Session 1aAA

Architectural Acoustics: Design, Simulation, and Perception of Architectural Acoustics

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Contributed Papers

8:00

1aAA1. Sound absorption of different green roof systems. Ilaria Pittaluga, Corrado Schenone, and Davide Borelli (DIPTEM, Univ. of Genova, Via all'Opera Pia 15/A, I 16145, Genova, Italy)

Experimental data on acoustical performances, in particular on sound absorption, of several green roof systems were evaluated and discussed. Measurements were performed on samples of three green roof systems, different for maintenance, plant setting and containment criteria, and categorized in extensive green roof (sample A), semi-intensive green roof (sample B), and common soil (sample C). Experimental values of normal incidence acoustic absorption coefficient and acoustic impedance were evaluated for each sample in one-third octave frequency bands from 160 to 1600 Hz by using a standing wave tube. Then, diffusive sound absorption coefficients and normal and diffusive weighted sound absorption coefficients were calculated in the same frequency range. Results show that green roofs provide high sound absorption, mostly if compared with the typical performances of traditional flat roofs. Curves of sound absorption coefficients result strongly dependent on the stratigraphy. Comparison between the different systems performed on the base of weighted sound absorption coefficients shows a better behavior for the sample B. Results obtained suggest that green roof technology, in addition to energy and environmental benefits, can contribute to noise control in urban areas by means of high sound absorption performances in relation to the size of the surface area.

8:15

1aAA2. Acoustic properties of green walls with and without vegetation. V. Kirill Horoshenkov, Amir Khan, Hadj Benkreira (School of Eng., Univ. of Bradford, Bradford, West Yorkshire, BD7 1DP, United Kingdom), Agnès Mandon, and Rene Rohr (Canevaflor, 24, Rue du Docteur Guffon, 69170 Tarare, France)

One substantial issue with the majority of modern methods for noise control is their heavy reliance on man-made acoustic structures which require continuous service and maintenance. In this respect, the use of the inherent noise control properties of vegetation appears particular attractive compared to other street/square treatments for reducing noise such as adding façade absorption and diffusion. A green wall with a carefully selected type of soil substrate provides an alternative to more conventional types of acoustic treatment. This work studies the influence of leaves (foliage) on acoustic absorption of soils, plants, and their combination which are typically used in green (living) walls. It is shown the the presence of plans with a particular type of leaves can result in a considerable (up to 50%) improvement in the absorption coefficient of a green wall with soil at a certain water saturation. The acoustic absorption coefficient of these systems is examined here through laboratory measurements and theoretical prediction models. The plants in this study were chosen to cover a range of possible leaf types, sizes, and densities.

8:30

1aAA3. Analysis of sound propagation in an experimental model using a high resolution scanning system. Aditya Alamuru, Ning Xiang, and Joonhee Lee (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

The aim of this work is to analyze low frequency and mid frequency sound propagation in a coupled volume system for different aperture configurations. The cross sectional areas of the primary room and the secondary room are scanned over high spatial resolution grids using an automated scanning system. This procedure is carried out systematically for different aperture configurations. A dedicated analysis algorithm converts experimentally measured room impulse response data at each grid point into energy distributions and instantaneous sound pressure levels as a function of time. The analysis algorithm provides streams of data for a selected frequency thereby creating an animation of sound propagating through the coupled volume system. This work will demonstrate wave phenomena for different aperture configurations at low frequencies and mid frequencies by using animations to analyze the experimentally measured data.

8:45

1aAA4. Architectural acoustic elements to reduce the decay time in a room. Bonnie Schnitta, Melissa Russo, Greg Greenwald, and Michael Cain (SoundSense, LLC, 46 Newtown Ln., Ste. One, E. Hampton, NY, 11937, bonnie@soundsense.com)

The paradise architectural acoustic devices are sound modifying architectural elements, or structures. The architectural structure can be made to look like any one of the standard architectural structures commonly used in a room, from baseboards to crown moldings or ceiling beams. Depending on the intended outcome, the architectural elements can either be a solid body, or what appears to be a solid element but has internal mathematically determined channels. This allows the architectural elements, or devices, to not only reduce or correct the decay time in the room, but also make certain that the architectural elements do not produce undesirable effects. The mathematical foundation of the various shapes and designs of the architectural acoustic structures, or elements, will be presented for a linear case. This will allow a better understanding of the underlining acoustical effects. An appreciation for a more precise mathematical description of the embodiments will also be discussed by additionally taking into account the nonlinear aspects of the various embodiments. Various views of an acoustic architectural device for reducing or correcting decay time will be presented. Additionally, the improvements to the acoustic environment of the room that result from the paradise architectural elements will be provided.

9:00

1aAA5. Analyzing the auditory nature of architecture. Daniel Butko (College of Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

Architects usually place primary focus on the physical nature of materials and the aesthetic qualities of how those various materials are assembled. Although visual appeal is quite valuable when creating the built environment and specific inhabitable space(s), attention to sound is also vital in making a project successful for its intended use and daily occupants. Third year architecture students at the University of Oklahoma were tasked with a series of analytical studies and experiments that focused on the auditory nature of the built environment. This served as a precursor to the semester design project. The students spent time listening and discovering what sounds and noises were present within various functions. Each student categorized their individual findings of analyzed space into three classifications: (1) What sounds and/or frequencies they deemed successful/helpful to the intended use, (2) What sounds and/or frequencies they deemed unsuccessful/hindrance to the intended use, and (3) What additional sounds or noises they believe could have supported the intended use. The results were presented in class, discussed among the entire group, and ultimately fueled the students' future design decisions. This paper focuses on how simple analytical auditory studies can alter the overall design process to include architectural acoustics.

