions combined with tumor-directed therapeutic ultrasound. No therapeutic effect from the nanodroplet /ultrasound combination was observed without the drug, indicating that therapeutic effect was caused by the ultrasound-enhanced chemotherapeutic action of the tumor-targeted drug, rather than the mechanical or thermal action of ultrasound itself. The mechanism of drug release in the process ultrasonically activated droplet-to-bubble conversion is discussed. [The work has been supported by the NIH R01 EB1033 grant to N.R.]

3:45

4pBB8. Passive mapping of cavitation activity for monitoring of drug delivery. Costas D. Arvanitis, Miklos Gyongy, Miriam Bazan-Peregrino, Bassel Rifai (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom), Leonard W. Seymour, and Constantin C. Coussios (Univ. of Oxford, Oxford OX3 7DQ, United Kingdom)

Acoustic cavitation has been recently shown to have tremendous potential as a mechanism for releasing and enabling improved uptake and biodistribution of therapeutic agents for the pharmacological treatment of cancer, cardiovascular, and neurodengenerative diseases. Passive cavitation mapping makes it possible to localize, characterize, and quantify cavitation activity by detecting the acoustic emissions produced by cavitating bubbles on an array of receivers, which are then back-propagated to form a cavitation-selective image with high spatio-temporal accuracy. In the present study, a 64-element diagnostic linear array was used to construct passive cavitation maps during pulsed HIFU exposure of a narrow channel transporting fluorescent dextran nanoparticles through a biocompatible porous hydrogel in the presence or absence of ultrasound contrast agents. Spectral analysis of the resulting acoustic emissions confirmed that different cavitation regimes were successfully instigated over the HIFU parameter range that was investigated. Superposition of cavitation selective images over the treatment course was used to construct maps of cavitation activity and dose as a function of location for treatment assessment and was found to correlate well with dextran extravasation. It is concluded that passive mapping of cavitation activity shows great promise for assessment and real-time monitoring of cavitation based drug delivery.

4:00

4pBB9. Combined ultrasonic and video-microscopic characterization of ultrasound contrast agents in a flow phantom. Parag V. Chitnis, Paul Lee, Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., 156 William St., 9th Fl., New York, NY 10038), Paul A. Dayton (UNC-NCSU, Chapel Hill, NC 27599), and Jeffrey A. Ketterling (Riverside Res. Inst., New York, NY 10038)

A high-frequency ultrasound pulse-echo system has been combined with a video microscope system for characterizing the response of two types of ultrasound contrast agents (UCAs). A focused-flow phantom was constructed using a glass micropipette (inner diameter 30 μm) mounted vertically in a test tank. The UCA solution was pumped using a syringe pump at a constant flow rate of 40 μl/min. A high-speed CCD camera operating at 10 000 frames/s with an exposure time of 4 μs was used to capture backlit images of individual UCAs. The camera was equipped with a 100× water-immersion objective with a field-of-view of 70×70 μm. A 40-MHz, focused transducer was aligned at the micropipette and used in pulse-echo mode to sonicate individual UCAs and receive the corresponding radio-frequency backscatter signals. The UCAs were subjected to ultrasound at varying incident-pressure levels (0.8–3 MPa) and pulse durations (10, 15, and 20 cycles). Echo signals from individual UCAs were windowed and spectra were calculated. The magnitude of the fundamental (40-MHz) backscatter was observed to increase monotonically with UCA diameter. Correlation between the magnitude of the sub-harmonic (20-MHz) response of UCAs and their size was investigated. [Work supported by NIH EB006372.]

4:15

4pBB10. Dynamic response of bubbles within a compliant tube. Neo Jang, Aaron Zakrzewski, Christopher Jensen, Robert Halm (Dept. of Mech. Eng., Univ. of Rochester, Rochester, NY 14627), and Sheryl M. Gracewski (Univ. of Rochester, Rochester, NY 14627, grace@me.rochester.edu)

The dynamic response of bubbles in a liquid that are partially constrained by a surrounding tube or channel is important in a variety of fields, including diagnostic and therapeutic biomedical ultrasound and for microfluidic devices. In this study, numerical simulations, lumped parameter models, and experiments are used to investigate the effects of a surrounding tube on a bubble's response to acoustic excitation. In particular, a coupled boundary element and finite element model and COMSOL MULTIPHYSICS models have been developed and used to investigate the nonlinear interactions of this three-phase system. Simulation results were compared to experimental measurements obtained using a scaled balloon model. The effects of tube parameters and bubble interactions on a bubble's natural frequency important for proposed clinical applications of ultrasound are investigated. Resonance frequencies agree well with one-dimensional lumped parameter model predictions for a bubble well within a rigid tube, but deviate for a bubble near the tube end. Simulations also predict bubble translation along the tube axis and the aspherical oscillations and induced tube stresses at higher amplitudes. [Work supported by NIH and NSF CMMI.]

4:30

4pBB11. Using optically activated nanoparticles to promote controlled lesion formation from high-intensity focused ultrasound exposures. James R. McLaughlan (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jmcl@bu.edu), Todd W. Murray (Univ. of Colorado, Boulder, Boulder, CO 80309), and Ronald A. Roy (Boston Univ., Boston, MA 02215)

The absorption of laser light in tissue can be enhanced through the use of gold nanoparticles. This enhancement can improve the signal-to-noise ratio of the thermoelastic emissions used for photoacoustic tomography (PAT) and photoacoustic microscopy (PAM). The ability to functionalize nanoparticles allows them to be used for the selective targeting and destruction of cancer cells through the formation of vapor bubbles. However, the laser fluence required to generate vapor bubbles from nanoparticles typically exceeds the maximum permissible exposure for *in-vivo* applications. In a previous study, it was found that the combination of laser light with high-intensity focused ultrasound (HIFU) significantly reduced the fluence and pressure thresholds for bubble formation. The presence of bubbles, or specifically inertial cavitation, in HIFU exposures is believed to locally enhance the heating, resulting in effective tissue ablation at lower HIFU intensities. Nanoparticle-doped tissue phantoms (polyacrylamide gels containing bovine serum albumin) were exposed to a continuous wave (5–30-s) 1.1-MHz HIFU field with and without pulsed laser light (532 nm, 1–10 Hz) to assess the potential for HIFU lesion enhancement from the controlled and repeatable generation of inertial cavitation. [Work supported by the Gordon Center for Subsurface Sensing and Imaging Systems, NSF ERC Award No. EEC-9986821.]

4:45

4pBB12. Effects of vaporized phase-shift nanoemulsion on high-intensity focused ultrasound-mediated heating and lesion formation in gel phantom. Peng Zhang and Tyrone Porter (Dept. of Mech. Eng., Boston Univ., 110 Cummington St, Boston, MA 02215, tmp@bu.edu)

In this study, we demonstrate the feasibility of using a phase-shift nanoemulsion (PSNE) as nuclei for bubble-enhanced heating and lesion formation during high-intensity focused ultrasound (HIFU) thermal ablation. In our experiments, different concentrations of PSNE (5%–15%) uniformly distributed throughout gel phantoms made from acrylamide and albumin were sonicated with a 1.1 MHz HIFU transducer. The PNSE at the transducer focus were first vaporized into bubbles and then acoustically driven with continuous wave HIFU at different amplitudes. Bubble-enhanced heating resulted in albumin denaturation and lesion formation. HIFU-mediated heating was conducted with and without vaporized PSNE, and the lesion size and location relative to the transducer focus were measured after HIFU exposure. In the presence of vaporized PSNE, lesions were formed more rapidly and reached a larger volume compared with no vaporized PSNE. We also noted that lesions formed due to bubble-enhanced heating were more often prefocal due to backscattering from the bubble cloud. Our results show that these effects were more pronounced at higher PSNE concentrations. Finally, a comparison between the effects of pulsed and cw exposure on cavitation activity and lesion shape was investigated.

5:00

4pBB13. Observations of microbubble translation near vessel walls. Hong Chen, Wayne Kreider, Andrew A. Brayman, Michael R. Bailey, and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

To exploit the potential clinical applications of ultrasound contrast agents, it is important to understand the basic physics of bubble-vessel interactions. In this work, high-speed micrography was used to investigate

the dynamics of microbubbles inside vessels from *ex vivo* rat mesentery under the exposure of a single ultrasound pulse. The study included arterioles and venules. The ultrasound pulses were about 2 μs long with a center frequency of 1 MHz and peak negative pressures between 1 and 7 MPa. High-speed photographs reveal that interactions between microbubbles and vessels caused bubbles to move away from nearby vessel walls. The motion depended upon the standoff distance between the bubble and the nearest vessel wall, where a normalized standoff γ can be defined as the ratio of the standoff distance to the maximum bubble radius. For $\gamma < 1$, obvious bubble movement away from the nearby vessel wall was observed during collapse. At $\gamma = 0.3$, a bubble inside a venule was observed to translate over 16 μm during 1.5 μs, achieving velocities over 10 m/s. Bubble translation away from the vessel wall may be caused by the rebound of the vessel wall that was distended during bubble expansion. [Work supported by NIH Grants EB00350, AR053652, DK43881, and DK070618.]

5:15

4pBB14. Bubble-boundary interactions relevant to medical ultrasound. Wayne Kreider, Hong Chen, Michael R. Bailey, Andrew A. Brayman, and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@u.washington.edu)

Because bubble-vessel interactions have been implicated in vascular damage associated with medical ultrasound, such interactions remain a topic of interest. Recent photographs have shown that bubbles inertially oscillating in blood vessels tend to cause a vessel invagination (displacement toward the lumen) that exceeds the corresponding distention. To gain an understanding of such interactions, a Bernoulli-type equation was derived for conditions in which flow near a bubble is confined by a nearby planar boundary. This formulation assumes incompressible, inviscid flow within a locally cylindrical geometry between the bubble and boundary. Using spherical, radial bubble dynamics inferred from the aforementioned photographs, pressures generated by the bubble at the boundary were calculated. The assumed sphericity was based on observations of bubble collapses and jetting behaviors. In addition, the form of the equation was compared to that for unconfined flow around a spherical bubble. The main conclusion is that tensile stresses corresponding to vessel invagination will be asymmetrically larger than compressive stresses associated with vessel distention as the amplitude of oscillation increases and the distance between the bubble and boundary decreases. As such, these tensile stresses may be an important mechanism for vascular bioeffects. [Work supported by NIH Grants EB00350, DK43881, and DK070618.]

5:30

4pBB15. PiggyBac transposon-based gene delivery with cationic ultrasound contrast agents into SCID and CD-1 mice. Pavlos Anastasiadis (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, pavlos@hawaii.edu), Terry O. Matsunaga (Univ. of Arizona, Tucson, AZ 85724), Stefan Moisyadi, and John S. Allen (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

Efficient integration of a functional gene, along with suitable transcriptional regulatory elements, poses a fundamental difficulty for the application of successful gene therapy. Though inducible gene expression after somatic cell gene transfer has been achieved by employing viral vectors, issues associated with cargo capacity, host immune response, the risk of insertional mutagenesis, and the requirement for separate viruses are limiting factors. Several researchers have used transposon-based approaches for gene delivery, and the piggyBac transposon system has recently been shown to be effective in human cell lines and for the transgenesis in mice. The use of engineering helper-independent plasmids with the mouse codon-biased piggyBac transposon (mPB) gene in combination with cationic ultrasound contrast agents (UCAs) is investigated for a range of insonification parameters. The plasmid pmGENIE-3 deactivates the mPB gene after insertion of the transposon, eliminating potentially negative effects which may occur from the persistence of an active piggyBac gene post-transposition.

5:45

4pBB16. Quantification of endothelial cell transfection as a function of permeability with targeted ultrasound contrast agents. Pavlos Anastasiadis (pavlos@hawaii.edu) and John S. Allen (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822)

Gene and drug delivery have shown to be enhanced by the localized destruction of ultrasound contrast agents (UCAs). Moreover, deoxyribonucleic acid (DNA) is rapidly degraded by serum DNases following its injection into the bloodstream, and several studies have shown that DNA may be protected from such DNases if it is conjugated onto the shell of UCAs. The optimal acoustic parameters for transfection and delivery are not well understood. Likewise, the real-time monitoring of the related cell membrane permeability changes has only been recently attempted for these applications. This study evaluates the endothelial cell-to-cell and cell-to-substrate gaps with the electric cell-substrate impedance sensing system (ECIS, Applied BioPhysics, Troy, NY). ECIS can detect in real-time the nanometer order changes of cell-to-cell and cell-to-substrate distances separately. Targeted UCAs were conjugated with plasmid vectors and subject to ultrasound insonification, while continuous measurements of the electrical resistance across the endothelial monolayers were conducted in real-time for the determination of membrane permeability changes. The achieved transfection rate is investigated over a range of acoustic parameters. [This work was supported by the National Institutes of Health Grants NIH 2 P20 RR016453-05A1 and NIH 2 G12 RR0030161-21.]

Session 4pEA

Engineering Acoustics and Signal Processing in Acoustics: Directional Microphone Arrays Signal Processing

James E. West, Chair
Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218

Chair's Introduction—1:00

Invited Papers

1:05

4pEA1. Optimal spherical array modal beamformer in the presence of array errors. Haohai Sun (Dept. of Electrons and Telecommunications, Norwegian Univ. of Sci. and Tech., 7491 Trondheim, Norway, haohai.sun@iet.ntnu.no), Shefeng Yan (Chinese Acad. of Sci., 100190 Beijing, China), and U. Peter Svensson (Norwegian Univ. of Sci. and Tech., 7491 Trondheim, Norway)

Spherical array modal beamforming has become an attractive technique, since three-dimensional beampatten synthesis is more flexible than other array geometries, and the modal beamforming can be performed using the elegant spherical harmonics framework. However, the performance of a modal beamformer in practical situations is known to degrade in the presence of array errors caused by non-perfect spatial sampling, sensor sensitivity and phase variations, and sensor self-noise. Although the white noise gain constraint has been widely used to control the robustness of modal beamformers, it is not clear how to properly choose the constrained parameter set based on the known range of errors. In this paper, a worst-case performance optimization approach is formulated for the spherical array modal beamformer to improve its robustness against the above mentioned errors occurring in practice; thus the optimum performance can be obtained based on the known maximum level of errors. The robust optimal modal beamforming problem is reformulated as a convex optimization within the spherical harmonics framework, which can be efficiently solved by using second order cone programming.

1:25

4pEA2. Analysis of the high-frequency extension for spherical eigenbeamforming microphone arrays. Jens Meyer and Gary W. Elko (mh acoustics LLC, 25A Summit Ave., Summit, NJ 07901)

The em32 Eigenmike® spherical microphone array is constructed with 32 omnidirectional pressure microphones mounted on the surface of a 8.4-cm rigid spherical baffle. Beamforming is accomplished by the spherical harmonic decomposition eigenbeamformer and modal-beamformer concept. Due to the array size, geometry, and average element spacing, this approach yields fully controllable beampattern control up to approximately 8 kHz. Above this frequency, spatial/modal aliasing begins to interfere with beampattern control. However, professional audio applications require an operating bandwidth to much higher frequencies. One solution is to use a single microphone capsule that is closest to the main axis of the beampattern direction at higher frequencies. It will be shown that there are some problems to this solution. As an alternative implementation it is shown that using a small subset of microphone capsules at high frequencies allows one to smoothly transition the beampattern from eigenbeamformer processing to higher frequencies.

1:45

4pEA3. Audio visual scene analysis using spherical arrays and cameras. Adam O'Donovan, Ramani Duraiswami, Dmitry Zotkin, and Nail Gumerov (PIRL Lab., Univ. of Maryland, 3366 AVW Bldg., College Park, MD 20742, adamod@gmail.com)

While audition and vision are used together by living beings to make sense of the world, the observation of the world using machines in applications such as surveillance and robotics has proceeded largely separately. We describe the use of spherical microphone arrays as "audio cameras" and spherical array of video cameras as a tool to perform multi-modal scene analysis that attempts to answer questions such as "Who?,", "What?," "Where?," and "Why?." Signal processing algorithms to identify the number of people and their identities and to isolate and dereverberate their speech using multi-modal processing will be described. The use of graphics processor based signal processing allows for real-time implementation of these algorithms. [Work supported by ONR.]

