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@rgpin.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-echo 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 92039, 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 multipurpose facility, and Cornea, a 10 × 10 × 10 in.6 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.

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@cavtoci.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 m³) present unique challenges compared to smaller, traditional practice rooms (6–12 m³) 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 (5.3×5.7×2.7 m³, 81 m³) 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

4A AA A5. 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


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

4A AA A7. 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

4A AA A8. 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

4A AA A9. Obtaining a frequency independent reverberation time in listening rooms. Niels Werner Adelman-Larsen (Flex Acoust., SCION-DTU, Diplomevj 377, 2800 Kgs. Lyngby, Denmark, nw1@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

4aAAAb1. 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

4aAAAb2. 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

4aAAAb3. 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

4aAAAb4. 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


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

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.
10:15


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
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 L. 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 Ocean Acoustics.]

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 extends 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.]


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.]
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


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.
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 92039-0238, 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-array, 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.

Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound Induced Cellular Bioeffects

E. Carr Everbach, Chair
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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 using low-intensity focused ultrasound bursts and microbubbles to temporally 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 agents, perhaps due to differences in vasoreactive 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.]
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 >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) = CO2(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.

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.

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 was placed, attached to microprobes, 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. 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 phenoxyphrine and acetylcholine indicates smooth muscle and endothelial cell damage. This has been confirmed by histology 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.
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

11:05

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 pPIV 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

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 growth of the fungus. This paper describes the device, which combines ultrasound and drug delivery systems, and presents preliminary results. 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

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


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 enhancement of sound image localization on flat display panels. Gabriel Pablo Nava, Keiji Hirata, and Yoshinari Shirai (NTT Commun. Sci. Labs., NTT Co., Hirakaidai 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 performed 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.

9:05

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.

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 Zg of the propagation medium is converted to a higher impedance at the PZT surface by the relation Zm = Zg/ZR (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 Zmax = Zg/ZR 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 ob-
served 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 Ogha (Ogha’s Acoust. Lab., 2-24-3, Tamanawa, Kamakura, 247-0071, Japan), Takehiko Adachi, Hiroki Saito, Ryoysuke Suzuki, Gen Takeda, Hajime Kubota (Chiba Inst. of Technol., Narashino 275-0016, Japan), Hirokazu Negishi (MIX Acons. Lab., Shiba, Minato-ku, Tokyo 105-0014, Japan), Kazuaki Maeda (TOA Corp., Takarazuka 665-0043, Japan), and Ikuko 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 force. 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

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 magnetostriuctive material. In addition to possessing competitive strain capabilities, strong magnetic 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 such 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

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_31 > 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_{P1} = 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_{0.5}Nb_{0.5})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+}, 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

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 ABAQUS, 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, lbbaumgar@usc.edu) and Eun Sok Kim (USC MEMS Res. Group, Los Angeles, CA 90089)

A micromachined piezoelectric speaker was fabricated on a 5 × 5-mm², 1-µm-thick silicon nitride diaphragm. A 4 × 4-mm² 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\text{rms} from 40 Hz to 4 kHz, demonstrating a response of 55 nm/V\text{rms} and an average THD of 5.1\%. The unwrinkled device is driven at 15 V\text{rms} over the same range, yielding a response of 5 nm/V\text{rms} 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. Intergated 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.]

Session 4aED

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

J. Arvelo, Cochair

P. Wilson, Cochair
Univ. of Texas at Austin, Dept. of Mechanical Engineering, J 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
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.

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 faculty. 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.

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


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.
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 globalization 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.

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.

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.

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 linear 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 while 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**

**Session 4aPAb**

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

Albert Migliori, Chair

*Los Alamos National Lab., MS E336, Los Alamos, NM 87545*

**Chair’s Introduction—10:00**

**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 microscopy and transmission electron microscopy. 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.]
10:45

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


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 10^3–10^6/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


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

Xiaqin 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, Xiaqin 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.
09:00


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.]

09:20

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.

09:40


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.]