9:15

1aAA6. Acoustic simulation of renaissance Venetian churches. Braxton B. Boren (Music and Audio Res. Lab., New York Univ., 35 W. 4th St. New York, NY 10012, bbboren@gmail.com) and Malcolm S. Longair (Univ. of Cambridge, Cambridge, CB3 0HE, United Kingdom)

The Venetian Renaissance was a confluence of innovative expression across many artistic disciplines. While architects like Palladio and Sansovino were designing architectural masterpieces in many of the churches built during this period, composers such as Willaert, the Gabrielli, and Monteverdi were composing complex polyphonic works for split-choir ensembles, exploring the tonal and spatial dimensions of musical performance. The large churches built during this period have extremely long reverberation times and provide low clarity for understanding the complex polyphony composed for these spaces. This paper uses modern acoustic simulation techniques to provide insights into the acoustics of large Venetian churches as they would have existed during the Renaissance. In consultation with architectural historians, the authors have collected data on the structure and layout of Palladio's Redentore and San Marco on festal occasions, when large crowds, extra seating, and wall tapestries would have provided extra absorption. Using Odeon, acoustic simulations predict that under festal conditions these churches would have had significant improvements in T30, EDT, and C80. The doge's position in San Marco's chancel has particularly good clarity for sources located in Sansovino's galleries, supporting historian Laura Moretti's hypothesis that these galleries were installed for the performance of split-choir music.

9:30

1aAA7. Temporal modeling of measurement conditions to enhance room acoustical parameter characterization. Stephen Roessner (Dept. of Elec. and Comput. Eng., Edmund A. Hajim School of Eng. and Appl. Sci., Univ. of Rochester, NY 14627), Gang Ren, Mark F. Bocko, and Dave Headlam (Univ. of Rochester, NY 14627)

Forming statistical combinations of the results of repeated acoustical measurements taken under identical conditions is a common practice to reduce the effects of random noise. The most common method is to calculate the arithmetic mean of an ensemble of test results, which is based on the assumption that all experiments were conducted under identical noiseless test conditions. For most room acoustic measurement scenarios, this assumption is not valid, and non-stationary sources of noise often contaminate the results. Traditional statistical averaging methods can be improved by explicitly modeling the ambient interference and noise. Using a signal model for the noise and interference, the proposed parameter estimation procedure provides more accurate results than simple averaging in low signal to noise ratio test scenarios. This method, in which multiple, low volume measurements replace high volume test signals, provides a practical and cost-effective approach for characterizing acoustical spaces.

9:45

1aAA8. Benchmark measurements of noise, reverberation time, and an estimate of speech intelligibility in a representative operating room at Nationwide Children's Hospital in Columbus, Ohio. Richard D. Godfrey, Lawrence L. Feth (Dept. of Speech and Hearing Sci. The Ohio State Univ., 1070 Carmack Rd., Columbus, OH, godfrey.20@osu.edu), and Peter Winch (Nationwide Children's Hospital, Columbus, OH 43205)

Nationwide Children's Hospital was interested in understanding speech communications: in their operating rooms and between the parents/child and the doctor in pre-operative rooms. Long-term hearing loss of the staff was of secondary interest. Before a comprehensive project was proposed, data in a single OR to gain some experience was conducted. A SLM was programmed to measure the following during 15 s intervals: overall A-weighted equivalent energy sound level, A-weighted equivalent energy sound level in octave bands from 16 to 16 kHz, and peak un-weighted level during the interval. Reverberation was also measured by an impulsive method. Measurements were made for 23 consecutive hours. The data were downloaded for analysis. It was concluded that (1) adding some absorption around the top of the walls would improve SI, (2) good SI is only possible with a high vocal effort, and (3) long term hearing loss is very unlikely. Follow up topics before a comprehensive project is proposed were (1) try other reverberation methods, (2) study more rooms while a variety of surgical procedures are performed, (3) identify the source and duration of peaks levels, and (4) investigate other measures of SI.

10:00

1aAA9. Impacts of classroom acoustics on elementary student achievement. Lauren M. Ronsse and Lily M. Wang (Architectural Eng. Prog., Univ. of Nebraska-Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0861)

This research investigates relationships between unoccupied classroom acoustical conditions and elementary student achievement. Acoustical measurements were gathered in all of the third and fifth-grade school classrooms (67 total) in a public school district in north-eastern Nebraska, USA. Traditional classroom acoustic parameters, including background noise level and reverberation time, have been correlated to the standardized achievement test scores from students in the surveyed classrooms. Binaural impulse response measurements were also conducted in a subset of the rooms (20 total) and correlated to the student achievement scores. Acoustical metrics calculated from the binaural impulse response measurements include speech transmission index, distortion of frequency-smoothed magnitude, interaural cross-correlations, and interaural level differences. The results from this research indicate that scores on fifth-grade student language and reading subject areas are negatively correlated to higher unoccupied background noise levels. Also, the distortion of frequency-smoothed magnitude, which is a perception-based acoustics metric, was significantly related to the student language achievement test scores.

10:15–10:30 Break

10:30

1aAA10. A survey of residential "speaking tubes." William J. Elliot and John T. Foulkes (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, welliot@cavtocci.com)

The speech transmission index is examined for a system of "speaking tubes" within a home in the Avon Hill neighborhood in Cambridge, Massachusetts. Designed in 1888 by noted Boston architects Henry Hartwell and William C. Richardson, the Shingle Style home was both wired for electricity and outfitted with piping for gas. An electronic paging system was used to summon servants, but when aural communication was necessary, the speaking tube system was used for inter-floor communication. This paper examines the measured STI for the speaking tubes which remain in the home within the context of simple passive waveguide sound propagation. The paper also provides a quantitative evaluation of this pre-electroacoustic technology as it appeared in several fashionable homes of the late 19th and early 20th centuries.