2:05

4pEA4. A comparison of sound localization methods in a room environment using a spherical microphone array. Colin Barnhill and James E. West (Dept. ECE, Johns Hopkins Univ., Baltimore, MD 21218, cb@jhu.edu)

Enclosures are one of the most challenging environments for source localization, especially for teleconferencing where the main goal is to enhance desired speech signals while suppressing noise. This can be accomplished using beamforming but the speech source locations must be known. Three passive methods will be compared for room environment localization: generalized cross-correlation (GCC), spherical harmonic-based multiple signal classification (MUSIC), and spatial power signal characteristics and gradients (SPSGs). SPSG is a new localization algorithm designed for room environments based on wideband third order beamforming and power

signal characteristics, differences, and gradients. MUSIC is a subspace method that relies on an eigen-decomposition of the beam correlation matrix and GCC is a time delay estimation algorithm based on microphone signal correlation. MUSIC and GCC are methods that perform well in their optimal environments but suffer in noisy and reverberant conditions. Results will be shown for multiple source situations with reverberation and noisy interferers.

Contributed Papers

2:25

4pEA5. Microphone array sensor node for distributed wireless acoustic beamforming. Alaa A. Abdeen and Laura E. Ray (Thayer School of Eng., Dartmouth College, 8000 Cummings Hall, Hanover, NH 03755)

The presentation will illustrate the use of field programmable gate arrays (FPGAs) in microphone array digital signal processing for wireless acoustic beamforming in source localization. We have developed a prototype of a node composed of ten microphone channels interfaced with a Xilinx-based FPGA board. The methodology involves basic signal conditioning and delay-and-sum beamforming programmed by VHDL. The prototype results illustrate significant increase in the sensitivity to acoustic sources in the direction of a signal, while rejecting noise from other directions. The sensitivity is also enhanced in relationship to the number of channels. This node prototype is a building block for a wireless sensor network, capable of distributed beamforming at about 10-kHz sample frequencies. Each node is designed to maximize throughput so that broadband acoustic sources are accommodated. We are now performing parametric studies to optimize the node design and microphone array geometry to provide maximum bandwidth, dynamic range, and adequate special resolution, while preserving the ability to resolve time delays between the microphones.

2:40

4pEA6. MEMS (microelectromechanical systems) microphone array on a chip. Joshua S. Krause and Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, joshua.krause@tufts.edu)

The design, fabrication, and characterization of a surface micromachined, front-vented, 64 channel (8×8), capacitively sensed microphone array-on-a-chip are described. The element pitch is 1.3-mm center-to-center and is targeted at resolving the high-wavenumber components of the pressure spectra underneath the turbulent boundary layer. Care is taken to minimize package topology to reduce flow generated self-noise. The array was fabricated using the MEMSCAP PolyMUMPs® process, a three layer polysilicon surface micromachining process, with the addition of a Parylene-C coating in post-processing. An acoustic lumped element model, including mechanical components, environmental loading, fluid damping, and other acoustic elements, is detailed. Laboratory calibration indicates a sensitivity of 1 mV/Pa for each microphone over a 200–40 000 Hz band. A strong resonance occurs at 280 kHz in close accordance with modeled results. Spatial mapping of the array reveals minimal electrical and physical cross-talk between elements. Preliminary element-to-element sensitivity comparisons exhibit a standard deviation of 13%, 12%, and 25% at 250 Hz, 500 Hz, and 1 kHz. Phase matching showed a 9, 10, and 15 deg standard deviation difference at the same frequencies. A more in depth analysis of acoustic sensitivity is ongoing as well as element-to-element variability.

2:55

4pEA7. Ultrasonic calibration of MEMS (microelectromechanical systems) microphone arrays. Katherine E. Knisely, Victor Singh, Karl Grosh, David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109, kknisely@umich.edu), and Serdar H. Yonak (Toyota Motor Eng. and Manufacturing, North America, Inc., Ann Arbor, MI 48105)

Ultrasonic microphone arrays have applications in imaging, scale model testing, and sound source localization. For these applications MEMS ultrasonic microphones are emerging as a natural choice because of their small size, low cost, and robustness to manufacturing treatments and environmental conditions. In this talk an array is constructed using commercially available Knowles SiSonic[TM] capacitive MEMS microphones. The absolute sensitivity and phase calibrations of the array elements in the 1–60-kHz frequency range are obtained through the use of replacement calibration and time-frequency analysis. For these results, a modulation in sensitivity and phase measurements for frequency ranges over 15 kHz is observed under certain conditions. The origin of this modulation is discussed as well as various techniques for its removal. Relative element calibrations are crucial for array performance and are obtained through a variety of sound source configurations, which vary in size and location relative to the array.

3:10

4pEA8. Characterization of a high-frequency pressure-field calibration method. Dylan Alexander (Mech. and Aero. Eng., Univ. of FL, 231 MAE-A, Gainesville, FL 32611, fhdpa86@ufl.edu), Casey Barnard, Benjamin A. Griffin, and Mark Sheplak (Univ. of Florida, Gainesville, FL 32611)

A method for the direct measurement of the pressure sensitivity of a microelectromechanical system (MEMS) microphone is developed for frequencies from 10 to 100 kHz. The use of a simultaneous pressure field measurement reduces the errors in calculating the pressure sensitivity from free field measurements, and it allows calibration of non-reciprocal and non-capacitive transduction schemes. An ionophone is used as a calibration source in an anechoic chamber. An aluminum plate is mounted in the anechoic chamber and contains a linear array of pressure field condenser microphones. Treatment of the plate edge is used to minimize diffraction within the pressure field. The microphone data are used to characterize the directivity of the ionophone and localize sources of acoustic diffraction that could impact future MEMS microphone magnitude and phase calibrations.

3:25

4pEA9. Experimental study of synthetic aperture microphone array on rotor blade noise. Jaehyung Lee, Wook Rhee, and Jong-Soo Choi (Dept. of Aerosp. Eng., Chungnam Nat'l. Univ., Daejeon 305-764, Republic of Korea aerojhl@cnu.ac.kr)

Recent development of signal processing technologies has encouraged the aero-acousticians to implement the state-of-the-art sensors and data acquisition devices. They sometimes cost too much; however, especially for the academic arenas, the efforts of reducing the expense of integrating a system and increasing performance of phased array have initiated the idea of designing the synthetic aperture microphone array. The application of beamforming technique in localizing rotor blade noise and the use of synthetic aperture array in wind tunnel testing are presented. To apply the synthetic aperture beamforming on rotating objects, three steps should be considered. The first step is to restore the original acoustic waveform radiated from a rotating sound source. The second step is to synthesize independent array measurement data by correcting phase information of the received acoustic signals. Finally, the beamformer calculates the power on each point of grid in the space. The paper describes the details of deploying the synthetic aperture microphone array and the data reduction methods associated with acoustic measurement in wind tunnel. The comparisons of various patterns of synthetic aperture array are made. The beampower maps for rotor blade noise of various flight conditions are presented.

3:40

4pEA10. Comparison of single-frequency monopulse techniques that mimic the results of multiple-frequency, single-aperture interferometry. Daniel S. Brogan and Kent A. Chamberlin (Dept. of Elec. and Comput. Eng., Univ. of New Hampshire, 33 Academic Way, Durham, NH 03824-2619, daniel.brogan@unh.edu)

The capability to resolve targets from each other and from noise is desirable in all target detection and tracking systems. In a previous presentation [G. R. Cutter, Jr. and D. A. Demer, J. Acoust. Soc. Am. **126**, 2232 (2009)] it was shown that single-frequency, multiple-aperture interferometry could be used to resolve multiple targets that could not be resolved using single-frequency, single-aperture interferometry. This was accomplished by using the phase differences from several pairs of subarrays with different

subarray spacing to mimic the operation of a multiple-frequency, single-aperture interferometric system. The differential-phase technique is one type of phase-comparison monopulse used in target bearing estimation. The current research compares the results of single-frequency, multiple-aperture interferometry to those of other single-frequency phase-comparison and amplitude-comparison monopulse techniques by using multiple receive subarrays and/or varying the receive arrays' elements' weights.

3:55

4pEA11. Non-uniform array synthesis concept and theory. Dehua Huang (NUWC, Newport, RI 02841)

Uniform or symmetric array is a common practice for most of the sonar array design, where the same performance transducer elements are used for numerical simulation and manufacture engineering conveniences. However, non-uniform array designs show their advantages for special array applications. This paper will theoretically address the non-uniform array synthesis concept and theory. A few numerical examples will also be presented for element hybrid and cluster special arrays. [The work is supported by the U.S. Navy.

4:10

4pEA12. Comprehensive development of the theory of nonuniformly spaced arrays. Jenny Y. Au and Charles Thompson (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854)

In this work, we will present the work of comprehensive development of the theory of nonuniformly spaced arrays. It will be shown that the position may be determined in terms of the Fourier transform of the objective function. Special consideration will be given to the objective function of G. J. van der Maas in 1954. These results will be compared with that presented by Wallace in 1982.

THURSDAY AFTERNOON, 22 APRIL 2010 HARBORSIDE B, 2:00 TO 3:25 P.M.

Session 4pID

Interdisciplinary: Science and Human Rights Coalition: The Interface Between the Human Rights and Scientific Communities

Edward J. Walsh, Chair

Boys Town National Research Hospital, 555 N. 30th St., Omaha, NE 68132

Chair's Introduction—2:00

Invited Papers

2:05

4pID1. The Science and Human Rights Coalition of the AAAS (American Association for the Advancement of Science): The interface between the human rights and scientific communities. Jessica Wyndham (AAAS, Sci. and Human Rights Coalition, 1200 Washington Ave., NW, Washington, DC 20005)

In July 2005, AAAS held a 2-day conference of scientific associations and human rights organizations to explore ways in which the scientific community could become more directly engaged in human rights. Born of a conference recommendation, the AAAS Science and Human Rights Coalition was officially launched in January 2009. Forty-two scientific membership organizations have since joined the Coalition which aims to build bridges and opportunities for collaboration within the scientific community and between the scientific and the human rights communities. Introducing the Coalition, Ms. Wyndham will answer four questions: What does the Coalition aim to accomplish? How will it do this? Who can join? How does it operate? One year young, the growing number of Coalition members reflects the commitment of scientific organizations to addressing human rights. This, however, is just the beginning. Building on the achievements to-date and lessons learnt throughout the course of the first year, Ms. Wyndham will outline the Coalition's plans for making a real and meaningful contribution to the realization of human rights.

2:25

4pID2. Aspects of participation in the Science and Human Rights Coalition of the AAAS (American Association for the Advancement of Science): Working groups and internal outreach. Clinton W. Anderson (American Psychol. Assoc., 750 First St., NE, Washington, DC 20002)

The goal of Dr. Anderson's presentation is to inform ASA members about two practical aspects of participation in the AAAS Science and Human Rights Coalition: Participation in the working groups that are at the core of the Coalition's activities and outreach by coalition member representatives internally within their organizations. The speaker will briefly introduce the missions and modes of operation of the five working groups that are currently active within the Coalition and focus more extensively on the operations and objectives of the Service to the Scientific Community Working Group. Based on the action plan of the Service to the Scientific Community Working Group and his own experience as an organization member representative for the American Psychological Association, the speaker will to explore ways in which ASA members might work to increase awareness and appreciation of human rights issues and alternatives within the ASA; to increase capacity of the ASA to address human rights issues through a variety of means, including the application of acoustical tools and techniques; and to increase engagement of ASA in human rights activities.

2:45

4pID3. The Acoustical Society of America and the Human Rights Coalition: An introduction. Edward J. Walsh (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68132)

Technological developments in recent years have significantly improved the capacity of human rights advocates to monitor and document abuses of authority, including violations of human rights at home and in remote isolated regions of the world. Recognizing the need for a platform from which scientists and scientific Societies might speak with a common voice to promote human rights protect the professional and personal interests of threatened colleagues, cooperate with human rights organizations, and press for the practical implementation of the right of all persons to the benefits of scientific progress and its applications, the AAAS Science and Human Rights Coalition was launched in 2008. Operating through the Panel on Public Policy, the ASA accepted an invitation to participate as an affiliate member of the Coalition, and the purpose of the Special Session is to introduce the membership at large to the principles and goals of the organization and to offer insight into the mechanism of operation and cooperation through which an effective interface between the scientific and human rights communities might be developed.

3:05

4pID4. The role of science in the human rights movement. Leonard Rubenstein (Ctr. for Public Health and Human Rights, Johns Hopkins School of Public Health, 615 N. Wolfe St., Baltimore, MD 21205)

"Scientists have contributed valuably in making human rights a reality for all. Notable scientific contributions to human rights include the forensic exhumation of mass graves, the use of DNA evidence to identify victims of mass killings, the introduction of information management techniques to illuminate large-scale human rights violations, and, more recently, the use of satellite imagery to document the destruction of communities in remote locations." [Rubenstein and Younis 2008)]. As a human rights lawyer and a consumer of science, Mr. Rubenstein will discuss the contributions of science to human rights challenges, and the importance of the scientific community becoming a constituency for human rights. He will address head-on some of the greatest misconceptions about human rights work, including the charge that it is too "political" as well as steps needed to assure that scientific work undertaken with human rights organizations and is conducted consistent with scientific traditions of impartiality and independent inquiry. Finally, he will demonstrate how the AAAS Science and Human Rights Coalition can be used as a vehicle for the application of scientific skills, to reinforce the commitment, and to amplify the voices of scientists working toward the realization of human rights for all.

THURSDAY AFTERNOON, 22 APRIL 2010 ESSEX A/B/C, 3:15 TO 5:30 P.M.

Session 4pMU

Musical Acoustics: General Topics in Musical Acoustics

Thomas R. Moore, Chair
Rollins College, Dept. of Physics, Winter Park, FL 32789

Contributed Papers

3:15

4pMU1. Novel exploration of the vibration of the Caribbean steelpan in response to authentic excitation. Patrick O'Malley, Joseph F. Vignola, and John A. Judge (Dept. of Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., Washington, DC 20064)

A novel conformal scanning laser vibrometer was used to determine the vibrational modes of a C-lead tenor steelpan. The measurements provide a rich mechanical snapshot of the artistry involved in the marriage of metallurgy and music theory that goes into the creation of the instrument. The data represent the surface-normal motion of the entire face of the instrument in response to an impulsive excitation intended to mimic the strike of a mallet. A traditional scanning laser Doppler vibrometer is limited to measurement of planar geometries and therefore is of limited utility on the highly contoured surface of the steelpan. A description of the novel measurement system is presented, followed by the surface velocity data and a summary of response shapes and the frequencies at which those responses occur. The data show that individual note areas respond when adjacent or non-adjacent musically related notes are struck and clearly illustrate the complex vibration of the steelpan and the coupling between notes that produce the rich distinctive nature of the steelpan sound.

3:30

4pMU2. Vibrational characteristics of Balinese gamelan metallophones. Molly E. Jones (Dept. Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48109, jonesmo@umich.edu), Kent L. Gee, and Jeremy Grimshaw (Brigham Young Univ., Provo, UT 84602)

A study of the 16 metallophones from a Balinese gamelan has been conducted. Acoustical recordings of metallophone bars being struck were used to examine ratios of overtone frequencies to the fundamental. Results showed large variability in the number and ratios of overtones present. Scanning laser Doppler vibrometry measurements made on several bars also revealed great variability in mode shapes present. The distribution of prominent overtones and their modal shapes do not appear to match those of Western metallophones. Notably, the overall gamelan metallophone characteristics are quite dissimilar to the glockenspiel, which disagrees with previous studies. [This study was supported by the National Science Foundation through its Research Experiences for Undergraduates program.]

3:45

4pMU3. An introduction to the acoustics of playing the horn. David J Zartman (Graduate Program in Acoust., Penn State Univ., 405 EES Bldg., State College, PA, 16801)

In the art of music, a good performer can make a low-quality instrument sound passable, while a bad performer will not play well even on the best instrument in existence. Rather than merely focusing on the properties of the instrument, it is useful to provide the musician with some of the acoustical principles involved with playing the horn well. At the same time, this provides a medium to enable the acoustician to understand some of the intricacies a musician must balance well to create a pleasing sound.