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


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.
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 excitation. 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

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 $m/3$ and $10m/3$, respectively. Results for the three solution methods are compared. [Work partly supported by ONPP 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 [Ali 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 $n_0$ cylinders/m$^3$ 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 $n_0$, 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, Guillelmette Ribay, and Julien de Rosny (Laboratoire de Gophysique Interne et Tectonophysique (LGIT), CNRS Université 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 parallelepipedic 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 $a^{1/2}d^d$, $d$ 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.
Digitized bits, representing the value of an analog signal measured on the inside transducer, are powered by harvesting the electrical output of the inside transducer. This wave generates an acoustic field. The inside electronics contains a continuous wave generating an acoustic field. The inside electronics includes 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 PSPOC. Simulation, finite element solutions, and experimental results are presented and discussed. Digital data communication rates exceeding 50 000 bits/s are achieved.

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 0QJ, 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 not 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

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 signal measured on the outside (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 PSPOC. Simulation, finite element solutions, and experimental results are presented and discussed. Digital data communication rates exceeding 50 000 bits/s are achieved.
Our previous studies have shown that English-native adult speakers of Spanish can be trained to perceive and produce the intervocalic /d/, /t/ and /tr/ 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@lumnet.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 syllables 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 co- das 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 [ɾ] in onset. Intervocally, they may appear as contrasting as tap or geminate lateral ([l][ɾ] wolf vs [ll][ɾ][ɾ] reason), differing in duration and laterality. Kim ([2007]) demonstrated that Korean listeners identified a shortened geminate lateral as geminate /l/ 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 [l] versus [ɾ] is motivated by the native language acoustics. Korean speakers produced 24 Korean words [(C)V_V], containing [l] or [ɾ] in two
speech modes (in a carrier sentence vs in isolation). The duration of [l] 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 [l] generally, but it showed a greater variability. However, F3 at the offset of the preceding vowel was lower for tap than for [l] 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 [l]/[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 CsCv sequences were harder 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/CVC/ and /CVCVC/, where CC was 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 (r = + 0.715, p<0.01). However, the major error type was vowel deletion, rather than vowel epenthesis in producing the CVC 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 Wil- low 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 uvu- lar 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 /t/ 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 Dwight 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 [e] sounds like [a], 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, sub-

jects whose [i] is heard as [i], a shift primarily in F1, compensated by producing a higher F2, while subjects whose central vowel [a] was heard as [e] 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 Lemaire, Arnaud Dessein (IRCAM, 1 place Stravinsky, 75004 Paris, France), Karine Aura (Université 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 hierarchi-
cal 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 per-
fectly 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. Seunghan Yang, Ji Sook Ahn, and Diana Van Lancker Sidtis (Dept. of Communicative Science & Disorder, 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 presenta-
tion of these utterances singly and in pairs without other context and iden-
tified 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 ana-
yzed 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-Burah 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 Disorder 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 [Sensometrics Corporation (1995)]. Each listener heard 180 trials of a “change-no-change” discrimination paradigm with two, three, four, and
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 vocal 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.


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., /p européen/) 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., /bzh unpleasant/) and CCCC clusters (e.g., /vzb unpleasant/) 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, prodorig@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 (e.g., 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 neutral 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. 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 pro-
duced 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, tamatit@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 (either the US south and east or west). This suggests the 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. Educ. Cmtr., 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 CV/CV frames in which the syllable final segment was a consonant with airstream blocking, and native speakers of Japanese produced more nasal /n/ and obstruct /q/ in CVNVCV/ and CVQCVCV/ frames. In Japanese stimuli, the consonant after the nasal or the obstruct 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 [Morrelli, (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 na{"i}ve Greek adult listeners identified syllables beginning with /spzl, /stl, /slkl, /pszl, /tszl, or /ths/ 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 /pszl/ and /kszl/ clusters than the /tszl/ and /s/-stop clusters, particularly in the ~6 and 0 SNRs. There was also a significant interaction between vowel, such that /pszl/ and /ths/ were identified more accurately before /a/, whereas /kszl/ 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 Mandarin learners. The goal is to show that this methodology is 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 among levels of proficiency and that the level and type of L2 exposure are 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 psycholinguistic 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 psycholinguistic 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 real-world and nonce 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 vocal contrast 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 [Nittouer 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 vocoid context—for each subject. Every pattern of cue used 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 contrasted stop release (e.g., [bɛt] = BET, [bɛt] = 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 “crocodile” and “crocodilo” 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 differences during the perception of accented speech? Rachel M. Miller, Kaumari Sanchez, Lawrence D. Rosenblum, James W. Dias, and Neal Dykmans (Dept. of Psych., Univ. of California, 900 Univ Ave., Riverside, CA 92521, rmill@ucr.edu@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. Listeners 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 idiolectic 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@uw.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.]
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