10:45

1aAA11. Comparison of the articulation test and speech transmission index values measured in two different acoustical poor conditions. Jorge Sommerhoff (Acoust. Inst., Universidad Austral de Chile, Independencia 641, Valdivia, Chile, jsommerh@uach.cl) and Claudia Rosas (Universidad Austral, Valdivia, Chile)

In this work, a comparison is made between articulation tests and speech transmission index (STI) values, measured in the same place, in two different acoustically poor room conditions, both with STI values of less than 0.4. In the first room condition, a 200 m^3 reverberation chamber with background noise of less than 32 dBA was used. In the second room condition, pink noise was added to a small room with reverberation time of less than 0.6 s, till values of STI lower than 0.4 were reached. The articulation test corpus consisted in a 1000 phonetically Spanish combination of a consonant, vowel, and consonant (CVC logatoms). The logatoms were recorded in an anechoic chamber. In the articulation test and STI measurement, both signals were emitted in the rooms using a NTI Talkbox with a sound power equivalent to a normal human voice. The STI and articulation was measured at the listener's seats which were located at different distances from the source but within STI values less than 0.4. The articulation test results of both acoustical conditions are correlated separately with the measured STI. The results of the measurements indicate that for the same STI value, the subjective response statistically differs.

11:00

1aAA12. The effect of listener head movement on perceived envelopment and apparent source width. Anthony J. Parks (Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180)

Current research examining listeners' perceived spatial impression of a concert hall relies on a fixed-head worldview, since the overwhelming majority of listening tests conducted to determine subjective spaciousness [listener envelopment (LEV) and apparent source width (ASW)] has required listeners to keep their heads fixed. Such a worldview is an incomplete one, because listeners make noticeable exploratory head movements while evaluating sonic environments, including the more common task of source localization as well as the more involved task of evaluating the spaciousness of a concert hall. This study investigates the role of listener head movement in the evaluation of perceived LEV and ASW under 15 different concert hall conditions simulated over eight loudspeakers using Virtual Microphone Control. The conditions consist of both varying ratios of front-to-back energy and varying levels of cross-correlated reverberant energy. Head movements are monitored in terms of angular rotation (azimuth, elevation, and roll) using a head tracker while listeners are prompted to give subjective ratings of LEV and ASW ranging from 1 (least) to 7 (most). The listening tests are then repeated while subjects are asked to keep their heads fixed. The head movements are analyzed and results of the tests are compared.

11:15

1aAA13. Applications of a binaural model with contralateral inhibition in room acoustics analysis. Timothy Perez, Jonas Braasch, and Ning Xiang (Grad. Prog. in Arch. Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, perezt2@rpi.edu)

In many cases the conventional, monophonic measures used in analyzing room acoustics show little agreement with real listeners' responses; a binaural perspective provides much-needed spatial and perceptual information that is important in acoustical quality judgments. Traditionally, binaural models extracted information about location and intensity of sound through a relatively simple cross-correlation procedure. This paradigm was extended by Lindemann [J. Acoust. Soc. Am. **80**, 1608–1630 (1986)] with the introduction of a contralateral inhibition process, which applies a time- and intensity-dependent weighting to a pair of binaural signals and accurately reproduces results from psychoacoustical tests where the traditional model fails, such as in the case of the Precedence effect. The model will be used to observe the broadening and splitting of auditory events based on the degree of interaural coherence, providing further validation of its adequacy. Then, applications in architectural acoustics will be investigated by processing binaural room impulse responses and producing a binaural activity pattern, which indicates the location and spatial extent of the resulting auditory events. These will be compared to visualizations drawn from spherical harmonic microphones. Implications of such a model on factors important in acoustical quality assessments, such as apparent source width and listener envelopment, will be discussed.

11:30

1aAA14. The audibility of direct sound as a key to measuring the clarity of speech and music. David H. Griesinger (David Griesinger Acoust., 221 Mt. Auburn St., Cambridge, MA 02138, dgriesinger@verizon.net)

Human ear/brain systems evolved to decode the direction, timbre, and distance of multiple sounds in a complex and noisy environment. In a reverberant space, this information is only available at the onset of a sound, before reflections overwhelm it. But since the time of Sabine acoustic science has concentrated on the decay of sound in a reverberant field, not on the audibility of the onset information. In addition, it is well known that the ability to separate multiple sound sources depends critically on pitch, but acoustic research studies only noise and impulses. This paper proposes that clarity requires the ability to separately analyze multiple sounds (the cocktail party effect) and that the cocktail party effect depends on phase relationships between harmonics of complex tones. These phase relationships are scrambled in predicable ways by reflections and reverberation. Well known properties of human hearing are used to develop both a physical model for the neurology of onset detection and an impulse response measure for localization and clarity in a reverberant field. A C language implementation of the physical model is capable of predicting and perhaps measuring the localizability of individual musicians in a binaural recording of live music.

11:45

1aAA15. Sound metric design based on psychological and physiological acoustics for the analysis of automotive sound. Young Joon Lee, Hong Sug Park, and Sang Kwon Lee (Dept. of Mech. Eng., Inha Univ., 253 Yonghyum Dong, In cheon 402-751, Korea)

This paper presents the correlation between psychological and physiological acoustics for the automotive sound. The research purpose of this paper is to evaluate the sound quality of interior sound of a passenger car based human sensibility. The conventional method for the objective evaluation of sound quality is to use the only sound metrics based on psychological acoustics. This method used not only psychological acoustics, but also physiological acoustics. For this work, the sounds of five premium passenger cars are used for the subjective evaluation. The correlation between this subjective rating and sound metrics based on psychological acoustics is calculated. Finally, the correlated sound metric is used for calculating the correlation between sound metric and the electron cephalogram signal measured on the brain. Throughout these results, the new evaluation system for the sound quality on interior sound of a passenger car has been developed.