4:00

4pMU4. Novel computer aided design of labial flue pipes. Brian Moss (CSRC/OFSRC, Univ. of Limerick, Limerick, Ireland), Elfed Lewis, Gabriel Leen, Kort Bremer, and Andrew Niven (Univ. of Limerick, Limerick, Ireland)

A labial flue pipe is a well known tone generator, which is familiar and easily recognizable as the organ pipes seen in many concert halls and churches. However, the design and understanding of the sounding mechanism of such pipes are fraught with difficulty. Traditionally labial pipes are constructed from age-old lookup tables that are closely guarded intellectual secrets. This paper discusses a novel computer program that facilitates the design and construction of such labial flue pipes. The computer program allows almost all aspects of the labial flue pipe design to be varied, the resultant frequency is generated, and in addition the Ising efficiency number is provided. Furthermore, a discussion is included related to the fact that even though an Ising number greater than 3 indicates that a pipe is overblown, the fundamental tone is still predominant. A comparison will also be made between a CFD simulation of the labial flue pipe jet mechanism and smoke trail plots typical of such analysis.

4:15

4pMU5. Acoustical studies on conch shells. M.G. Prasad and B. Rajavel (Dept. of Mech. Eng., Noise and Vib. Control Lab., Stevens Inst. of Technol., Hoboken, NJ 07030, mprasad@stevens.edu)

Sound from blowing a conch shell is known for its tonal quality. Its use in many cultures is known. In Hindu culture, conch shell is blown on several auspicious occasions such as festivals, worship, and rituals. A conch shell can be modeled as a horn in its acoustical behavior. This work presents an acoustical study in terms of the spectral characteristics of the shell. The study also presents the influence of parameters such as placing hand in the mouth of the conch shell and the size of conch shell on the spectral characteristics.

4:30

4pMU6. On the role of acoustics in the Vedic Hindu tradition and philosophy. M. G. Prasad and B. Rajavel (Dept. of Mech. Eng., Noise and Vib. Control Lab., Stevens Inst. of Technol., Hoboken, NJ 07030, mprasad@stevens.edu)

Acoustics has received a very high importance in the Vedic Hindu tradition and philosophy of ancient India. The Vedic literature which refers to both the Vedas and the subsequent literature based on Vedas emphasize that the roles of sound both as source signal and as hearing are important. This work presents the various ways such as chants, Sanskrit language, vocal, and instrumental music in which acoustics has played a major role.

4:45

4pMU7. The shape of musical sound: Real-time visualizations of expressiveness in music performance. Gang Ren (Dept. of Elec. and Comput. Eng., Edmund A. Hajim School of Eng. and Appl. Sci., Univ. of Rochester, Rochester, NY 14627), Justin Lundberg, Mark F. Bocko, and Dave Headlam (Univ. of Rochester, Rochester, NY)

Despite the complex multi-dimensional nature of musical expression in the final analysis, musical expression is conveyed by sound. Therefore the expressiveness of music must be present in the sound and therefore should be observable as fundamental and emergent features of the sonic signal. To gain insight into this feature space, a real-time visualization tool has been developed. The fundamental physical features—pitch, dynamic level, and timbre (as represented by spectral energy distribution)—are extracted from the audio signal and displayed versus time in a real-time animation. Emergent properties of the sound, such as musical attacks and releases, the dynamic shaping of musical lines, timing of note placements, and the subtle modulation of the tone, loudness, and timbre can be inferred from the fundamental feature set and presented to the user visually. This visualization tool provides a stimulating music performance-learning environment to help musicians achieve their artistic goals more effectively. Such visualizations and interactions with musical sound also will promote the semantic understanding of an expressive musical language.

5:00

4pMU8. Robust audio watermarking scheme based on spectral modeling synthesis. Pranab Dhar (School of Comput. Eng. & Information Technol., Univ. of Ulsan, Korea, pranab_cse@yahoo.com), Cheol Hong Kim (Chonnam Natl. Univ., Chonnam, Korea), and Jongmyon Kim (Univ. of Ulsan, Korea)

This paper proposes a new watermarking scheme based on spectral modeling synthesis for copyright protection of digital contents. In our proposed scheme, a short time Fourier transform (STFT) is applied to the original signal. Prominent spectral peaks of each frame are then detected and represented by sinusoidal parameters. The residual component is computed by removing all the prominent peaks from the spectrum, transforming the spectrum back to the time domain using inverse fast Fourier transform, and overlap-adding the frames in time. In addition, a peak tracking unit keeps the sinusoidal parameters and forms peak trajectories. Watermarks are then embedded into the most prominent peak of each frame in the peak trajectories. Watermarked signal is computed by the sinusoidal synthesis process and then adding the sinusoids with the residual signal. Watermarks are detected by using the inverse operation of watermark embedding process. Simulation results indicate that the imperceptible watermarks embedded into four different kinds of music sounds using the proposed scheme are robust against several attacks such as noise addition, cropping, re-sampling, re-quantization, and MP3 compression, showing similarity values ranging from 0.8 to 0.9. [Work supported by KOSEF, R01-2008-000-20493-0.]

5:15

4pMU9. Effects of tempo and other musical features on stress responses to college students. Wei-Chun Wang (Dept. of Digital Lit. and Arts, St. John's Univ., 499, Sec. 4, Tam King Rd., Tamsui, Taipei, Taiwan, vgnwang@hotmail.com)

It is believed that music has the power to soften emotions and alleviate pains. This study is aimed to explore the effects of tempo and other musical elements on stress-associated responses of college students when they listen to music. The effects of musical experience, preference, and awareness of music content on stress ratings are also investigated. College students were asked to listen to eight recorded musical excerpts and fill in a questionnaire to rate their relaxation and anxiety levels in a Likert five-point scale and to specify the musical elements. With a variety of tempo being considered, the musical excerpts were chosen from different types of music, including western classical music, Chinese pop music, Chinese traditional music, rock music, jazz blue, and new age music. Preliminary results showed that there was a correlation between relaxation and tempo, and the degree of personal music preference promoted stress reduction. It suggests that music used to enhance relaxation should match a person's musical taste.

Session 4pPA

Physical Acoustics: Nonlinear Acoustics

Martin D. Verweij, Cochair
Delft Univ. of Technology, Delft, The Netherlands

Robin O. Cleveland, Cochair
Boston Univ., Dept. of Aerospace and Mechanical Engineering, 110 Cummington St., Boston, MA 02215

Contributed Papers

1:00

4pPA1. Miniature thermoacoustic engine: Experiments and modeling. Konstantin I. Matveev and Sungmin Jung (MME School, WSU, Pullman, WA 99164-2920, matveev@wsu.edu)

A miniature standing-wave thermoacoustic engine was constructed and tested with four quarter-wavelength resonators having lengths in the range of 57–124 mm. 80-ppi reticulated vitreous carbon (RVC) foam was applied as a stack, and an atmospheric air served as a working fluid. The critical temperature difference between the stack ends, corresponding to the sound onset, was recorded in the range of 200–300 °C. A thermoacoustic model was developed to predict the critical temperature difference across the stack. The model allows for user-defined thermoacoustic functions for the stack. Results obtained with the capillary-based theory for tortuous media and thermoacoustic functions previously measured for RVC demonstrate a trend qualitatively consistent with our data, while the quantitative agreement is lacking. Modeling results obtained with thermoacoustic functions for a parallel-plate stack can approximate experimental data, if the appropriate plate spacing is selected. [Work supported by the NSF Grant No. 0853171.]

1:15

4pPA2. Characterization of the heat flux through the heat exchangers of a thermoacoustic cooler. Gaëlle Poignand, Philippe Blanc-Benon, Emmanuel Jondeau (LMFA, UMR CNRS 5509, Ecole Centrale de Lyon, 69134 Ecully Cedex, France, gaelle.poignand@ec-lyon.fr), Etienne Gaviot, Lionel Camberlein, Guillaume Penelet, and Pierrick Lotton (l'Universit du Maine, 72085 Le Mans, France)

A small-scale standing wave thermoacoustic cooler with a couple of stack and heat exchangers is studied. In addition to classical instrumentation in such a device, thermal heat flux sensors specifically developed using MEMS technology equipped the heat exchangers of this refrigerator. These sensors give the temporal evolution of the heat fluxes through the hot and the cold heat exchangers. Hence, they provide a better understanding of heat transfer between the stack and the heat exchangers. In this work, the temporal evolution of the temperature along the stack and of the heat flux through the heat exchanger is measured versus the acoustic pressure. The results show that for high-pressure levels, the heat flux extracted at the cold exchanger rapidly increases until a maximum value and then stabilizes at a lower value. The origin of this limitation may come from the formation of vortices behind the stack highlighted in the work of Berson *et al.* [Heat Mass Transfer **44**, 1015–1023 (2008)]. This work is supported by ANR (project MicroThermAc NT051_42101).

1:30

4pPA3. Inertial cavitation threshold dependence on high static pressures. Kenneth, B. Bader, Joel Mobley, Jason Raymond (Natl. Ctr. for Physical Acoust., Coliseum Dr, Univ., MS 38677, kbader@olemiss.edu), and D. Felipe Gaitan (Impulse Devices, Inc., Grass Valley, CA 95945)

The creation of a bubble in a liquid is energetically favorable when the liquid is subject to a net/an overall tension less than of its saturated vapor pressure. This gain is offset by the creation of a liquid-vapor/gas interface, and only bubbles larger than some critical size will spontaneously grow. The magnitude of the acoustic pressure required to produce a cavity of the critical size is termed the cavitation threshold of the liquid. The dependence of the cavitation threshold on hydrostatic pressure has previously been reported up to 130 bars in terms of electrical power applied to the acoustic driving transducer. These measurements used a standing wave set-up in a stainless steel spherical resonator (24.1 cm outer diameter, 1.9 cm thick) with a $Q >$ 10 000 when fluid loaded. This work will extend into higher pressure regimes and will be used to extrapolate the cavitation threshold at 1 kbar.

1:45

4pPA4. Determination of bubble size distributions in an ultrasonic cavitation field. Stéphane Labouret and Jacques Frohly (Département d'Opto-Acousto-Electronique, Institut d'Electronique, Micro-électronique et Nanotechnologies (IEMN), Université de Valenciennes et du Hainaut-Cambresis, le mont Houy, F-59 313 Valenciennes, France, labouret@laposte.net)

An electromagnetic method is proposed for determining bubble size distribution in the range 2–12 m in ultrasonic cavitation fields. The method is used to investigate the effect of the continuous and pulsed irradiations and of the presence of increasing concentration of a surfactant (sodium dodecyl sulfate) on the bubble size distribution in a standing wave ultrasound field. The electromagnetic sizing technique shows considerable promise as a means of tracking changes in the bubble size distribution during insonation and determining the role of such changes in transitioning from a field that contains a few discrete bubbles to one that exhibits widespread cavitation activity.

2:00

4pPA5. Pressure, temperature, and dissolved gas dependence of dielectric breakdown in water. Jonathan Sukovich and R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

It has been shown experimentally that the optical breakdown strength of water is a pressure dependent quantity, growing with increasing pressure. The dependence of the breakdown strength on temperature and dissolved gas concentration over a larger range of pressures will be observed. Using a custom fabricated pressure vessel and high-power Nd:YAG laser, breakdown events will be generated and observed over a range of pressures from 0 to 25 kpsi. Observations of breakdown events will be made using a high-speed photodetector located behind the pressure vessel's optical windows. Dissolved gas concentration will be controlled and varied using a custom water preparation system over a range from water's vapor pressure ($\sim$20 torr) to atmospheric pressure. Temperature will be monitored using a thermocouple attached to the pressure vessel and the temperature dependence will be measured over a range from 20 to 35 °C. A comparison between current single detector methods and previous imaging methods of using breakdown to determine absolute pressure will then be made. [Work supported by Impulse Devices, Inc.]

2:15

4pPA6. A study on nonlinear ultrasonic waves in bubbly media: One and two-dimensional numerical experiments. Vanhille Christian (Escet, Univ. Rey Juan Carlos, Tulipan s/n, 28933 Mostoles Madrid, Spain, christian.vanhille@urjc.es) and Campos-Pozuelo Cleofé (Instituto de Acustica, CSIC, 28006 Madrid, Spain, ccampos@ia.cetef.csic.es)

Nonlinear ultrasonic waves are studied in bubbly media, typically liquids with gas bubbles. The acoustic behavior is modeled by a differential system which couples the acoustic pressure and the nonlinear dynamic of the bubbles. The differential equations are solved by means of a numerical model that gives the acoustic pressure and bubble volume variation as time traces (SNOW-BL code) in two configurations: one and two dimensions. Results are shown for one-dimensional progressive and standing waves as well as for two-dimensional progressive waves, with or without focus. Experiments are carried out with bubbles evenly distributed or forming clouds, for example, in the pre-focal zone. [This work is part of the research project DPI2008-01429.]

2:30—2:45 Break

2:45

4pPA7. Temporal evolution of laser-nucleated bubble clouds in an acoustic resonator. Phillip A. Anderson and R. Glynn Holt (Dept. Mech-Eng, Boston Univ., 110 Cummington St., Rm. 101, Boston, MA 02215)

Using high-speed digital imaging, the evolution of laser-nucleated bubble clouds over multiple acoustic cycles is observed. Five 7×7 custom phase gratings are employed to produce 245 beams, which are then focused into the center of a spherical resonator capable of high static and acoustic pressure. The dependence of the evolution of the bubble cloud on several system parameters is measured. Notably the resulting cloud(s) depend strongly on the phase of the laser firing, the laser energy per pulse, and the acoustic pressure. Number of bubbles, radius of individual bubbles, and effective radius of clouds will be reported as functions of time. The interplay between cloud dynamics and shock waves will also be discussed. [Work supported by Impulse Devices, Inc.]

3:00

4pPA8. Theoretical analyis of Schwinger's conjecture on sonoluminescence in relation to the boosting of sonluminescent transduction efficiency. Harvey C. Woodsum (Sonetech Corp., 10 Commerce Park North, Unit 1, Bedford, NH 03110)

In a previous paper [Spring ASA, Salt Lake (2007)], we have considered the potential viability of Schwinger's conjecture that sonoluminescent light radiation, which results from acoustically developed cavitation, derives from zero-point energy of the vacuum. In our prior paper, we considered the nonlinear interaction of zero-point electromagnetic modes in a cavitating bubble, through the Euler–Heisenberg theory for the scattering of light by light. Good agreement was found between theory and experiment with regard to both the efficiency and spectrum of the sonoluminescent radiation generated using a single acoustic frequency. We now consider a comparison of this theory with other experiments [Phys. Rev. Lett. **81**, 1961–1964 (1998)] which have demonstrated that cavitation bubbles created with acoustic waves having both first and second harmonics present, and having particular amplitude and phase arrangements, can result in sonoluminescent transduction efficiency increases of up to 300%. At this point in time, the predictive power of the current theory appears promising.

3:15

4pPA9. Nonlinear poroacoustic flow in rigid porous media. Pedro Jordan and Jim Fulford (U. S. Naval Res. Lab., Stennis Space Ctr., MS 39529)

An acoustic acceleration wave is defined as a propagating singular surface (i.e., wavefront) across which the first derivatives of the velocity, pressure, or density exhibit jumps. In this talk, the temporal evolution of the amplitude and the propagation speed of such waves are investigated in the context of nonlinear acoustic propagation in rigid porous media. By considering the exact conservation/balance equations, it is shown that there exists a critical value, the constant $\alpha^*(>0)$, of the initial jump amplitude such that the acceleration wave magnitude either goes to zero, as $t\to\infty$, or blows up, in finite time, depending on whether the initial jump amplitude is less than or greater than α^*. In addition, stability is addressed; a connection to traveling wave phenomena is noted, for which an exact traveling wave solution is obtained; and a comparison with the linearized case, i.e., the well-known damped wave equation, is also presented. Finally, the numerical solution of an idealized, nonlinear initial-boundary value problem involving sinusoidal signaling in a fluid-saturated porous slab is used to illustrate the finite-time transition from acceleration to shock wave, which occurs when the initial jump amplitude exceeds α^*. [Work supported by ONR/NRL funding.]

3:30

4pPA10. Different regimes of nonlinear pulse propagation in porous medium. Diego Turo and Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, Salford, Greater Manchester M5 4WT, United Kingdom)

High-amplitude pulse propagation in rigid porous media has been investigated numerically and experimentally. At high-sound levels, strong interactions between different spectral components of the pulse make any frequency domain models difficult to use, therefore a time domain approach has been applied for the present research. The effect of Forchheimer nonlinearity (i.e., flow resistivity growth with particle velocity amplitude) in porous media is well studied. However, much less is known about the influence of memory effects on high-amplitude pulse propagation. The aim of this work is to study their relevance for high-amplitude pulses of different durations. A numerical finite difference time domain scheme has been developed which accounts for second order convection nonlinearity, Forchheimer correction, and memory effect simultaneously. It is shown that Forchheimer nonlinearity dominates for longer duration high-amplitude pulses, while convection terms and memory effect contribution become noticeable for shorter pulses of moderate amplitude. The numerical results are confirmed in a series of experiments with different duration pulses in the amplitude range from 120 Pa to 40 kPa.