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.
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 side lobes, as the 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


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, 3911 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.]
Underwater Acoustics: Propagation and Scattering in Heterogeneous 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 Office of Naval Research.]

8:15

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@noddyl.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/UA, 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 algorithm as an alternative to ray tracing to solve for the acoustical 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.]
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 non-perturbative transform presented here broadly generalizes that of Dozier one-dimensional, rough sea surface onto a domain with a flat top. The non-rough and periodic sea surfaces.

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.]

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.

A conformal transform is presented here that maps an acoustic domain with a one-dimensional, rough sea surface onto a domain with a flat top. The non-perturbative transform presented here broadly generalizes that of Dozier [J. Acoust. Soc. Am. 75, 1415–1423 (1984)] to include many wavelengths of the surface variation. A two-dimensional, flat-top domain permits the direct application of a parabolic equation model as 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.]

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.

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, 0.001 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, 2.0 Hz, coherence parameters depend on mode number, with the lower order modes always more coherent than successive higher order modes. At still higher frequencies, f>100 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 coherency is already degraded by boundary roughness so that the slightest of internal wave activity completely randomizes the signals.

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.]
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 conventional 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.

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
Schomer and Associates, 2117 Robert Dr., Champaign, IL 61821

D. J. Evans, Chair
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
as applied to machines, vehicles and structures
BP America, 501 Westlake Park Blvd., Houston, TX 77079

R. Taddeo, Co-Chair
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
701 NE Harbour Terrace, Boca Raton, FL 33431

R. Taddeo, Vice Chair
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

C. Peterson, Chair
200 Dixie Ave., Kalamazoo, MI 49001

V. Nedzelnitsky
U.S. Technical Advisor (TA) for IEC/TC 29 Electroacoustics
National Institute of Standards and Technology (NIST), 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:

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

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:

<table>
<thead>
<tr>
<th>U.S. TAG Chair/Vice Chair</th>
<th>TC or SC</th>
<th>U.S. Parallel Committee</th>
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<tr>
<td><strong>ISO</strong></td>
<td></td>
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<tr>
<td>P. D. Schomer, Chair</td>
<td>ISO/TC 43 Acoustics</td>
<td>ASC S1 and S3</td>
</tr>
<tr>
<td>P. D. Schomer, Chair</td>
<td>ISO/TC 43/SC1 Noise</td>
<td>ASC S12</td>
</tr>
<tr>
<td>D. J. Evans, Chair</td>
<td>ISO/TC 108 Mechanical vibration, shock and condition monitoring</td>
<td>ASC S2</td>
</tr>
<tr>
<td>W. C. Foiles, Co-Chair</td>
<td>ISO/TC 108/SC2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures</td>
<td>ASC S2</td>
</tr>
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</tr>
<tr>
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<td>ASC S2/S3</td>
</tr>
<tr>
<td>D. D. Reynolds, Chair</td>
<td>ISO/TC 108/SC5 Condition monitoring and diagnostics of machines</td>
<td>ASC S2</td>
</tr>
<tr>
<td>D. J. Vendittis, Chair</td>
<td>ISO/TC 108/SC6 Vibration and shock generating systems</td>
<td>ASC S2</td>
</tr>
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<td><strong>IEC</strong></td>
<td>IEC/TC 29 Electroacoustics</td>
<td>ASC S1 and S3</td>
</tr>
</tbody>
</table>
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.

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.