Session 1aAB

Animal Bioacoustics: Acoustic Ecology

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Thomas Norris, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

1aAB1. Marine acoustic ecologies and acoustic habitats: Concepts, metrics, and realities. Christopher W. Clark, Aaron N. Rice, Dimitri W. Ponirakis, and Peter J. Dugan (Bioacoustics Res. Prog., Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, cwc2@cornell.edu)

Whales, dolphins, and porpoises (cetaceans) are adapted to produce and perceive sounds that collectively span 4–6 orders of magnitude along space, time, and frequency dimensions. Two important concepts, acoustic ecology and acoustic habitat, emerge from this perspective: where acoustic ecology is the study of acoustics involved in interactions of living organisms, and acoustic habitat as ecological space acoustically utilized by particular species. Cetaceans are dependent on access to their normal acoustic habitats for basic life functions. Communication masking from anthropogenic sounds that are chronically present can result in measurable losses of cetacean acoustic habitats, especially for low-frequency specialists, baleen whales. A communication masking model, informed by multi-year datasets, demonstrates cumulative influences of multiple vessels on fin, humpback and right whale acoustic habitats at spatial, temporal, and spectral scales matched to ecologically meaningful habitats. Results quantify acoustic habitat spatio-temporal variability over ecologically meaningful scales. In some habitats with high vessel traffic and vessel noise, predicted habitat loss and area over which animals can communicate is dramatically reduced compared to what it would be under non-vessel conditions. From a large-scale, ecological perspective, these acoustic habitat reductions likely represent significant costs for species for which acoustic communication is biologically critical.

8:30

1aAB2. Evaluating the potential spatial extent of chronic noise exposures of sufficient magnitude to raise concerns of wildlife impacts. Kurt M. Fristrup (Natural Sound and Night Sky Div., Natl. Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

Noise is probably the fastest growing pollutant in the United States. Traffic levels for many forms of transportation and recreation are increasing in much faster rates than population size. The consequences of chronic noise exposure for natural ecosystems are numerous and potentially severe. Decreases in pairing success, recruitment, population density, and community diversity have been documented for a variety of taxa. This presentation provide a capsule summary of documented biological impacts. These findings will be used to interpret the results from acoustical monitoring in a variety of National Park units, as well as predictions from noise models.

Contributed Papers

8:55

1aAB3. Endangered Stephen's kangaroo rats respond to road noise with footdrumming. Debra M. Shier, Amanda J. Lea, and Megan A. Owen (San Diego Zoo Global Inst. for Conservation Res., 15600 San Pasqual Valley Rd., Escondido, CA 92027)

On-road vehicles have become a pervasive source of low frequency noise in both urban and protected areas. Because many species rely on low-frequency signals to communicate, they are likely vulnerable to signal masking and other adverse effects of road noise exposure. We recorded and quantified both road noise and low frequency footdrumming signals from endangered Stephen's kangaroo rat (*Dipodomys stephensi*; SKR), and found the two signals to overlap extensively. We then played back footdrumming overlaid with experimental (road noise), as well as positive (crickets) and negative control (no noise) sounds to SKR. SKR showed no response to footdrumming playbacks overlaid with road noise, suggesting that noise may mask conspecific signals. Furthermore, playbacks of road noise alone provoked similar behavioral responses to those of footdrumming controls. It appears that road noise itself may mimic footdrumming and prompt a false response in SKR. Therefore, anthropogenic noise may not only mask signaling, it may also function as a deceptive signal to wildlife. For SKR, the combined effects of communication disruption and signal deception may further tax already endangered populations. Road margins serve as dispersal corridors and refugia for SKR, yet these areas may function as ecological traps if anthropogenic noise negatively affects populations.

9:10

1aAB4. Influence of impulsive sources on the soundscape of under-ice Arctic marine mammals. Juan I. Arvelo, Jr. (Natl. Security Technol. Dept., The Johns Hopkins Univ. Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099, juan.arvelo@jhuapl.edu)

The Arctic Ocean exemplifies the danger in using sound propagation and interaction models without a clear understanding of the physics of the problem or weaknesses inherent in these models. The zone-of-influence for under-ice marine mammals should be expected to differ significantly from that of open-sea organisms. However, increasing the effective sound attenuation, to empirically account for sound interactions with keel drafts, may lead to grossly erroneous conclusions from predictive and forensic studies. Ice elasticity and ridges combine to increase water/ice low-frequency sound penetration enabling long-distance transmission along this liquid–solid interface in the form of an evanescent wave. Consequently, sound pressure and exposure levels near the canopy are significantly higher even when the acoustic wavelength is several times longer than the ice thickness. Therefore, these physical mechanisms should also be taken into account in Arctic environmental impact assessment calculations. For example, it is more efficient for under-ice marine mammals to mitigate exposure to subsequent active sonar events by diving just a few meters deeper under the ice cap rather than increasing range by hundreds of meters. Therefore, some under-ice marine mammals are likely to exhibit a stronger preference toward diving avoidance behavior. [Work funded by UAF sub-award under NOAA Grant NA09NOS4000262.]

9:25

1aAB5. Seasonal presence of ringed, ribbon, and bearded seal vocalizations in the Chukchi Sea north of Barrow, Alaska. Joshua M. Jones, Ethan Roth, Bruce J. Thayre, Ian Sia (Marine Physical Lab., Scripps Inst. of Oceanogr. Univ. of California San Diego, La Jolla, CA 92093-0205), Michael Mahoney, Clarissa Zeller, Malorie Johnson, Christine Jackson, Kyle Kitka, Daniel Pickett (Mount Edgecumbe High School, AK 99835-9418), Robert Small (Alaska Dept. of Fish & Game, Juneau, Alaska), Zoe Gentes, Sean Wiggins, and John Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093-0205)

Long-term autonomous acoustic recordings were collected between September and June from 2006 through 2009 in the Northeastern Chukchi Sea along the continental slope 120 km north of Barrow, Alaska. These recordings were analyzed for the presence of vocalizations of ringed seals (*Phoca hispida*), ribbon seals (*Histrophoca fasciata*), and bearded seals (*Erignathus barbatus*). We present detailed descriptions of the acoustic repertoire of each species in addition to three-year time series of seal vocalizations and mean daily sea ice concentration. Ringed seal vocalizations are present throughout each winter and spring, indicating that they both overwinter and breed in offshore pack ice. Ribbon seal calls occur only during the open water period in 2008, but their acoustic behavior is more varied than previously described. Bearded seal vocalizations closely match well-documented calls recorded offshore near Point Barrow but have shorter duration and smaller frequency range, suggesting that demographic or behavioral differences related to breeding habitat selection may exist within the population. Bearded seal calls peak during the breeding season from March through June, but also occur in December and January annually. These long-term autonomous recordings provide details of seasonal distribution and behavior of Arctic seals that previously have not been possible to observe with other methods.