3:45

4pPA11. Directivity and frequency control of an intense underwater laser acoustic source for navy applications. Melissa Hornstein, Theodore Jones, Antonio Ting, and Michael Nicholas (Acoust. Div., U.S. Naval Res. Lab., Washington, DC 20375)

We develop an intense laser acoustic source, in which a tailored laser pulse can compress underwater at a predetermined remote location. Optical compression results in laser-induced breakdown (LIB), localized heating, and acoustic shock generation. Recent experiments include near-field acoustic source characterization using lens-focused 400-, 800-, 532-, and 1064-nm pulses of Ti:sapphire and Nd:YAG lasers. Sound pressure levels over 215-dB were achieved using a compact laser. We have demonstrated control of the shape of the LIB plasma volume, and thereby control of the acoustic frequency spectrum and acoustic source directivity. By superposition of volume elements within the LIB, the acoustic pulse duration in a given propagation direction is determined by the parallel dimension of the LIB, divided by the parallel acoustic transit time across the LIB. Thus, the shape of the LIB strongly affects the acoustic pulse duration and directivity, and aspherical LIB volumes result in strongly anisotropic acoustic sources. In our experiments, the LIB volume shape was varied by laser pulse length, energy, optical bandwidth, and focusing angle. We also studied acoustic propagation in a 30 000-gal bubbly salt water tank. [This work is supported by the U.S. Office of Naval Research.]

4:00

4pPA12. Investigation of acoustic streaming in the cochlea. Katherine Aho, Elaine Vejar, and Charles Thompson (Dept. Elec. Eng., CACT, FA 203, 1 University Ave, Lowell, MA 01854)

In this work we will analyze acoustic streaming in the cochlea. A model will be developed to examine the steady flow induced by the harmonic excitation of a fluid in a rigid two-dimensional waveguide. The strength of the flow will depend on the time-phase between the axial velocity and the boundary velocity. It was found in previous work that the amplitude of the flow is directly related to the streaming Reynolds number. These results will be used to compare with Bekesy's observations.

4:15

4pPA13. Acoustic performance predictions via a general theory for the scattering of sound by sound with experimental data from an operational parametric sonar system. Harvey C. Woodsum (Sonetech Corp., 10 Commerce Park North, Unit 1, Bedford, NH 03110)

In results previously described [J. Acoust. Soc. Am. 95, 2PA14 (1994)], a general theory for the scattering of sound by sound has been developed as an exact solution to the Lighthill–Westervelt equation of nonlinear acoustics. Most recently, a computer model based on this theory has been developed in order to support design analysis and performance prediction for parametric array echo-ranging systems and has been applied to a currently deployed operational parametric sonar. The present theory and model have the potential to support a wide range of analyses, including development of beam patterns, source levels, transient waveform effects, as well as the analysis of scattering of sound by sound from intersecting beams as well as the prediction of performance of parametric sources having multiple simultaneous transmit beams. Both the theoretical basis for the model and agreement with experimental data are presented.

4:30

4pPA14. Imaging multiple masked nonlinear scatterers applying a combination of time reversal principles and the selective source reduction method. Siegfried Vanaverbeke, Koen Van Den Abeele (Wave Propagation and Signal Processing, K.U. Leuven Campus Kortrijk, E. Sabbelaan 53, B-8500 Kortrijk, Belgium, koen.vandenabeele@kuleuven-kortrijk.be), Brian E. Anderson (Brigham Young Univ., Provo, UT 84602), and Marco Scalerandi (Politecnico di Torino, 10129 Torino, Italy)

Inherent limitations of the traditional time reversal process in the case of multiple sources or scatterers make it impossible to distinguish them individually. The selective source reduction (SSR) method employs a subtraction technique to selectively suppress in amplitude (and ideally eliminate) a time reversed focal signal that is masking another focus. In previous work, Scalerandi *et al.* and Anderson *et al.* successfully applied the SSR method to identify masked primary sources in a fully linear medium. Here, we extend the capabilities of the SSR method to deal with (i) secondary sources (for example, scattering caused by embedded defects) and (ii) nonlinear scattering generated during the ultrasonic wave propagation as well. We call this new method selective source reduction based on nonlinear time reversed acoustics (SSR-NLTRA). In the extended approach, the contribution of all primary sources is first eliminated by means of the scaling subtraction method. Subsequently, the SSR-TRA method is applied to the remaining nonlinear content of the signals. We show by means of two dimensional wave propagation simulations that the new method can be applied iteratively to successfully image multiple masked nonlinear defects.

THURSDAY AFTERNOON, 22 APRIL 2010 GRAND BALLROOM VII/VIII/IX, 2:00 TO 6:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Physiology, Models, and Auditory Processing (Poster Session)

Monita Chatterjee, Chair
Univ. of Maryland, Dept. of Hearing and Speech Science, College Park, MD 20742

Contributed Papers

All posters will be on display from 2:00 p.m. to 6:00 p.m. To allow contributors an opportunity to see other posters, contributors or odd-numbered papers will be at their posters from 2:00 p.m. to 4:00 p.m. and contributors or even-numbered papers will be at their posters from 4:00 p.m. to 6:00 p.m.

4pPP1. Vacuum ear plug. Martin L. Lenhardt (Dept. of Biomedical Eng., Virginia Commonwealth Univ., P.O. Box 980168 MCV, Richmond, VA 23298-0168, lenhardt@vcu.edu)

A passive inexpensive hearing protection device (HPD) will be described that provides hearing protection, which is simple to use and comfortable over hours of use. Protection from impact sounds will be provided by creating a negative pressure between the plug tip and the eardrum (-100-mm H2O or -1 kPa). The negative pressure will be created by squeezing the peripheral plug end, evacuating a fixed amount of air from the canal. The aperture diameter is 0.010 in. In effect the pump plug will have a similar acoustical action on the eardrum as a normal stapedius muscle contraction, but without its limitations (too slow and fatigue). An external flexible bladder forms the end which is held in the fingers to be inserted into the ear canal. The vacuum mechanism consists of bulb on a polypropylene frame embedded in a closed cell urethane flanged housing. With a negative pressure of -2 kPa the transmission loss is 20–25 dB. It is estimated that a negative pressure of -1 kPa would result in a transmission loss of about 15 dB. The vacuum effect is maximal in the low frequencies which will also attenuate body conducted sounds by reducing ossicular inertial.

4pPP2. Development and validation of a computational model of bone-conducted sound transmission for improved hearing protection design. Odile H. Clavier (Creare Inc., 16 Great Hollow Rd., Hanover, NH 03755, ohc@creare.com), Margaret Wismer (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801), Jed Wilbur, Anthony Dietz (Creare Inc., Hanover, NH 03755), and William O'Brien, Jr. (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

Bone-conducted sound is the limiting factor in current hearing protection for very high-noise environments such as those encountered by maintainers of military aircraft. The University of Illinois has developed a three-dimensional acoustic wave propagation model for the computation of sound transmission into, around, and through fluid and solid-shell bodies. Creare has implemented experimental techniques to validate the computational results. Three cases are presented. The first is a fluid sphere, for which experimental, computational, and analytical results were obtained. The second is a solid-shell, fluid-filled sphere, for which experimental and computational results were obtained to determine the effect of the solid shell. The third is a representative human head with and without passive hearing protection. For this case, the computational model and an experimental head simulator were both developed from a computed tomography scan of a living human head. The head simulator was built around a rapid-prototype instrumented skull using silicon organs and simulated tissue. Experimental, computational, and analytical results were all in good agreement for the fluid sphere test case. While general agreement was obtained for the other two cases, specific discrepancies in the results are outlined and limitations of the models are discussed.

4pPP3. Application of matched asymptotic expansions to the analysis of compressional bone conduction. David Chhan and Charles Thompson (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, david_chhan@student.uml.edu)

In this work, we will investigate the compressional mode of bone conduction and its effect on the displacement of the cochlear partition. We will use the method of matched asymptotic expansions to obtain the velocity of the partition. In this mode of bone conduction, the cochlea is compressed alternatively due to an applied vibratory force. Velocity difference between the oval and round windows is shown to be critical in determining the amplitude of the partition motion. From this physical process, we will analytically derive equations that will be used to solve for the velocity of the cochlear partition due to the compressional velocity on the cochlear shell.

4pPP4. Exploring the traveling wave dispersion in the cochlea. Sripriya Ramamoorthy, Ding-Jun Zha, and Alfred Nuttall, L (Dept. of Otolaryngol ./Head and Neck Surgery, Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., NRC 04, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, ramamoor@ohsu.edu)

In the measurements of the basilar membrane velocity as well as the auditory nerve responses to acoustic stimulation, it has been observed that low frequencies arrive at the measurement location (or the tonotopically located auditory nerve fiber) earlier and high frequencies arrive later, with significant delay noticed in the characteristic frequency component. In the derived impulse responses as well as click responses, this phenomenon manifests as the instantaneous frequency glide from low to high frequencies. The origin of these frequency glides has not yet been satisfactorily explained. In this paper, a simple elucidation along with experimental validation using measurements made on guinea pigs is presented for the plausible origin of the frequency glides observed in the cochlea.

4pPP5. Observing chartacteristic frequency shifts using chirp-evoked otoacoustic emissions. Luke Shaheen, Michael Epstein, and Ikaro Silva (Dept. Speech-Lang. Path. and Audiol., Auditory Modeling and Processing Lab. (AMPLab), Ctr. for Comm. and Digitial Signal Processing (Dept. Elec. and Comp. Eng.), Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, m.epstein@neu.edu)

Otoacoustic emissions (OAEs) are often used to assess the integrity of specific regions of the cochlea. There is, however, evidence that the frequency that causes maximal response for a particular cochlear region shifts toward lower frequencies at high-stimulus levels. Therefore, OAE-based assessments of cochlear activity over a wide range of levels may need to take into account these shifts. The present study employed two experiments in order to confirm and characterize the extent to which this shift in cochlear excitation as a function of level can be observed in OAEs. In the first experiment, chirp-evoked otoacoustic emissions (ChOAEs) were used to track changes in the frequency spectra of each listener. The results showed a shift toward low frequencies in the spectral peak of ChOAE response in a majority of listeners (−9.1% average shift). This shift was then confirmed using a single-listener measurement of tone-burst otoacoustic emission input/output functions measured at three different frequencies. These results support the contention that it may be necessary to vary stimulus frequency to take the cochlear excitation shift into account when making assessments of the cochlear activity at a particular location across a wide range of levels.

4pPP6. Growth of otoacoustic emission suppression as a function of frequency and level in humans. Michael P. Gorga, Stephen T. Neely, Judy, G. Kopun, and Hongyang Tan (555 N. 30th St., Omaha, NE 68131)

DPOAE suppression was measured for 11 suppressor frequencies (f3) surrounding each of eight f2 frequencies (0.5–8 kHz) and six L2 levels (10–60 dB SL). A total of 63 normal-hearing subjects participated, with data from 20 subjects at each f2, L2 combination. Measurement-based stopping rules were used, such that averaging continued until the noise −25 dB SPL, the SNR 25 dB, or 210 s of artifact-free averaging had been completed. These stopping rules resulted in reliable measurements, even for conditions in which the SNR typically is poor (low f2 frequencies and low L2 levels). Suppression growth as a function of f3 was similar across f2 frequencies and L2 levels. Specifically, low-frequency suppressors relative to f2 had higher suppression thresholds but more rapid growth of suppression, compared to suppressor frequencies close to f2. Suppression growth for f3 frequencies above f2 was slow, even in comparison to suppression growth when f3 ¡Ö f2. This overall pattern was evident for all f2 frequencies and for a wide range of L2 levels. These findings suggest that suppression growth is similar across a wide frequency range in humans. [Work supported by the NIDCD R01 2251 and P30 4662].

4pPP7. Otoacoustic emission suppression tuning curves in humans. Michael, P. Gorga, Stephen, T. Neely, Judy, G. Kopun, and Hongyang Tan (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

DPOAE suppression data as a function of suppressor level (L3) for f2 frequencies from 0.5 to 8 kHz and L2 levels from 10 to 60 dB SL were used to construct suppression tuning curves (STCs). DPOAE levels in the presence of suppressors were converted into decrement versus suppressor level (L3) functions, and the L3 resulting in 3-dB decrements was obtained by transformed linear regression. These L3 levels were plotted as a function of f3 to construct STCs. STCs when f3 is plotted on an octave scale were similar in shape across f2 frequency. These STCs were analyzed to provide estimates of gain (tip-to-tail differences) and tuning (QERB). Both gain and QERB decreased as L2 increased, regardless of f2, but the increase with f2 was not monotonic. A roughly linear relation was observed between gain and QERB at each frequency, such that gain increased by 4–13 dB (mean xa1;xd6; 8 dB) for every unit increase in QERB, although the pattern varied with frequency. These findings suggest consistent nonlinear processing across a wide frequency range in humans, although the nonlinear operation range is frequency dependent. [Work supported by the NIDCD R01 2251 and P30 4662].

4pPP8. Influence of hair bundle configuration on biomimetic hair sensor sensitivity. Shuangqin Liu and Robert White (Mech. Eng., Tufts Univ., Medford, MA 02155, shuangqin.lin@tufts.edu)

Hair cells are the sensory receptors of the auditory system. When the stereocilia (hair buddles) sitting on top of the hair cells are stimulated, a current flows through the hair cell transducing mechanical stimuli into electrical response. For the outer hair cell (OHC), the hair bundles are stimulated by displacement of the tectorial membrane relative to the reticular lamina. For the inner hair cell (IHC), the hair bundles are stimulated by force due to motion of the fluid around the hair bundle. In addition, OHC and IHC hair bundles are shaped differently. In this paper, biomimetic micromachined hair sensors are designed, fabricated, and tested to investigate how the configurations of the hair bundles affect the sensitivity of the hair sensor. These hair sensors use a capacitive scheme and have different hair configurations including the W and U shape representing OHC and IHC hair bundles. Computational results achieved to date indicate that sensitivity can be strongly affected by the placement of the engineered hair bundles. Microfabrication and experimental work are ongoing.

4pPP9. Spatiotemporal coding of signals in the auditory periphery. Ramdas Kumaresan, Vijay Peddinti (Dept. of Elec. and Comput. Eng., Univ. of Rhode Island, Kelley Hall, Kingston, RI 02881), and Peter Cariani (Harvard Med. School, Newton, MA 02460)

Signal representation in the cochlea is often thought to involve either rate-place profiles or purely temporal, interspike interval codes. Spatiotemporal coding strategies based on phase-locking, cochlear delays, and coincidence detectors have also been proposed [Loeb *et al.*, Biol. Cybern. (1983); K. & Shamma, J. Acoust. Soc. Am. **107** (2000); and Carney *et al.*, Acoustica **88**, 334–337 (2002)]. In this view, spatiotemporal patterns of spikes locked to relative phases of the traveling wave at specific cochlear places at a given time can convey information about a tone. We propose a general mathematical basis for using such spatial phase/amplitude patterns along the frequency axis to represent an arbitrary (approximately) time and bandwidth-limited signal. We posit that the spatial pattern of phases and amplitudes corresponds to locations at which (real and/or imaginary parts of) the Fourier transform of the signal crosses certain levels (e.g., zero level). Given these locations, we show that we can accurately reconstruct the original signal by solving a simple eigenvalue problem. Using this approach, we propose an analysis/synthesis algorithm to represent speech-like signals. We conjecture that a generalized representation of the forms of signals can be inferred from spatial, cross-CF patterns of phase relations present in the auditory nerve. [Work supported by AFSOR FA9550-09-1-0119.]