9:40

1aAB6. The acoustic ecology of minke whales in the tropical north pacific. Thomas Norris (Bio-Waves, Inc., 517 Cornish, Encinitas, CA 92024), Steven W. Martin (Space and Naval Warfare Systems Ctr. Pacific, 53560 Hull St., San Diego, CA 92152), Tina M. Yack (Bio-Waves Inc., Encinitas, CA 92024), Len Thomas (Univ. of St. Andrews, Fife KY16 9LZ, Scotland), and Julie Oswald (Bio-Waves, Inc., Encinitas, CA 92024)

The minke whale (*Balaenoptera acutorostrata*) is a ubiquitous but rarely sighted species that occurs in subtropical waters of the North Pacific during winter and spring. It produces a unique vocalization called a boing that is easy to detect and localize. We characterized the acoustic ecology of minke whales by detecting and localizing boings using passive acoustic methods. We conducted passive acoustic line-transect surveys in 2006 for a large area around the Northern Mariana Islands (site 1), and in 2010 for a smaller area off the Pacific Missile Range, Kauai, HI (site 2). We also recorded acoustic data from deep waters using cabled seafloor hydrophones at site 2. Densities of calling animals were estimated from line-transect surveys and will be used to estimate calling rates for use in spatially explicit capture–recapture analysis of fixed seafloor hydrophone data. Spatial analysis of acoustic localizations in relation to bathygraphic and oceanographic variables will be discussed. We provide examples of counter-calling and complex responses to vessel noise. These results provide important information about the acoustic ecology and behavior of minke whales that can be used to improve the conservation and management of this elusive but common whale. [Work sponsored by ONR and NAVFAC.]

9:55

1aAB7. Characterizing the vocalizations of North Atlantic right whales (NARWs) in Florida, with emphasis on mother-calf pairs. Edmund Gerstein (Dept. of Psych., Charles E. Schmidt College of Sci. Florida Atlantic Univ, Boca Raton, FL 33431, gerstein2@aol.com), James Hain (Associated Scientists of Woods Hole, Woods Hole, MA 02543), James Moir (Marine Resources Council, Palm Bay, FL 32905), and Stephen McCulloch (Harbor Branch Oceanograph. Inst., Ft. Pierce, FL 34946)

The repertoire of calls, frequency of occurrence with respect to social composition, and ontogeny of vocal learning is being investigated for NARWs in their southeastern critical habitat of Florida. Small synchronized, GPS-instrumented, digital autonomous recording packages (DARPs) are deployed in the vicinity of photo-identified individuals, mother-calf pairs, and groups. The unobtrusive DARP buoys enable the recording of relatively rare, low-intensity vocal exchanges between mothers and calves. We hypothesize that it may be advantageous to be silent or to call softly to avoid predators or harassment by males in the area. These mother-calf exchanges could also be related to early developmental aspects of vocal communication. Site-specific bathymetry and active propagation measurements are conducted with vertical arrays to estimate source levels, and ranges of detection. Preliminary data indicate that exemplar (up-calls) recorded in northern habitats and used for training detection algorithms for passive auto-detection buoys are likely not appropriate for Florida waters. Social demographics, associated behavior, along with water depth and environmental parameters contribute to differences in call rates, types, source levels and respective propagation characteristics between the southeastern and northern habitats. Critical acoustic data required for selecting more appropriate calls for effective DCL algorithms in Florida waters are currently being collected.

10:10–10:25 Break

Invited Papers

10:25

1aAB8. Environmental influences on acoustic communication in frogs. Peter M. Narins (Dept. of Integrative Biology and Physio., UCLA, 621 Chas. E. Young Dr. S., Los Angeles, CA 90095, pnarins@ucla.edu)

Many species of animals, including man, face the formidable task of communicating in naturally noisy environments. The effects of noise on both the calling behavior of frogs and the temporal, and spectral filtering ability of the amphibian auditory pathway are discussed. Moreover, the role of spectral, temporal, and spatial separation in minimizing background noise masking will be examined. Behavioral evidence is presented suggesting that environmental noise may act as a strong selective force in sculpting the communication systems of two species of Old World frogs. One torrent frog (*Odorrana tormota*) calls frequently from vegetation along fast-flowing mountain streams in Central China. These streams produce high-level, broadband noise spanning the human hearing spectrum. In addition to the high-pitched audible components, the males' calls contain prominent ultrasonic harmonics. Another frog, *Huia cavitympanum*, lives in a very similar habitat in Borneo. Unlike *O. tormota*, *Huia* can modulate its call spectrum to produce purely ultrasonic calls. It is thought that the upward shift of the call frequencies and the upper limit of sensitivity of both *O. tormota* and *H. cavitympanum* are responses to the selection pressures from their noisy habitats. [Work supported by NIDCD DC-00222, Paul S. Veneklasen Research Foundation, UCLA Academic Senate (3501).]

10:50

1aAB9. Chorussing in delphinids. V. M. Janik (School of Biology, Univ. of St. Andrews, Fife KY16 8LB, United Kingdom), P. Simard (Univ. of South Florida, St. Petersburg, FL), L. S. Sayigh (Woods Hole Oceanograph. Inst., Woods Hole, MA), D. Mann (Univ. of South Florida, St. Petersburg, FL), and A. Frankel (Marine Acoust. Inc., Arlington, VA)

The evolution of communication is strongly influenced by the social structure of animals. Here, we report how a group of offshore bottlenose dolphins in the Gulf of Mexico used chorusing of the same whistle type, while no such behavior was observed in inshore populations of the same species. We recorded 166 whistles from a group of 6 bottlenose dolphins in the Gulf of Mexico, 19 nm from the Florida coast. In an examination of the timing of whistle production, we found nine sequences in which there was considerable overlap (i.e., >50%) between whistles and another eight sequences with almost perfect overlap of the same whistle type produced by two to six animals simultaneously. Such synchrony was not expected by chance. To investigate how unique this behavior was, we also analyzed 300 h of recordings of inshore bottlenose dolphins in Florida and Scotland. In these data we found three non-significant cases of two animals showing >50% overlap. Thus, chorusing appears to be absent in inshore animals. Our data suggest that offshore bottlenose dolphins live in closed social units, which could be the result of enhanced difficulties in maintaining contact if home ranges are large.