4pPP10. Perception and discrimination of synchronous and asynchronous tones. Magdalena Wojtczak, Andrew J. Oxenham, and Anna C. Baird (Dept. of Psych., Univ. of Minnesota, Elliott Hall N218, 75 East River Rd., Minneapolis, MN 55455)

Studies of auditory-brainstem responses show that the response to a click is less synchronized than the response to an upward-chirp designed to compensate for the dispersion of basilar-membrane traveling-wave [Dau *et al.*, J. Acoust. Soc. Am. **107**, 1530–1540 (2000)]. However, upward-chirps are perceived as less compact than the clicks despite producing a more synchronized response along the basilar-membrane, suggesting the existence of a mechanism compensating for different traveling-wave delays [Uppenkamp *et al.*, Hear. Res. **158**, 71–83 (2001)]. This study evaluated synchrony and asynchrony perception and discrimination using tonal stimuli that excited remote places along the basilar membrane. Potential within-channel cues were masked using noise bands that were geometrically centered between the test tones. Level effects were investigated using two levels of the test tones, 20-dB SL and 85-dB sound pressure level (SPL). In addition, perception and discrimination of synchrony and asynchrony were measured for tones presented in noise bands at signal-to-masker ratio of 20 dB. The pattern of results for masked 85-dB SPL tones resembled that for unmasked 20-dB SL tones. The role of auditory-filter bandwidths and the existence of a compensating mechanism in the perception of across-channel synchrony and asynchrony will be discussed. [Work supported by NIH Grants R01DC03909 and R01DC010374].

4pPP11. Limiting factors in auditory discrimination of frequency ratios. Christophe N. J. Stoelinga and Robert A. Lutfi (Dept. of Communicative Disord., Auditory Behavioral Lab., Univ of Wisconsin, Madison, Wisconsin 53706)

Frequency ratios convey meaningful information for speech and many natural sounds. However, little is known regarding our ability to discriminate frequency ratios [see Fantini and Viemeister (1987). *Auditory Processing of Complex Sounds* (Lawrence, Hillsdale, NJ)]. The present study measured the relative contribution of two factors expected to limit discrimination: decision criterion and internal noise. In an adaptive, two-interval, forced-choice procedure with frequency rove, 11 highly trained listeners discriminated a change in the frequency ratio between two equal-intensity tones. The separate influence of decision criterion and internal noise was determined from scatter plots giving listeners' trial-by-trial responses as a function of the frequencies of the two tones. The plots reveal listeners to initially base their judgments on one tone alone (almost always the higher of the two) but after training to place greater reliance on both tones. Some listeners were able to achieve the optimal decision criterion of weighting both tones equally, so that the deviation from perfect performance was due only to internal noise. With continued practice over the course of many trials (at least ten 1000-trial sessions), listeners improved their performance by either improving their decision criterion or reducing internal noise. [Research supported by NIDCD grant 5R01DC006875-05.]

4pPP12. Diassociating spectral and temporal influences in an AM-QFM (amplitude modulated and quasi-frequency modulated) discrimination task. Ewa Borucki and Berg Bruce (Dept. of Cognitve Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697-5100, eborucki@uci.edu)

This study investigated temporal and spectral influences in a task used to investigate the bandwidths of phase sensitivity. Subjects discriminated amplitude modulated (AM) tones and quasi-frequency modulated (QFM) tones in a 2IFC. An adaptive threshold procedure was used to estimate modulation depth needed to make the discrimination as a function modulation frequency for a 2000-Hz carrier. Threshold functions were often nonmonotonic, with nonmonotonicities observed at higher-modulation frequencies. This is likely due to the effects of distortion products creating salient spectral cues. When stimulus duration is decreased from 200 to 50, 20, or 10 ms, thresholds for low-modulation frequencies decreased to near-chance levels, whereas thresholds in the region of nonmonotonicities were largely unaffected. The decrease in stimulus duration appears to hinder the listener's ability to use temporal cues in order to discriminate between AM and QFM, whereas spectral information derived from distortion product cues appears to be resilient. [Work supported by NSF BCS-07464003.]

4pPP13. The influence of practice on the discrimination of spectro-temporal modulation depth. Andrew T. Sabin (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60201, a-sabin@northwestern.edu), David A. Eddins (Univ. of Rochester, Rochester, NY), and Beverly A. Wright (Northwestern Univ., Evanston, IL 60201)

The pattern of sound energy spread across frequency and time (spectro-temporal modulation) is a crucial stimulus cue for sound identification. Improvements in the sensitivity to this modulation could potentially aid performance on numerous real-world tasks, yet it is unknown how practice influences this sensitivity. To investigate this issue, normal-hearing adults ($n=8$) were trained ~1 h/day for 7 days to discriminate between noises with the same spectro-temporal modulation drifting upward in audio frequency but with different modulation depths. Performance on the trained condition and on four untrained conditions was examined both before and after the training phase. Depth-discrimination thresholds improved significantly on the trained condition and on an untrained condition with a downward-drifting modulation. Discrimination thresholds also improved in subsets of listeners on untrained isolated component modulations (spectral or temporal) of the trained spectro-temporal modulation. Finally, the ability to detect the trained spectro-temporal modulation worsened significantly in proportion to the amount of improvement on the trained discrimination task. These data suggest that training on depth discrimination may be a means of improving sensitivity to spectro-temporal modulation depth in real-world stimuli, but potentially at the cost of the ability to detect these modulations. [Work supported by NIH/NIDCD]

4pPP14. Transient sex differences during development on two auditory tasks attributable to earlier maturation in males. Julia Jones Huyck and Beverly A. Wright (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, juliajoneshuyck@u.northwestern.edu)

Recent evidence suggests that the development of naïve performance on some auditory perceptual tasks can continue well into adolescence. Of interest here was whether there are sex differences in the maturation rate on three such tasks, temporal-interval discrimination, tone detection in forward masking, and tone detection in backward masking. To investigate this issue, performance was compared between males and females in multiple age groups on these three tasks (n per sex in each age group=9–20). There were no sex differences for backward masking. However, males reached adult-like performance earlier than females on the other two tasks: forward masking (males at ~12 years, females at ~15 years) and temporal-interval discrimination (males at 14 years, females at >14 years, on each of three

separate conditions). Surprisingly, the male advantage occurred only during adolescence. Thus, on two of the three late-developing auditory abilities examined, males matured more quickly than females, but did not differ from females either early in development or once mature performance had been reached. These results demonstrate that there can be transient sex differences during development owing to differences in maturational rate. [Work supported by NIH/NIDCD.]

4pPP15. Spectral weight analysis of commodulation masking release. Hisaaki Tabuchi and Bruce G. Berg (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza. Irvine, CA 92617-5100)

Spectral weight estimates are obtained in a commodulation masking release (CMR) experiment in which masking and flanking bands each consist of five equal-intensity tones spanning 20 Hz. The masking and flanking bands are centered at 1000 and 900 Hz, respectively, and the signal is an increment in the intensity of the 1000 Hz component. The phases for the five components of the masking band are randomly sampled on each trial and are identically assigned to the components of the flanking band, yielding commodulated bands. A small, random intensity perturbation is added to each component of the stimulus, and an analysis of trial-by-trial responses provides spectral weight estimates. Spectral weights are also obtained from simulations of two multiple channel models [Hall *et al.*, J. Acoust. Soc. Am. **76**, 50–56 (1984); S. Buus, J. Acoust. Soc. Am. **78**, 1958–1965 (1985)] and one single-channel model [B. G. Berg, J. Acoust. Soc. Am. **100**, 1013–1023 (1996)]. Data for most subjects show strong support for Hall's multiple-channel model. [Work supported by NSF BCS-07464003.]

4pPP16. Relationship between physiological and psychoacoustical sensitivity to amplitude and frequency modulation. Michelle Hsieh, Craig Champlin, and Su-Hyun Jin (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, michellehsieh@mail.utexas.edu)

Various researchers have found a correlation between the auditory steady state response (ASSR) and corresponding behavioral measurements such as speech recognition scores [e.g., Dimitrijevic *et al.* (2001)]. However, relatively few studies have examined the sensitivity to small changes in amplitude and frequency modulation depths. In the present study, it was hypothesized that physiological individual differences in sensitivity to amplitude and frequency modulation depths would be reflected in corresponding psychophysical measures. Auditory steady-state responses were collected in response to amplitude- and frequency-modulated pure tone carriers (500 and 3000 Hz) in normally hearing listeners over a range of modulation depths at several different modulation rates. Participants also completed a psychophysical task in which they were asked to detect the modulated tone in a two-interval forced choice testing paradigm. The relationships between the ASSR and the psychophysical performance will be discussed.

4pPP17. A hybrid procedure for psychometric function estimation. Harisadhan Patra, Daniel L. Valente, and Walt Jesteadt (Boys Town Natl. Res. Hospital, 555 N 30th St., Omaha, NE 68131)

Threshold, defined as the stimulus level required for a predefined percent correct response, is often used to measure a listener's performance. Psychometric functions (PFs) provide better insight to the underlying decision process. Two parameters, threshold (α) and slope (β), are sufficient to define a PF. Estimation of α and β from individual trials of adaptive procedure tracks is time-consuming, costly, and dependent upon the step-size choice. A procedure has been developed and written in MATLAB, which provides stable PF parameter estimates of α and β and also their confidence intervals. The procedure consists of three stages. First, signals are presented adaptively to estimate levels corresponding to 71% and 87% correct. Second, the signal is pseudo-randomly presented at one of five fixed levels equally spaced over that range. Finally, the program implements a maximum-likelihood procedure updated after every trial to estimate signal levels corresponding to 63%, 71%, 76%, 79%, and 87% correct. After 140 trials, the program estimates α and β from the entire run and generates a PF curve that typically accounts for at least 90% of the total variance. The reliability and validity of the procedure are analyzed using simulations and observed data. [Work supported by NIH.]

4pPP18. Fundamental frequency and pitch shift discrimination. William A. Yost, Christopher A. Brown, and Farris Walling (Speech and Hearing Sci., Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

The pitch-shift of the residue has been used as an argument for the importance of temporal fine structure in pitch perception. Discrimination of a change in fundamental frequency (F0) for harmonic complexes was compared to discriminating a change in the shifted frequency (f) in pitch-shift of the residue stimuli. Patterson and Wightman [J. Acoust. Soc. Am. **59**, (1976)] showed that a linear relationship exists between matched pitch and the frequency of the lowest spectral component for pitch shift of the residue stimuli (stimuli for which the spacing between adjacent spectral components is constant, but all components in the complex are shifted up or down in frequency). Obtaining pitch matches for pitch shift of the residue stimuli is a difficult task because of the ambiguous nature of the pitch shift of the residue in several conditions. Using the two discrimination experiments allows for an estimate of the slopes relating matched pitch to the frequency of the lowest component. Slopes estimated from the discrimination experiments will be compared to those obtained by the previous authors to determine if the discrimination experiments provide a valid way to predict the matched pitch of pitch-shift of the residue stimuli. [Research supported by NIDCD.]

4pPP19. Extraction of fundamental pitch using Euclid's algorithm implemented through iterative demodulation. Ramdas Kumaresan, Jiun-yee Li (Dept. of Elec. and Comput. Eng., Univ. of Rhode Island, Kelley Hall, Kingston, RI 02881), and Peter Cariani (Harvard Med. School, Newton, MA 02460)

Strong periodicity or "virtual" pitches are heard at the fundamentals (F0s) of harmonic complexes. As an alternative to both spectral pattern matching and autocorrelation-like analysis, we propose a novel mechanism that extracts the pitch directly from the signal by iterative demodulation and low-pass filtering. In effect the mechanism computes the greatest common divisor (gcd) of component signal harmonic numbers, using a Euclid-like iterative subtraction algorithm. For example, F0 of 560- and 320-Hz tones is 80 Hz, obtained by iteratively subtracting smaller frequencies from larger ones until they converge, e.g., $560-320=240$ Hz, $320-240=80$, $240-80=160$, and $160-80=80$. This algorithm can be adapted to compute pitches of harmonic complexes. Consider two tones, 560 and 320 Hz. As in Euclid's algorithm, the tones are iteratively multiplied and low-pass filtered to obtain the first (240-Hz), second (160-Hz), and third (80-Hz) difference tones. At the end only F0 remains. For inharmonic tones, incomplete demodulation results if there is a lower-frequency limit to the process. The mechanism converges on one pitch rather than multiple subharmonics. Such repeated multiplication and low-pass operations could conceivably be implemented in the ascending auditory pathway via phase-locked spike trains with synchrony capture, successive cross-CF convergences, low-pass modulation tunings, and cumulative decline of phase-locking at successive stations. [Work supported by AFOSR FA9550-09-1-0119.]

4pPP20. Musical context affects detection of pitch differences in tones with different spectra. Elizabeth M. O. Borchert and Andrew J. Oxenham (Dept. Psych., Univ. of Minnesota-Twin Cities, 75 East River Rd., Minneapolis, MN 55455)

Pitch plays an important role in complex auditory tasks such as listening to music and understanding speech, yet pitch comparisons are more difficult when tones have different spectra. Since tones are rarely heard in isolation, the surrounding context may help or hinder pitch comparisons. The first study presented listeners with tone pairs in isolation or immediately following a tonal context, which consisted of a portion of a descending major scale with the target as the tonic. Listeners' performance improved in the presence of the tonal context, but only when the tones within a pair differed in spectral content. In the second experiment, a variety of contexts were used, with the goal of discriminating between effects of tonal hierarchy and effects of predictability. Even with maximal predictability, presenting tone pairs after an "atonal" context consisting of notes from an octatonic scale yielded poorer performance than tone pairs in isolation. The results suggest that any advantage listeners derived from the tonal context in our first experiment was related more to placing the tone into an over-learned tonal hierarchy than to the likelihood of the target pitch within a predictable context. [Work supported by NIH Grant R01 DC 05216.]

4pPP21. Neural timing nets for fundamental frequency (F0)-based auditory scene analysis. Peter Cariani (Dept. of Otology & Laryngology, Harvard Med. School, 629 Watertown St., Newton, MA 02460) and Ramdas Kumaresan (Univ. of Rhode Island, Kingston, RI 02881)

Neural timing nets are idealized networks of delay lines, coincidence detectors, and adaptive processing elements that operate in the time domain on temporally coded signals to compare, extract, and separate auditory objects [Cariani, Neural Networks **14**, 737–753 (2001); J. New Music Res. **30**, 107–135; IEEE Trans. Neural Netw. **15**(5) (2004)]. Timing nets constitute an alternative, potential mode of neural signal processing in which information resides in neural signals rather than in patterns of activated elements. Recurrent timing nets with delay loops act as dense arrays of recursive, comb-like filters to effect a period-by-period analysis that builds up and separates component invariant time patterns with different fundamentals (F0s). Using both linear and nonlinear processing rules, the latter were used to process and separate synthetic double vowels, running speech, and polyphonic musical excerpts, with varying results. Relations to, and combinations with, processing strategies based on autocorrelation and all-order interspike intervals, adaptive comb filtering, correlogram duplex analysis, cancellation, Fourier zero-crossings, and demodulation, with and without prior bandpass filtering, are discussed. Bottom-up/top-down mechanisms for dynamic facilitation of lower level temporal processing loops are also considered. [Work supported by AFOSR FA9550-09-1-0119.]

4pPP22. The effect of stimulus context on cortical measures of pitch processing using functional magnetic resonance imaging. Daphne García (MRC Inst. of Hearing Res., Univ. Park, Nottingham, United Kingdom, daphne@ihr.mrc.ac.uk), Christopher Plack (Univ. of Manchester, Manchester, United Kingdom), and Deborah Hall (Nottingham Trent Univ., Nottingham, United Kingdom)

Many different paradigms and pitch-evoking stimuli have been used to study pitch. A growing body of neurophysiological evidence shows that cortical responses are sensitive to the context from which stimuli are presented. In this human fMRI study, we tested the hypothesis that the stimulus context influences the pattern of pitch-related auditory cortical activation. Fifteen listeners participated in a blocked design experiment. A diotic harmonic complex tone and a dichotic Huggins pitch stimulus were presented within either a noise or silent context and activation was contrasted with matched noise control conditions. Results revealed significant main effects of both context (noise/silence) and pitch stimulus (diotic/dichotic) ($p < 0.05$, corrected). While the response to context was primarily localized in primary auditory cortex, the response to pitch was more posterior. The context significantly modulated the pitch response, especially in subregions of planum temporal; non-primary auditory cortex. While the response to the diotic pitch was greater than the dichotic pitch, the overall results were broadly comparable. We therefore conclude that if the feature specificity of the pitch-related response is to be inferred from fMRI data, future studies should include careful controls for stimulus context.