Contributed Papers

11:15

1aAB10. Tracking dolphins using long-term autonomous acoustic recorders. Sean M. Wiggins, Martin Gassmann, Kaitlin Fraiser, and John A. Hildebrand (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0205, swiggins@ucsd.edu)

Tracking marine mammals over long periods can provide information on their movement patterns including base-line behavior and responses to natural and anthropogenic stimuli. Autonomous acoustic recorders provide a cost effective and portable means of tracking these sounds over long periods, but until recently these devices have been restricted to tracking low-frequency large whales because of limited recording capabilities. In this paper, we will present long-term, passive acoustic tracking of high-frequency dolphin whistles and clicks using autonomous hydrophone recording arrays with kilometer- and meter-scale apertures.

11:30

1aAB11. High-frequency modulated signals of killer whales (*Orcinus orca*) in the North Pacific. Anne E. Simonis, Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr. La Jolla, CA 92093-0205), Erin Oleson (NOAA Fisheries, Honolulu, HI 96814), Mariana L. Melcón, Martin Gassmann, Sean M. Wiggins, and John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093-0205)

Killer whales (*Orcinus orca*) use acoustic signals to echolocate and communicate, although there are differences in acoustic behavior among ecotypes. Atlantic resident populations recently have been reported to produce acoustic signals at higher frequencies than previously known. Acoustic recordings from ship-based acoustic and visual surveys and from autonomous acoustic recorders reveal that killer whales across a broad range of the North Pacific Ocean also use similar high frequency modulated signals. The median peak frequency of these signals ranged from 19.6 to 36.1 kHz at different locations, with median durations from each location ranging from 50 to 163 ms. All observed high frequency modulated signals were frequency downswept with no or few inflection points. Killer whales are generally believed to use whistles for close range communication in social contexts; however, these uniform, repetitive, down-swept signals are similar to bat echolocation signals and may have echolocation functionality. The large geographic range of occurrence suggests that these signals are utilized by different killer whale ecotypes. [Work funded by Michael Weise at the Office of Naval Research, Frank Stone at Navy CNO-N45, Bob Haskell at Pacific Life, and Mark Spaulding at the Ocean Foundation.]

11:45

1aAB12. Long-term passive acoustic monitoring of parrotfishes (Scaridae) in the *Hawaiian Archipelago*. Lisa M. Munger (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1346 Kaneohe, HI 96744, lmunger@hawaii.edu), Pollyanna Fisher-Pool, Kaylyn McCoy, Marc O. Lammers, Timothy Tricas, Whitlow W. L. Au (Univ. of Hawaii, Honolulu, HI 96744), Kevin Wong, and Russell E. Brainard (NOAA Fisheries, Honolulu, Hawaii)

Parrotfishes (family *Scaridae*) are an important component of coral reef ecosystems, and this key functional group plays a major role in algae removal and bioerosion of reef substrate. They are also heavily fished in many locations, which may lead to ecosystem-wide impacts such as increased algal cover. In the State of Hawaii, parrotfish management is a priority for marine resource managers, with an ongoing need for accurate population monitoring that is currently addressed by diver-based visual surveys. However, parrotfishes are highly mobile and somewhat skittish around SCUBA divers, particularly in areas where fishing pressure is high. Because parrotfishes produce frequent audible scraping and crunching sounds associated with feeding, passive acoustic monitoring (PAM) can provide information on parrotfish occurrence without requiring the invasive presence of divers. Here, we present results from analyses of parrotfish foraging sounds in long-term acoustic recordings from 10 shallow reef locations throughout the Hawaiian Archipelago dating back to 2006. Parrotfish sounds are compared spatially across a fishing pressure gradient, from heavily fished areas in the main Hawaiian Islands to protected waters within the Papahānaumokuākea Marine National Monument (Northwestern Hawaiian Islands). Results from PAM are compared when possible with data from adjacent visual censuses conducted by divers.

Session 1aAO

Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Van Holliday Memorial Session I

Whitlow W. L. Au, Cochair
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Kelly J. Benoit-Bird, Cochair
College of Ocean and Atmospheric Science, Oregon State Univ., 104 COAS Admin Bldg., Corvallis, OR 97331-5503

Michael J. Buckingham, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Timothy K. Stanton, Cochair
Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Inst., Bigelow 201, Woods Hole, MA 02543-1053

Chair's Introduction—8:15

Invited Papers

8:20

1aAO1. Dale (Van) Holliday was so much more than a scientist. Dorian S. Houser (Natl. Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106)

Dale (Van) Holliday is well known for his acoustic studies of ocean life over a scale from plankton to mysticete whales. Van was a globally acknowledged expert valued for his participation in national and international councils governing ocean issues. Multiple times he was awarded high honors, such as the Prix d'Excellence and the Silver Medal in Acoustical Oceanography. More importantly, Van was a man of faith and no respecter of persons; he unselfishly gave of his time and knowledge and treated both students and career scientists with the level of respect due an individual. I was professionally mentored by Van and worked alongside him as a church deacon. As a deacon, I observed Van's compassion—the numerous hospital trips to visit the sick, the visitations of the home bound, and the phone calls to just "check in." His interest in people spilled into his professional relationships, particularly with young and emerging scientists whom Van joyfully assisted and advised. I, and numerous others, have and continue to benefit from our friendship and professional relationship with Van. He was truly unique. He will be missed, but not forgotten, as his science and investment in people will remain impactful for decades to come.

8:40

1aAO2. Van Holliday: Ocean acoustician and biological oceanographer. Richard E. Pieper (Fish Harbor Marine Lab., 820 S. Seaside Ave., Terminal Island, CA 90731)

I first met Van Holliday in 1970 at the International Symposium on Biological Sound Scattering in the Ocean. We were both graduate students, Van at Scripps and I was at the UBC. Two of us were interested in higher frequency acoustics and scattering from zooplankton. In the Working Group on Bioacoustics Recommendations, Brooke Farquhar and Van Holliday wrote "By using acoustic techniques over large frequency ranges (100 Hz to 10 mHz), the distribution of nekton and plankton could be obtained." I moved to USC after graduation and began to meet Van at ONR site reviews. We received funding from ONR and NSF in mid 1970. I bought a plankton pump and Van put together a four-frequency acoustic array (0.5, 1.16, 1.80, and 3.08 MHz). The program began and continued for over 30 yr. With the success of our four-frequency system, we obtained a recently de-classified Shadowgraph system from the Navy and Van built a 21-frequency array (100 kHz to 10 MHz) which was used in many ocean environments. The uniqueness of scattering from zooplankton led Van to develop an inversion algorithm to calculate zooplankton size spectra from the acoustic data. The four-frequency TAPS was designed and is used by many today.

9:00

1aAO3. Van Holliday: A lifetime of contributions to acoustical oceanography. David M. Farmer (The Graduate School of Oceanogr., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882)

Van Holliday's achievements have transformed the acoustical study of marine biology. Working closely with colleagues from fisheries and marine biology, he brought his keen understanding of acoustics and lively curiosity to the solution of many challenging

problems. In reviewing his accomplishments one is struck by their range and by the special way in which he reached out to the fields to which he contributed. For example, when he solved a problem such as acoustical scattering by fish schools, he quickly saw the possibilities of exploiting his new understanding to study schooling behavior. He explored the characteristics of ice noise, he was the first to track the movement of whales acoustically and he made major contributions to our understanding of the scattering of sound by zooplankton. His work on plankton acoustics led to the important new sub-discipline of spectral acoustic determination of zooplankton populations and their behavior. His profound contributions to biology, biophysics, and other marine topics display an intellectual breadth, an enthusiasm for crossing disciplinary boundaries, and a deep curiosity that serve us well as a model for achievement in the field of acoustical oceanography.

Contributed Papers

9:20

1aAO4. On Van Holliday's shoulders. Orest Diachok (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD 20723-6099)

The year was 1992. The Cold War was over, and funding for resolving deep water issues was fading. I was at the NATO Undersea Research Centre, simultaneously searching for a new research challenge that I could sink my teeth into, and formulating a research program for the Centre, which would be appropriate for the post-Cold War era. I stumbled onto Van's classic paper on resonance scattering by fish, discussed it with him by telephone, and within an hour became a convert to marine bioacoustics. I became familiar with David Weston's work, discussed it with him, and soon thereafter started my research in bioacoustics. Van and David made important contributions to the design of my bioacoustics experiments. After I returned to the USA, I often met with Van to discuss a broad spectrum of fundamental and applied topics in bioacoustics. He was always generous with his time, knowledge, and insights. His explanations of seemingly complex phenomena were always readily understandable, concise, and precise. Standing on Van Holliday's shoulders, greatly expanded my vision of what is possible through technical innovation, interdisciplinary collaboration, dedication, and perseverance. I, along with many others, miss and will continue to miss this amazing and unique individual.

9:35

1aAO5. Acoustics as a tool to answer fundamental marine ecological questions: Thanks Van! James E. Eckman (California Sea Grant Prog., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr. 0232, La Jolla, CA 92093, jeckman@ucsd.edu)

Van Holliday was an acoustician with a sound appreciation (apologies) of the complexity, importance, and beauty of marine animals and plants. His bio-acoustical work, motivated initially by an interest in the significant impacts of fish on acoustic backscatter, rapidly led to studies that addressed the value of that backscatter in answering scientific questions about fish. His research and interests expanded rapidly to provide us with new tools capable of detecting and mapping zooplankton. Moreover, his later efforts contributed to an appreciation of the dynamics and importance of the productivity of benthic microalgae in shallow sediments. While many appreciate the value of his contributions to acoustics, and to the basic detection and mapping of planktonic and nektonic life forms, fewer scientists appreciate that the tools (hardware and software) he developed have helped open a door that enables us to address fundamental biological and ecological questions regarding plankton dynamics and benthic-pelagic couplings: questions that still remain inadequately explored. From my perspective as a former program manager at ONR, I will discuss some of Vans contributions that provide a gateway to asking, and answering, a wide-range of ecologically important questions.

9:50

1aAO6. Broadband active acoustic sensing of fish and zooplankton in the kilohertz to megahertz range: Pursuing Van Holliday's vision. Timothy K. Stanton (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543-1053, tstanton@whoi.edu), Dezhang Chu (NOAA Natl. Marine Fisheries Service, Seattle, WA 98112), and Andone C. Lavery (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543-1053)

Van Holliday had a vision for sensing marine organisms with active acoustics over a very wide range of frequencies. This inspired us at the Woods Hole Oceanographic Institution, Woods Hole, MA, to conduct a series of measurements, both in the laboratory and in the ocean, and associated modeling, over the range of 1.5 kHz to 3 MHz over the past 23 years. The organisms were as small as millimeter-size copepods and as large as 20-cm herring and squid. Broadband acoustic scattering measurements were conducted in the laboratory as a function of frequency (24 kHz 3 MHz) and angle of orientation (0–360 deg) of many species. Instruments were developed to measure broadband acoustic scattering in the ocean over the range 1.5 kHz 1.2 MHz with some gaps. Scattering models, based on the laboratory data, were also developed for several major anatomical groups of organisms and spanned a range of complexity, from low-resolution models that account only for length, width, and general shape to high resolution models that account for shape of the body and heterogeneities within the body in three dimensional at fine scale as well as including roughness. In this presentation, we review the laboratory measurements and scattering models, as well as development of the broadband ocean instruments and their use at sea.