4pPP23. Vocal pitch regulation depends on the baseline voice F0 feedback: An ERP study for investigating the role of auditory feedback for voice pitch error correction. Roozbeh Behroozmand and Charles Larson (Dept. of Commun. Sci. and Disord., Speech Physio. Lab., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, r-behroozmand @northwestern.edu)

Voice production and control require neural interactions between the vocal motor and auditory mechanisms. The comparison between the incoming auditory feedback and the predicted sensory input (efference copies) from a self-produced vocalization allows the detection of feedback mismatches for voice error detection and correction. However, the sensitivity of the audio-vocal system for feedback mismatch detection seems to depend on the extent of feedback deviation from the predicted vocal output. The present study investigated this effect by examining event-related potentials (ERPs) in response to a +100-cent voice feedback pitch perturbation stimulus while the extent of pre-stimulus (baseline) feedback pitch deviation was randomly manipulated at 0, 50, 100, 200, and 400 cents. Results showed that the neural responses to +100-cent pitch-shift stimuli grew systematically larger as the extent of pre-stimulus baseline pitch deviation became smaller. This finding suggests that the extent of disparity between the predicted and incoming sensory feedback of self-produced voice can affect the neural tuning processes that adjust the sensitivity of the audio-vocal mechanisms for voice pitch error detection and correction. Lower sensitivity to larger feedback mismatches may imply robustness against the disruptive effect of highly deviant or externally generated sounds during vocal production and control.

4pPP24. Molecular analysis of the effect of onset asynchrony in the identification of a rudimentary sound source in noise. Robert A. Lutfi, Ching-Ju Liu, and Christophe N. J. Stoelinga (Dept. of Communicative Disord., Auditory Behavioral Res. Lab., Univ. of Wisconsin, 1975 Willow Dr., Madison, WI 53706, ralutfi@wisc.edu)

The threshold for detecting a target in noise is often greater when the target is gated on simultaneously with the noise than when it is gated on after some delay [Zwicker, E. (**1965**). J. Acoust. Soc. Am. **37**, 653–663]. One explanation is that the perceptual principle of grouping causes the target and noise with simultaneous onsets to be perceived as a single sound source. This idea was tested for a rudimentary sound source using perturbation analysis. In a two-interval, forced-choice procedure listeners identified as target the impact sound produced by the larger of two stretched membranes. The noise on each presentation was the impact sound of a variable-sized plate. Grouping predicts that the decision weights on the noise should be positive when target and noise have simultaneous onsets, but that they should approach zero when target and noise are gated on asynchronously. This prediction was confirmed when the noise preceded the target by a fixed interval (100 ms), but not when it followed the target by the same interval or when either interval was selected at random on each presentation (100 or −100). [Research supported by NIDCD Grant 5R01DC006875-05.]

4pPP25. Target enhancement and noise cancellation in the identification of rudimentary sound sources in noise. Robert A. Lutfi, Ching-Ju Liu, and Christophe N.J. Stoelinga (Dept. of Communicative Disord., Auditory Behavioral Res. Lab., Univ. of Wisconsin, 1975 Willow Dr., Madison, WI 53705, ralutfi@wisc.edu)

The identification of targets in noise is believed to entail two processes, one that acts to enhance the target and the other to cancel the noise [Durlach *et al.* (2003). J. Acoust. Soc. Am. 2984–2987]. The relative contribution of these processes in the identification of rudimentary sound sources is unknown. In the present study, perturbation analysis was used to determine the relative contribution in terms of the sign and magnitude of listener decision weights on the noise. In a two-interval, forced-choice procedure, listeners identified as target the impact sound produced by the larger of two stretched membranes. The noise on each presentation was the impact sound of a variable-sized plate. For four of five listeners showing significant interference, the weights were positive indicating enhancement; for the remaining listeners, they were negative indicating cancellation. In a second condition, the target was the membrane hit with harder force, and the noise was a plate hit with variable force. The noise weights for all listeners in this condition indicated cancellation. The results are consistent with an interpretation in which noise cancellation is the predominant strategy for identifying targets unless it is precluded by uncertainty regarding the spectral properties of the noise. [Research supported by NIDCD Grant No. 5R01DC006875-05].

4pPP26. The effect of binaural coherence on envelope interaural time difference thresholds. Jessica J. M. Monaghan, Katrin Krumbholz, and Bernhard U. Seeber (MRC Inst. of Hearing Res., Nottingham NG7 2RD, United Kingdom, jessica@ihr.mrc.ac.uk)

Room reflections alter the envelope of sounds differently at both ears, reducing binaural coherence. Experiment 1 measured interaural time difference (ITD) discrimination thresholds for broadband and transposed speech. A sentence was convolved with binaural room impulse responses for different source-receiver distances, and the envelopes extracted and multiplied with 4-kHz tones. Preliminary results indicate that envelope ITD thresholds increased from <100 μs without reverberation to >700 μs at four times the reverberation radius. ITD thresholds for unprocessed speech were more robust to the addition of reverberation, only rising above 350 μs at the same distance. To investigate whether reverberation is detrimental owing to reduction in binaural coherence, stimulus envelopes were created by temporally jittering raised cosine pulses around a 10-ms separation. Bilaterally independent jittering allowed variation in coherence while minimizing change in other envelope parameters. Preliminary results show ITD thresholds <100 μs for coherent envelopes, increasing to >700 μs for coherences of 0.6, a value consistent with the room simulations above. Envelope coherence strongly affects ITD discrimination, suggesting that ITDs extracted from high-frequency channels may not provide useful information in many realistic situations. This has implications for bilateral cochlear implant users, as current devices provide ITDs only in envelopes.

4pPP27. Tonotopic gradients in neural interaural time difference processing: A modeling study of the medial superior olive. Yoojin Chung (Eaton-Peabody Lab., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, yoojin_chung@meei.harvard.edu) and H. Steven Colburn (Boston Univ., Boston, MA 02215)

The interaural time difference (ITD) is the primary cue for sound localization; yet many basic questions about the biophysical mechanisms remain. In the mammalian auditory system, neurons in the medial superior olive (MSO) are tuned to ITD as well as frequency. In this study, the effects of key parameters in MSO models, such as the membrane time constants and the number and strength of synaptic inputs on ITD sensitivity to pure tones, were explored with particular attention to the best-frequency dependence of these parameters. Results show that ITD tuning is dependent on neuron membrane characteristics (i.e., the shape and time constant of the membrane response), on the strength of individual synaptic inputs, and on stimulus properties such as frequency and intensity. Models with slow response times and weak synaptic inputs show good ITD sensitivity for low-frequency tones, whereas models with fast response times and strong synaptic inputs exhibit good ITD tuning for high-frequency tones. This dependence of ITD sensitivity on membrane properties in the model suggests that parameters important for ITD tuning depend on the best frequency of the neurons, in contrast to the view that the binaural mechanism is homogeneous along the tonotopic axis.

4pPP28. Level-dependent changes in perception of temporal envelope cues. Xin Wang, Jayne B. Ahlstrom, Amy R. Horwitz, and Judy R. Dubno (Dept. of Otolaryngol.-Head and Neck Surgery, Medical Univ. of South Carolina, 135 Rutledge Ave., MSC 550, Charleston, SC 29425-5500, wanxi@musc.edu)

Level-dependent changes in speech recognition may reveal effects of basilar-membrane nonlinearities on temporal envelope fluctuations. It is hypothesized that, as a result of the compressive effects of the active cochlear mechanism, the "effective" magnitude of speech envelope fluctuations will be reduced as speech level increases from lower (more linear) to conversational (more compressive) regions. With further increases from conversational levels (to a more linear region), temporal envelope fluctuations will become more pronounced. Accordingly, speech recognition will be maximized at conversational levels due to the optimal "flattened" envelope and then decline at lower or higher levels. To test these assumptions, speech recognition scores were measured as a function of level for adults with normal hearing. Speech stimuli were spectrally degraded using "noise vocoder" processing so that perceptual effects of modifications to the speech temporal envelope can be revealed. As vocoded speech level increased, background noise level also increased, maintaining a fixed signal-to-noise ratio to minimize sensation-level effects on speech recognition scores. Discussion will focus on level-dependent effects for different speech stimuli and the role of nonlinearities on perception of temporal envelope cues. [Work supported by NIH/NIDCD]

4pPP29. The use of confusion patterns to evaluate the neural basis for concurrent vowel identification. Ananthakrishna Chintanpalli (Weldon School of Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., West Lafayette, IN 47907) and Michael G. Heinz (Purdue Univ., West Lafayette, IN 47907)

Perceptual studies of concurrent vowel identification suggest that listeners with normal hearing (NH) are better able to use differences in fundamental frequency (F0) than listeners with sensorineural hearing loss (SNHL). However, the neural basis for this difference remains unknown. The present study sought to validate a neural model of concurrent vowel identification based on specific confusion patterns made by NH listeners. A standard set of five vowels was used. In each concurrent vowel pair, vowel 1 had F0 = 100 Hz, while the F0 of vowel 2 was varied between 100–126 Hz (4 semitones). NH listeners made similar confusions across all F0 differences, with a reduction in occurrence as F0 difference increased. F0 benefit varied significantly across concurrent vowel pairs. A phenomenological auditory-nerve model was cascaded with F0 segregation algorithms to predict the perceptual observations. Neural predictions showed similar confusion patterns to the perceptual data for many (but not all) concurrent vowel pairs. Validating a NH physiological model with specific confusion patterns will allow the model to be used to predict effects of specific physiological changes related to SNHL, which may be useful for improving cochlear-implant and hearing-aid signal processing strategies. [Work supported by Purdue University.]

4pPP30. Auditory channel weights for consonant recognition in normal-hearing listeners. Frederic Apoux and Eric W. Healy (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210, fred.apoux@gmail.com)

The present study evaluated the relative contribution of various regions of the frequency spectrum to consonant recognition in normal-hearing listeners. The method used in this study was specifically designed to provide an estimate of the importance of each band (i) consistent with the frequency resolution of the auditory system and (ii) irrespective of the location of information elsewhere in the spectrum. Speech stimuli were divided into 30 adjacent bands with each band corresponding approximately to the bandwidth of an auditory filter. Listeners were presented with a subset of bands to avoid ceiling effects. The importance of each band was derived from the drop in performance observed when that particular band was omitted. The spectral location of the bands was always chosen randomly except for the one band whose importance was being assessed. The results indicated a fairly homogeneous contribution of all 30 bands to consonant recognition (i.e., a flat auditory channel importance function) with only the five lowest bands (below 300 Hz) having lesser weight. In contrast, additional analyses revealed a non-uniform contribution of the bands to the transmission of voicing, manner and place of articulation. [Work supported by NIDCD.]

4pPP31. Perception of temporally interrupted speech: Effects of two concurrent gating rates on intelligibility. Valeriy Shafiro, Stanley Sheft, and Robert Risley (Comm. Dis. Sci., Rush Univ. Med. Cntr., 1015 AAC, 600 S. Paulina St., Chicago, IL 60612, valeriy_shafiro@rush.edu)

Perception of temporally interrupted speech was investigated with either one or two concurrent square-wave gating functions. Sentences were interrupted at a single rate of 1, 2, 4, 8, 16, 24, or 32 Hz with a 50% duty cycle. For each rate between 1 to 8 Hz, stimuli were additionally gated at a secondary rate ranging from twice the primary rate up to 32 Hz. The secondary gating rate thus interrupted only the segments of speech which were left intact after application of the primary gating function to reduce by half the remaining speech content independent of secondary rate. With a single gating function, intelligibility scores increased with rate, reaching perfect accuracy at 8 Hz. Application of the second gating function led to a decrease in intelligibility which was greatest when the secondary rate was twice the primary rate, a condition equivalent to a 25% duty cycle of the primary rate. Further increase in the secondary rate generally led to an improvement in intelligibility, although never to the level achieved with the primary rate alone. The incomplete recovery of intelligibility scores at fast secondary gating rates suggest disruption of separate perceptual processes by each of the two concurrent interruption rates.

4pPP32. Noise robust representation of speech in the primary auditory cortex. Nima Mesgarani (Johns Hopkins Univ., Baltimore, MD, mnima@umd.edu), Stephen David, Jonathan Fritz, and Shihab Shamma (Univ. of Maryland, College Park, MD)

It is well known that humans can robustly perceive phonemes despite substantial variability across speakers, context and natural distortions. This study examines the responses of neurons in primary auditory cortex (A1) to phonetically labeled speech stimuli in clean, additive noise and reverberant conditions. Using a linear decoder [Bialek (1991)] to reconstruct the input stimulus spectrogram from the population response, we observed that

spectrograms reconstructed from the neural responses to noisy speech were closer to the original clean spectrograms than to the noisy ones. This indicates that sound representations in A1 serve to enhance information about natural speech signals relative to noise, thus extracting signal from noise. Examining the average reconstructed phoneme spectrograms in clean and noisy speech revealed a remarkable robustness in the encoding of features important for discrimination of different phonemes. In addition, it was found that the strict linear spectro-temporal receptive field (STRF) model of A1 neurons is insufficient to explain the noise robustness observed in the neural data. However, when a non-linear synaptic depression is integrated into the inputs for the STRF model, the noise was reduced in the reconstructed spectrograms similar to what observed with the actual neural data.

THURSDAY AFTERNOON, 22 APRIL 2010 DOVER A/B, 2:00 TO 4:45 P.M.

Session 4pSA

Structural Acoustics and Vibration: Applications of Structural Acoustics and Vibrations II

Joel M. Garrelick, Chair
Applied Physical Sciences, 49 Waltham St., Ste. 4, Lexington, MA 02141

Contributed Papers

2:00

4pSA1. Wave propagation model for acoustic evaluation of polymeric thin films. Hyeong Sick Ju (Graduate Program in Acoust., The Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, huj110@psu.edu) and Bernhard R. Tittmann (The Penn State Univ., University Park, PA 16802)

Polymeric thin films are fabricated on crystalline wafers to serve as a photoresist in lithography processes of semiconductors or MEMS. Essentially these film structures are viscoelastic layers on anisotropic substrates. Acoustic evaluation for these material structures requires an appropriate wave propagation model. This paper presents a leaky surface acoustic wave (LSAW) model for the evaluation of polymeric film integrity. Scanning acoustic microscopy (SAM) operating at relative high-frequency above 100 MHz is utilized to measure the LSAW velocity. For the use of SAM, the model modifies the pre-existing anisotropic layered model by employing hysteretic absorption in viscoelastic polymer and water-loading. The model produces the mean reflectance function and mean dispersion curve, which are specialized for spherical acoustic lenses. The predicted mean dispersion shows good agreement with LSAW velocities measured at several frequencies with spherical lenses.

2:15

4pSA2. Active vibration control modules for damping, compensation, measurement and dynamic testing. Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

The active vibration control module was introduced previously as part of active vibration damping system for optical tables and other precision vibration isolated platforms. This paper describes steps to expand the application of those modules to other tasks, namely, compensation of forced vibration in local areas and dynamic testing of structures. Vibration damping of most significant structural modes had been achieved using a small number of properly placed active dampers. This did not affect forced tonal vibration. Current state of the art does not offer a practically feasible way to suppress all vibration, forced and normal, over the total table surface. However, by placing vibration control modules around a local area of the table supporting a vibration sensitive device, it is possible to abate forced tonal vibration in this area. Feedback control for vibration compensation will be discussed along with experiments demonstrating stable concerted work of several vibration control modules. In application to dynamic testing, the actuator is excited by white noise, and the sensor signal is processed to calculate the dynamic compliance. The test data show that the vibration control modules can be used to measure dynamic compliance with precision comparable to that of dedicated vibration measurement systems.

2:30

4pSA3. When is a system random? Richard H. Lyon (60 Prentiss Ln., Belmont, MA 02478)

Statistical energy analysis (SEA) is a method for estimating structural acoustic transfer functions based on a statistical model of the system. The SEA model assumes random noise excitation in frequency bands and an underlying ensemble of "similar" systems. However, since the results of a SEA calculation are nearly always applied to a single system, there is a natural question of in what sense a single system can be random. A particular structural acoustics system is considered that displays both deterministic and random behavior with a distinct transition between the two. System parameters (damping, modal density, and coupling factors) are examined to see how they might be controlling the transition from deterministic to random.

2:45

4pSA4. Novel lightweight vibration absorbers for marine structures. Ryan L. Harne and Chris R. Fuller (Virginia Tech Vib. and Acoust. Labs., 131 Durham 0238, Blacksburg, VA, 24061, rharne@vt.edu)

When combined with attached motors and rotating machinery, the lightly damped, thick plating required in maritime applications becomes a broadband noise and vibration control problem. A typical solution is to adhere heavy and dense damping materials for dissipation of the plate vibrational energy. In order to attenuate low frequencies, significant mass must be added to the structure. This paper will review the development of two, new passive treatments intended to resolve this issue. HG blankets are constructed using small masses embedded into poroelastic material. Together with the inherent stiffness of the poroelastic material, the masses become embedded mass-spring dampers and their presence is found to notably increase the low-frequency transmission loss of the host material. DVAs are compact vibration absorbers that distribute continuous mass and spring elements over the surface while generating ample reactive damping at low frequencies. This paper will overview the concepts and development of adapting DVAs and HG blankets for use on heavy plate structures, their testing for broadband control performance, as well as their versatility for thinner panels. A comparison with a conventional, marine noise control treatment will be considered. [This work was supported by Northrop Grumman Shipbuilding-Newport News.]

3:00

4pSA5. Bloch response of framed wings with upper and lower skins. R. Martinez and M. Eash (Alion Corp., 84 Sherman St., Cambridge, MA 02143)

Aircraft wings are typically composed of spanwise-periodic framing elements running from leading to trailing edge. Each frame is connected to the wing's upper and lower skins through continuity of motions among frames

and skins. This paper presents calculations of a generic wing's low-wavenumber Bloch response in global flexure per the following physical content: (1) the frames are mass controlled both in vertical translation and rotation; (2) the skins are plate-like, and respond dynamically to normal and compressional shears against each of the frames; (3) the two plates bend symmetrically with respect to the normal component of the virtual foces driving them, and antisymmetrically in compression with respect to their tangent component. The need for compressional forces and a corresponding in-plane response for each of the wing's two skins stems from the theory's lack of restrictions on frame height, which for our thick wing implies a neutral axis for global flexure that is far from the system's uppermost and lowermost material points. Our calculations showcase the effects of the skins' compression on the system's low-wavenumber response, as well as how compression defines the structure's low-frequency limit for a stiffness-controlled effective medium rendered anisotropic by its chordwise-running frames.

3:15—3:30 Break

3:30

4pSA6. Noise and vibration transmission reduction using multi-element multi-path structures. Cassidy Palas and Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Hudson Hall, Durham, NC 27708, sid.palas@duke.edu)

Principles of structural acoustics are utilized in novel ways to cancel the transmission of sound and vibration through multi-element flexible barriers. Configurations analyzed include two and three layered plates with elastic interconnections. The substructures have different wave propagation properties and boundary conditions. Both continuous and periodic discrete elastic coupling methods are examined. This research demonstrates that flexibility and controlled resonant behavior can be used to block vibration and sound transmission, even with low structural damping. The main strategies utilized are structural wave cutoff with multi-element multi-path (MEMP) structures and relative phase changes due to boundary reflections. Examples of acoustic transmission loss through panel barriers are presented, and the potential advantages and possible shortcomings of the approach are evaluated. Practical configurations for layered sound reduction materials include designs allowing multiple substructural plates to produce radiation on a given surface, leading to net cancellations of transmitted sound in certain frequency ranges. Experimental results show vibration transmission reduction for several configurations. The work has particular application to the reduction in vehicle interior noise and addresses the need for good acoustic performance of lighter weight flexible structures.

3:45

4pSA7. An improved formulation for predicting low-frequency noise transmitted through double-pane windows. Dayi Ou, Cheuk Ming Mak (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong, 08900950r@polyu.edu.hk), and Kai Ming Li (Purdue Univ., West Lafayette, IN 47097-2031)

With superior sound insulation properties over a single-panel configuration, a double-panel structure with the presence of a cavity has found a wide range of applications for sound insulation. A classical method for combining a finite element method (FEM) with a boundary element method (BEM) is used to examine the transmission of low-frequency noise through double-pane windows in the present study. The technique of component mode synthesis is applied to adjust the stiffness matrix in the FEM formulation in order to examine the effects of elastic boundary conditions on the sound transmission through these structures. However, the Green function for predicting sound propagation in a rectangular long enclosure is used in favor of the free-field Green function for predicting the pressure inside the cavity of the window panes. The predicted pressure in the cavity is then coupled with the FEM formulation for the window panes and the BEM formulations for the sound fields at the outer surfaces of window panes. A parametric study is conducted systematically to allow a detailed examination for the characteristics of sound insulation of a double-pane window at different frequency bands especially for the low-frequency components.

4:00

4pSA8. Acoustic signatures of partial electric discharges in different thicknesses of Kapton. Daniel P. Hanley (US Naval Acad., P.O. 12571, Annapolis, MD 21412, m102592@usna.edu) and Edward J. Tucholski (US Naval Acad., Annapolis, MD 21401)

Thin polymer films in the presence of high electric fields undergo partial discharge and have characteristic acoustic emissions. It is hoped that studying these acoustic signals can aid in anticipating failure of the films, thereby providing a tripwire to reduce the electric voltage in high-energy capacitor applications before failure actually occurs and the capacitor is permanently damaged. This study compared the acoustic emission from a variety of thicknesses of Kapton film between 7 and 55 μm. A laser Doppler vibrometer with a frequency response from 0–22 kHz was used to study surface vibrations of a gold coated polymer sample as voltage was raised at a controlled rate of 500 V/s from 0 V to material failure. Emissions from partial discharges prior to failure are studied. The results of these tests demonstrate the relationship between characteristic frequencies for the acoustic emission and the polymer thickness.

4:15

4pSA9. Seismic surface wave method for near surface soil exploration. Zhiqu Lu (Natl. Ctr. for Physical Acoust., The Univ. of MS, Univ. MS 38677, zhiqulu@olemiss.edu)

There are many applications that require the information of near surface soil properties. The related areas include agricultural land management, levee/dam evaluation, landmines/UXO/tunnel detection, battle field condition assessment, and site foundation characterization, to name a few. To obtain soil properties in a non-invasive manner, a multi-channel analysis of surface wave (MASW) method based on laser-Doppler vibrometer was developed recently and reported in 157th ASA Meeting, Portland, OR. This talk will present the latest development. In particular, two methods of determination of the dispersive curve, i.e., the phase slope method and the two dimensional FFT transformation method, will be discussed. The emphasis will be given to demonstrate the capabilities of the two methods in identifying the fundamental mode and high modes of Rayleigh waves. An inversion algorithm using the measured dispersive curve can back calculate the soil profile, i.e., the shear wave velocity as a function of depth. Several case studies of the MASW method will be addressed.

4:30

4pSA10. Fault classification for rotating machines using neural network. Hyungseob Han and Uipil Chong (Dept. of Comput. Eng. and Information Technol., Univ. of Ulsan, Ulsan, Korea, overhs@naver.com)

When the rotating machines in the plants malfunction during operating, they can cause huge economic losses and many casualties. For these reasons, fault detection and diagnosis of rotating machines have become very important issues. This paper proposes a system to detect and diagnose abnormal states for induction motors. Through an effective combination of wavelet transform and neural network for measured vibration signals from the motors, successful fault detection and diagnosis can be achieved. This system is divided into two parts: analysis and classification. In the analysis part, vibration signals are divided into eight subbands by using wavelet transform. The most significant features of the signals are shown in the lowest-frequency band. For an efficient classification in the neural network, input samples chosen in that band were minimized. In the classification part, through one of the representative techniques, multi-layer perceptron, all kinds of vibration signals are trained and tested. From the experimental results, the proposed system perfectly classified input signals into each fault case. Furthermore, since it does not need reference data for classification, it can perform very quickly and be implemented to a real-time system.

Session 4pSCa

Speech Communication: Speech for Tracking Human Health State, Performance, and Emotional State I

Suzanne E. Boyce, Chair
Univ. of Cincinnati, Dept. of Communication Disorders, Cincinnati, OH 45267-0379

Chair's Introduction—1:00

Invited Papers

1:05

4pSCa1. Speech analysis in accident investigation. Malcolm Brenner (Natl. Transportation Safety Board, 490 L'Enfant Plaza SW, Washington, DC 20594, brennem@ntsb.gov)

The NTSB investigates major transportation accidents in the United States to make recommendations to prevent their recurrence. The Safety Board also examines new technologies that might assist investigations. In conducting its work, the NTSB has found speech analysis a useful new technology for providing secondary evidence on operator state for issues such as psychological stress, alcohol impairment, physical straining, and hypoxia. This talk provides examples from two investigations: the grounding/oil spill of the Exxon Valdez tanker and the crash of a Boeing 737 airliner at Pittsburgh.

1:25

4pSCa2. Common voice measures as indicators of fatigue. Cynthia M. LaJambe (201 Transportation Res. Bldg., Penn State Univ., University Park, PA 16802, cml149@psu.edu), Frederick M. Brown (Penn State Univ., University Park, PA, 16802), Rebecca M. Reichardt (Towson Univ., Towson, MD, 21252), Malcolm Brenner (Natl. Transportation Safety Board, Washington, DC, 20594), and Robert A. Prosek (Penn State Univ., University Park, PA, 16802)

Unobtrusive, economical, and readily accessible fatigue-monitoring technologies are needed especially in transportation, military operations, and security industries. Voice analysis is compatible with operational settings, given its minimal interference with hands-on work duties. Controlled laboratory studies are underway to establish the sensitivity of this fatigue-monitoring method. A recent study evaluated sleep-deprivation consequences on basic voice attributes using multiple speech tasks. Twenty-six native English-speaking 18-26 year-old subjects were screened for physical and psychological problems. Several sleep/wake cycles were monitored with actigraphy prior to laboratory participation. Vocal measures were compared between 13 speakers sleep deprived for 36 hours and 13 non-sleep-deprived controls. In the laboratory, speech was recorded during baseline sessions and on the following day. Group differences varied by speech task and vocal measure, with more sleep-deprivation sensitivity found, for example, in speaking rate as compared to fundamental frequency. Fatigue-related changes in vocal measures were associated with decrements in psychomotor reaction times and cognitive performance. Results are compared with previous studies relating psycho-physiological states to basic voice measures. Design considerations are discussed.

1:45

4pSCa3. Speaker assessment: The impact of environment on speech systems and individuals. John H. L. Hansen (Dept. of Elec. Eng., CRSS: Ctr. Robust Speech Systems, Univ. of Texas at Dallas, Richardson, TX 75083, john.hansen@utdallas.edu)

Assessing speaker variability is critical in developing a scientific understanding or model of human speech production. The environmental context plays a significant role in how this variability plays out. In this study, recent findings are presented on the variability of speech production due to environmental factors that influence man-machine interaction as well as human-to-human interaction. Speech produced under task stress, emotional stress, and background noise (resulting in Lombard effect) all cause speech production changes. This impacts both speech processing algorithms intended for speech recognition/technology and human-to-human interaction. Specifically, two speech production domains are briefly considered including Part-1: speech production under varying types/levels of background noise and how this produces flavors of Lombard effect and impacts speaker recognition systems, and Part-2: assessing the stress/emotional state of parents/care-givers in quantifying the language learning exposure of children (ages 10–36 months). In Part-1: the UT-Scope corpus is employed with speech from 30 subjects (19M,11F) for analysis of duration and spectral tilt as well as developing an automatic Lombard effect classification scheme which is incorporated into speaker recognition. Next, Part-2: considers how neutral versus stressed/emotional state of adults impacts conversational turns and adult word-count in a child language learning environment (20 child-parent interactions).

2:05

4pSCa4. Intelligibility of speech produced under sleep-deprivation conditions. Suzanne Boyce (Dept. of Comm. Sci. Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267-0379, boycese@ucmail.uc.edu), Joel MacAuslan (S.T.A.R. Corp., Bedford, MA 01730), Sandra Combs, and Alexandra Blood (Univ. of Cincinnati, Cincinnati, OH 45267-0379)

In previous work, we applied the technique of acoustic landmark detection to speech produced under rested vs sleep-deprived conditions. We found significant differences for both the number and pattern of landmarks detected. In a parallel study we found significant differences in clear vs conversational styles of speech. While it has been shown by a number of investigators that clear and conversational speech styles differ in the degree to which intelligibility to listeners is preserved in noise, it is not clear whether the articulatory changes found in sleep-deprived speech affect the ability of listeners to understand what is said, especially in noise. In this paper, we present the results of a study in which normal-hearing listeners are asked to transcribe speech presented with and without background noise. Similar levels of background noise have been shown to reduce speech intelligibility for clear vs conversational speech presented to normal listeners. The speech in this study was produced under rested and sleep-deprived conditions. Results will be compared to effects of clear vs conversational speech presented in noise and in quiet.

2:25

4pSCa5. On the acoustics of emotion in speech: Desperately seeking a standard. Bjoern Schuller (Inst. for Human-Machine Commun., Technische Universitaet Muenchen, D-80333 Muenchen, Germany, schuller@tum.de)

Researchers concerned with automatic recognition of human emotion in speech have proposed a considerable variety of segmental and supra-segmental acoustic descriptors. These range from prosodic characteristics to voice quality to acoustic correlates of articulation and represent unequal degrees of perceptual elaboration. Recently, evidence has been reported from first comparisons on multiple speech databases that spectral and cepstral characteristics have the greatest potential for the task [B. Schuller *et al.*, Linguistic Insights **97**, 285–307 (2009)]. Yet, novel acoustic correlates are constantly proposed, as the question of the optimal representation remains disputed. The task of evaluating suggested correlates is non-trivial, as no agreed "standard" set and method of assessment exists, and inter-corpus substantiation is usually lacking. Such substantiation is particularly difficult owing to the divergence of models employed for the ground-truth description of emotion. To ease this challenge, using the potency-arousal-valence space as the predominant means for mapping information stemming from diverse speech resources, including acted and spontaneous speech with variable and fixed phonetic content on well-defined binary tasks is proposed. Among the various options for automatic classification, a method combining static and dynamic features representing pitch, intensity, duration, voice quality, and cepstral attributes is recommended.

2:45

4pSCa6. The detection of stress, emotion, and deception from speech: The intersection of phonetics, policy, and politics. James Harnsberger (Dept. Comm. Disord., Univ. of Florida, 68 Dauer Hall, Gainesville, FL 32611)

While prior research on the detection of stress, emotion, and deception from speech and language has shown limited progress, this has not prevented the marketing of commercial devices that purport to detect these states to a variety of customers, such as law enforcement agencies, the military, intelligence agencies, homeland security, and insurance companies. For the major products currently on the market, all independent studies to date have failed to verify their efficacy with a wide range of speech materials collected under various experimental conditions, ranging from laboratory studies with carefully controlled speech to "mock crimes" to speech produced under realistic levels of jeopardy. This literature (including two studies by the author) will be reviewed and discussed in terms of how their experimental design and results are shaped and used in policy debates by private manufacturers, elected officials and their staffers, academic researchers, and others.

3:05—3:20 Break

3:20

4pSCa7. Automatic methods to monitor the speech of Parkinson's patients with deep brain stimulators. Craig van Horne (Caritas Neurosurgery, 736 Cambridge St., CCP 8, Brighton, MA 02135), Karen Chenausky, Joel MacAuslan (STAR Corp., Bedford, MA 01730), Carla Massari, and Marianna McCormick (Caritas-St. Elizabeth's, Brighton, MA, 02135)

Parkinson's disease (PD) is a neurodegenerative disease causing hypokinetic dysarthria, associated with "blurred" or underarticulated speech, imprecise consonants, and, sometimes, irregular syllable trains. Within the past decade, deep brain stimulation (DBS) of the subthalamic nucleus (STN) has provided substantial benefit to PD patients. DBS treatment has largely been directed toward the motoric features of PD: bradykinesia, rigidity, and tremor, but its effects on speech vary. The speech of PD patients receiving DBS treatment, with or without accompanying medical therapy, was analyzed for rate (syllables per second), regularity (relative deviation of syllable length), stop consonant spirantization (a measure of stop consonant precision), vowel ratio (length of vowel to length of syllable), and other features using automatic routines written specifically for the purpose. Patients' speech is more variable on DBS stimulation than on medication or no treatment. It is possible to find a combination of DBS settings for each patient that relieves their motor symptoms and returns their speech to normal. These findings suggest that it is possible to improve speech along with the general motor symptoms of PD. Furthermore, automatic analyses show promise as sources of feedback for neurologists to use in optimizing DBS settings for speech.

3:40

4pSCa8. Impact of cognitive load and frustration on drivers' speech. Hynek Bořil (Erik Jonsson School of Eng. and Comp. Sci., The Univ. of Texas at Dallas, 2601 N. Floyd Rd. 75080, Richardson, TX 75083-0688, hynek@utdallas.edu), Tristan Kleinschmidt (Speech and Audio Res. Lab., Queensland Univ. of Technol., GPO Box 2434, Brisbane, Queensland 4001, Australia), Pinar Boyraz, and John H. L. Hansen (The Univ. of Texas at Dallas, Richardson, TX 75083-0688)

Secondary tasks such as cell phone calls or interaction with automated speech dialog systems (SDSs) increase the driver's cognitive load as well as the probability of driving errors. This study analyzes speech production variations due to cognitive load and emotional state of drivers in real driving conditions. Speech samples were acquired from 24 female and 17 male subjects (approximately 8.5 h of data) while talking to a co-driver and communicating with two automated call centers, with emotional states (neutral, negative) and the number of necessary SDS query repetitions also labeled. A consistent shift in a number of speech production parameters (pitch, first format center frequency, spectral center of gravity, spectral energy spread, and duration of voiced segments) was observed when comparing SDS interaction against co-driver interaction; further increases were observed when considering negative emotion segments and the number of requested SDS query repetitions. A mel frequency cepstral coefficient based Gaussian mixture classifier trained on 10 male and 10 female sessions provided 91% accuracy in the open test set task of distinguishing co-driver interactions from SDS interactions, suggesting—together with the acoustic analysis—that it is possible to monitor the level of driver distraction directly from their speech.

THURSDAY AFTERNOON, 22 APRIL 2010 GRAND BALLROOM V, 4:05 TO 5:05 P.M.

Session 4pSCb

Speech Communication: Speech for Tracking Human Health State, Performance, and Emotional State II (Poster Session)

Suzanne E. Boyce, Chair
Univ. of Cincinnati, Dept. of Communication Disorders, Cincinnati, OH 45267-0379

Contributed Papers

All posters will be on display and all authors will be at their posters from 4:05 p.m. to 5:05 p.m.

4pSCb1. Using temporal cycles in spontaneous speech to quantify linguistic impairments in patients with neurodegenerative disorders. Eden Kaiser (Prog. in Linguist., Univ. of Minnesota, 214 Nolte Ctr., 315 Pillsbury Dr. SE, Minneapolis, MN 55455, kaise113@umn.edu), Serguei V. S. Pakhomov, Angela K. Birnbaum, Daniel Boley (Univ. of Minnesota, Minneapolis, MN 55455), and David S. Knopman, (Mayo Clinic, Rochester, MN 55905)

Fronto-temporal lobar degeneration (FTLD) is a form of dementia which may manifest through symptoms similar to Alzheimer's, including language-specific impairment. Linguistic manifestations of the disorder are often described in subjective assessments of the disorder, but not always easily quantifiable using objective tests. This paper investigates a speech characteristic not yet assessed in FTLD patients, that of "temporal cycles" [Henderson *et al.* (1966); Butterworth and Goldman-Eisler (1979)]. Temporal cycles in the speech of healthy adults consist of alternating and roughly equal periods of fluent and hesitant speech [Roberts and Kirsner (2000)]. A time series analysis of temporal cycles was conducted using spontaneous speech from 45 adults diagnosed with FTLD. Patients' cognitive functioning was assessed using the language-specific clinical dementia rating (CDR) scale by board-certified neurologists. Periodicity of temporal cycles was quantified using the proportion of the energy in the highest peak to the total energy in the power spectrum. It was found that this measure was correlated with independent CDR assessments. The results of this study indicate that temporal cycles may be used to characterize the effects of neurodegenerative disorders on speech communication. [Work supported by US NIA Grants Nos. R01-AG023195, P50-AG16574, P30-AG19610, and NIH NIA-1R01AG026390, and Univ. of MN Academic Health Center.]

4pSCb2. Transition characteristics in speakers with dysarthria and in healthy controls: Part IV: Additional data on vital capacity transitions and stroke patients. Gary Weismer, Christina Kuo, and Phoebe Allen (Dept. of Communicative Disord., Univ. of Wisconsin-Madison, 1975 Willow Dr., Madison, WI 53706, gweismer@wisc.edu)

Formant transitions are known to provide important cues for speech perception, sound identification, and inferences to articulatory behavior. This study describes and examines three types of formant transitions [consonant-vowel (CV), vowel-consonant (VC), and diphthong transitions] in four groups of speakers: healthy, ALS, Parkinson's disease, and stroke. This is an extension from previous work by Weismer *et al.* [J. Acoust. Soc. Am. **121**, 3135 (2007)] and Weismer *et al.* [J. Acoust. Soc. Am. **124**, 2558 (2008)], who showed shallower slopes for CV and diphthong transitions in persons with dysarthria (ALS and PD). To better understand the characteristics of the different transition types in healthy and disordered populations, two questions are addressed here. First, are CV transitions associated with dysarthria different from those in healthy speakers in a way comparable to the observed differences in diphthong transitions? Second, do VC transitions show the same normal characteristics, and are the differences between healthy speakers and speakers with dysarthria the same as in CV transitions? Distributional analyses for 50-ms CV, 50-ms VC, and diphthong (highest-lowest/lowest-highest) F2 transition measures will be presented for 18 healthy speakers, 4 speakers with ALS, 4 speakers with PD, and 20 stroke patients. [Work supported by NIDCD R01 DC003723.]

4pSCb3. Measuring speech and language characteristics of effects of medications on cognition. Serguei Pakhomov, Susan Marino, Angela Birnbaum, Chamika Hawkins-Taylor, and Ilo Leppik (Serguei Pakhomov, Univ. of Minnesota, Minneapolis, MN 55455, pakh0002@umn.edu)

Many of the drugs prescribed for various diseases and syndromes, including chronic pain and epilepsy, significantly impair cognitive functioning with large individual variation. Language is a highly individualized and directly observable product of human cognition that is central to our everyday functioning. A system for automated language and speech analysis (SALSA) was developed that relies on using an automatic speech recognition engine (HTK 3.4) [Young *et al.* (2008)] to perform forced-alignment between the audio of spontaneous speech samples and their transcripts to measure a number of speech and language characteristics including fluency, speaking rate, change in fundametal frequency, and information content of spontaneous narratives. The system was piloted on a population of 14 healthy volunteers who participated in a randomized, placebo-controlled study of cognitive effects of an anti-epileptic medication (topiramate). Our preliminary results suggest that SALSA captures a number of fluency and lexical characteristics sensitive to the effects of topiramate, thus providing an objective mechanism to quantify the degree of cognitive impairment in individulas affected by medications. Our current results are consistent with prior work investigating speech and language correlates of mild cognitive impairment [Roark *et al.* (2007)] and fronto-temporal dementia [Pakhomov *et al.* (2009)]. [Work supported by a grant from the Univ. of Minnesota Academic Health Crt.]

4pSCb4. The differential effect of altered auditory feedback on speech in early- and late-onset of Parkinson disease. Emily Q. Wang (Dept. of Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1653 West Congress Parkway, 203 SENN, Chicago, IL) and Leo Verhagen Metman (Rush Univ. Medical Ctr., Chicago, IL)

Advanced speech symptoms in Parkinson disease (PD), such as "festinating speech" or palilalia and frequent hesitations, are a clinical challenge. This report is a part of a larger study testing the hypothesis that the use of altered auditory feedback (AAF) would improve speech intelligibility in patients with PD and more advanced speech impairment. Our initial report showed that the use of AAF indeed improved speech intelligibility in these patient's spontaneous speech [Wang *et al.* (2008)]. In this report, the spontaneous speech samples produced by four older-onset (aged 72.4 SD 6.02 years) vs four young-onset PD patients (aged 54.2 SD 4.44 years), with and without the use of AAF, were analyzed. The results indicated that, regardless of the use of AAF, the older-onset patients performed poorer than the young-onset patients in many areas including the mean length of utterance in words, number of different word roots, within utterance pause time, and words per minute. These findings are consistent with the report that patients with an older age at onset had more rapid progression of PD than those with a younger age at onset in mentation, freezing, and activities of daily living. Possible underlying mechanism for the findings will be discussed.

4pSCb5. Temporal analysis of simultaneous nasopharyngoscopic and nasometric recordings. Wei Tian (Dept. of Hearing and Speech Sci., Univ. of Maryland, 0141D Lefrak Hall, College Park, MD 20742)

The velopharyngeal port closes and opens by the velar and pharyngeal wall motion during speech. Velopharyngeal dysfunction (failure to close or open appropriately) often involves insufficient motion or mistiming of the motion. Despite the fact that evaluation of the velopharyngeal port size by nasopharyngoscopy has been popular in clinical settings, the temporal pattern of velar and pharyngeal motion has never been studied. On the other hand, separate nasal and oral recording of speech acoustics with a Nasometer is reported to demonstrate speech nasality change in customized utterances. The present study aimed at investigating the correlations between velopharyngeal motion and speech acoustic change by simultaneous recordings of nasopharyngoscopy and nasometry. Kymographic analysis of the motion in the velum and lateral pharyngeal walls were performed in two female adult speakers during five repetitions of six VCVNVCV at a controlled speech rate. Although the velar and pharyngeal motions show the similar pattern, they do not initiate and end simultaneously. There is strong correlation between the physiologic motion and acoustic change, but they are not synchronized and have very different temporal patterns. There is also difference across speakers and phonemes.

THURSDAY AFTERNOON, 22 APRIL 2010 — ESSEX A/B/C, 1:30 TO 2:55 P.M.

Session 4pSP

Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics: Maximum Entropy and Bayesian Signal Processing II

Zoi-Heleni Michalopoulou, Cochair
New Jersey Inst. of Technology, Dept. of Mathematics, Newark, NJ 07102–1982

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

Invited Papers

1:30

4pSP1. Bayesian parameter estimation in adaptive psychometric procedures: Simulated and experimental results. Jeremiah J. Remus (Dept. of Elec. and Comput. Eng., Clarkson Univ., Box 5720, Potsdam, NY, 13699) and Leslie M. Collins (Duke Univ., Durham, NC 27708)

The wide use of psychometric assessments and the associated time required to conduct such experiments have motivated a substantial research effort focused on developing more efficient psychometric procedures to expedite the estimation of parameters of interest. Adaptive step size psychometric procedures, which have increasingly received attention as an alternative to fixed step size procedures, can be viewed as having two separate stages: estimation of the parameters of interest based on the available data and a selection of the experimental settings in order to maximize the information obtained from the next trial. Psychometric procedures that utilize a Bayesian framework for parameter estimation have become more prevalent in recent years. In this talk, we will present experimental results for Bayesian psychometric procedures from both psychoacoustics studies and computer simulations. Several computer simulations explored the sensitivity of the procedure to different model parametrizations and evaluated performance using stimulus selection rules based on minimizing entropy and maximizing information gain. Psychoacoustic studies using two different listening tasks were used to investigate the robustness and stability of estimates provided by the Bayesian parameter estimation procedure. Collectively, the outcomes suggest that a Bayesian framework may be an important component in the development of more efficient psychometric procedures.

1:50

4pSP2. Nested sampling for room-acoustics energy decay analysis. Ning Xiang, Tomislav Jasa, and Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Room-acoustic energy decays often exhibit single-rate or multiple-rate behavior in a wide variety of enclosures. There has been an ever increasing need to investigate where and under what conditions the single-rate and or multiple-rate energy decays occur and to quantify/characterize the energy decay process by estimating a set of decay parameters. In this architectural acoustics application, both energy decay model selection and decay parameter estimation are of practical significance. This paper discusses a model-based sound energy decay analysis within a Bayesian framework and applies the most recent Bayesian sampling method (nested sampling) to sound energy decay analysis. The nested sampling reverses the historical Bayesian computation approaches, yielding the Bayesian evidence as the prime target with representative posterior samples available as optional by-products. Bayesian evidence is crucial for model selection, while the representative posterior samples are of central importance for parameter estimation. Taking the energy decay analysis in architectural acoustics as an example, this paper demonstrates that two different levels of inference, decay model-selection and decay parameter estimation, can be cohesively accomplished by nested sampling.

Contributed Papers

2:10

4pSP3. Comparison of fading statistics for shallow and deep acoustic sources in a continental shelf environment. Alexander W. Sell, R. Lee Culver, Colin W. Jemmott, and Brett E. Bissinger (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, aws164@psu.edu)

Previous work has shown a noticeable difference in the effects of the ocean environment on signals from shallow and deep moving sources. These effects are seen in received amplitude statistics, and such statistics can be used in passive acoustic depth classification. This talk presents a statistical analysis of signals from a September 2007 shallow water acoustic transmission test performed along the continental shelf off the coast of southeast Florida. The data used include low frequency (between 25 and 450 Hz), continuous-wave signals from a towed source at 100 m depth, as well as tones from surface ships in the area. Using statistical class models in a Minimum Hellinger Distance Classifier, the usefulness of received signal amplitude statistics for passive acoustic source-depth classification is discussed. [Work supported by ONR Undersea Signal Processing.]

2:25

4pSP4. Depth discrimination in shallow water by matched-field tracking. Donald R. DelBalzo and James H. Leclere (QinetiQ North America, Hwy. 190 East, Slidell, LA 70461, donald.delbalzo@qinetiq-na.com)

Passive acoustic sonars have difficulty in detecting quiet sources in noisy shallow-water environments. The standard approach to improving detection performance 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 apply matched-field correlation techniques and 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 potential applications preclude the use of vertical arrays for practical reasons, like damage caused by fish trawling. In this work, we examine the utility of matched-field processing on linear and planar horizontal arrays on the ocean bottom in shallow water. We employ a track-before-detect strategy to convert matched-field localizations into submerged source detections in various physical and acoustic environments. We show the advantages of planar compared to linear arrays to discriminate submerged from surface sources as a function of SNR.

2:40

4pSP5. A feasibility study on the low probability of intercept sonar. J. Daniel Park, David J. Miller, John F. Doherty (Dept. of Elec. Eng., The Penn State Univ., University Park, PA 16802), and Stephen C. Thompson (The Penn State Univ., State College, PA 16804)

The feasibility of low probability of intercept for sonar is explored. Using a noise-like active sonar signal, the transmitter (platform) employs a matched filter for echo detection while the target is assumed to use an energy detector. Decision statistic distributions are developed and detection performances are compared with a previous work (Gaussian distribution assumed). We then explore the detection advantage the platform can achieve by evasive on-off keying and by optimization of transmitted power. A favorable operating region of the platform in the (low power, small range) region of the (range, power) plane is identified. This suggests that the platform should start detection using a low-power probing signal, increasing power until a reliable detection rate is first achieved while ensuring that the detection rate at the target does not exceed a specified level. Second part of the feasibility study is about covert range estimation. Two matched-filter based algorithms were developed for the platform to obtain credible range estimations while ensuring the target fails to detect the platform. A platform-target encounter scenario was designed for detailed waveform-based Monte Carlo simulation. Characterization of variables in the model was performed to show the feasible conditions of covert range estimation. [Work sponsored by ONR, ULI.]

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

P. Battenberg, Chair, ASC S1
Quest Technologies, Inc., 1060 Corporate Center Dr., Oconomowoc, WI 53066 4828

R. J. Peppin, Vice Chair, ASC S1
Scantek, Inc., 6450 Dobbin Rd., #A, 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 8:00 a.m. on Thursday, 22 April 2010.

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.

THURSDAY AFTERNOON, 22 APRIL 2010 KENT A/B, 3:00 TO 4:15 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C. A. Champlin, Chair, ASC S3
University of Texas, Dept. of Communication Sciences & Disorders, CMA 2-200, Austin, TX 78712

D. A. Preves, Vice Chair, ASC S3
Starkey Laboratories, Inc., 6600 Washington Ave., S., Eden Prairie, MN 55344

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 8:00 a.m. on Thursday, 22 April 2010.

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.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. K. Delaney, Chair, ASC S3/SC 1
USA CERL, 2902 Farber Dr., Champaign, IL 61822

M. C. Hastings, Vice Chair, ASC S3/SC 1
6422 Crosswoods Dr., Falls Church VA 22044 1214

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/SC1: 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.

THURSDAY EVENING, 22 APRIL 2010 7:30 P.M.

OPEN MEETINGS OF ASA 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 immediately after the Social Hours. On Wednesday, one technical committee 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 discussions.

Committees meeting on Thursday are as follows:

Animal Bioacoustics	Grand Ballroom I/II
Architectural Acoustics	Grand Ballroom III/IV
Biomedical Ultrasound/Bioresponse to Vibration	Kent A/B
Speech Communication	Laurel A/B
Underwater Acoustics	Laurel C/D