10:05–10:20 Break

10:20

1aAO7. Planktonic layers: New insights stimulated by Van Holliday. Timothy J. Cowles (Consortium for Ocean Leadership, 1201 New York Ave. NW, Washington, DC 20005, tcowles@oceanleadership.org) and Kelly Benoit-Bird (Oregon State Univ., Corvallis OR 97331, kbenoit@coas.oregonstate.edu)

During the late 1980s and early 1990s, initial evidence for persistent thin layers of phytoplankton (less than 1 m in thickness) in the upper ocean was obtained with optical instrumentation. These observations were inconsistent with the existing understanding of small-scale vertical mixing processes and raised questions about the ecological consequences of such structures, should they prove to occur on a regular basis. Dr. D. Van Holliday quickly saw the need for new approaches in acoustical oceanography to address these important ecological questions. Over a period of a few years, Dr. Holliday developed new acoustical tools that could be deployed in concert with other sensors. His moorings with customized high-frequency transducers allowed acoustical resolution over scales of centimeters, sufficient to capture the acoustic signature of marine zooplankton and their spatial relationship to their food supply. The insights provided by Van's work, and his constant encouragement to his colleagues, have led to significant breakthroughs in our understanding of upper ocean physical, biological and chemical processes. This presentation will incorporate recent observations and build upon Van's observations to illustrate the importance of his contributions to ocean science.

10:35

1aAO8. Parsing the prose of D. V. Holliday. Kenneth G. Foote (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

The published works of D. V. Holliday have been praised as exemplifying the scientific method [K. G. Foote, J. Acoust. Soc. Am. **115**, 2521 (2004)]. They also exemplify a direct writing style that embodies some elements of a physical theory. Paraphrasing P. Duhem in The Aim and Structure of Physical Theory (Princeton University, 1954), these works, by Holliday, communicate “as simply, as completely, and as exactly as possible, a whole group of experimental” findings. These stylistic features are

illustrated by reference to three papers published in JASA under Holliday's name, one with a coauthor, during the period 1972–1980.

10:50

1aAO9. Active acoustic examination of the diving behavior of murres foraging on patchy prey. Kelly J. Benoit-Bird (College of Oceanic and Atmospheric Sci., Oregon State Univ., 104 COAS Admin Bldg., Corvallis, OR 97331, kbenoit@coas.oregonstate.edu)

Van Holliday used sound to study nearly every type of organism living in the ocean, contributing to our understanding of acoustics as well as marine ecological processes. To honor Van, I will talk about the one animal group that I could not find in his long publication list—seabirds. We combined visual observations of murres with active acoustics, fish trawls, zooplankton net tows, and hydrographic measurements in the area surrounding breeding colonies in the southeastern Bering Sea. We detected thousands of unique acoustic targets that were strongly correlated with the number of visually detected murres, providing a new tool for quantitatively studying the foraging ecology of diving birds. Diving murre abundance was related to the combined availability and vertical accessibility of squid, krill, and pollock. Individual krill patches targeted by murres had higher krill density and were shallower than mean values but were similar in total krill abundance and overall size. Murres found outside of krill showed a depth distribution similar to that of juvenile pollock. The high proportion of diving murres in aggregations and their consistent inter-individual spacing support the hypothesis that intra-specific local enhancement may facilitate foraging in these predators.

11:05

1aAO10. Zooplanktivory of fishes measured with a dual wide band sonar/optical imaging system. Amatzia Genin (Hebrew Univ. and Interuniversity Inst. Mar. Sci. Eilat 88203, Israel), Jules Jaffe (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238), Yoav Lindemann (Hebrew Univ. and Interuniversity Inst. Mar. Sci. Eilat 88203, Israel), Paul Roberts (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238), and Yonathan Guttel (Hebrew Univ. and Interuniversity Inst. Mar. Sci. Eilat 88203, Israel)

Recent experiments in several near shore habitats in Eilat, Israel, were performed to measure various aspects of predation on zooplankton. Toward this goal, two sets of (4) wide band echo-sounders that can identify echoes from individual sub-millimeter copepods in liter-sized volumes were deployed. The echo sounders were augmented with an underwater optical shadowgraph imaging system that concurrently imaged a 100 ml volume with 25 μm resolution that provided many usable *in-situ* images of the animals that were producing the acoustic reflections. Sonars were located both upstream and downstream from a group of zooplanktivorous fish, allowing measurement of zooplankton abundance in water parcels before and after they passed through the fish group. Information on fish abundance and movements was obtained with concurrent video records. Together with ancillary environmental measurements of flow rate and light level, zooplanktivory rates of: animals consumed per cubic meter per fish were computed from the differential concentration of animals as inferred from the two sets of four echo-sounders. The results permit the estimate of important environmental features of the ecosystem such as the dependence of predation rate on prey abundance, their size, and flow speed.

11:20–11:40 Panel Discussion

MONDAY MORNING, 31 OCTOBER 2011 SUNSET, 8:15 TO 11:25 A.M.

Session 1aSA

Structural Acoustics and Vibration: Multifunctional Composite Structures

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

Noureddine Atalla, Cochair
Dept. of Mechanical Engineering, Univ. de Sherbrooke, Sherbrooke QC J1K 2R1, Canada

Chair's Introduction—8:15

Invited Papers

8:20

1aSA1. Multifunctional composite materials and structures: A brief review. Ronald F. Gibson (Dept. of Mech. Eng., MS-312, Univ. of Nevada-Reno, Reno, NV 89557)

Recent advances in the area of multifunctional materials and structures are reviewed. Interest in this area has been driven by the need for systems that simultaneously perform structural and nonstructural functions. Multifunctional materials are by nature composite materials and new nanoreinforcements, in particular, have made it possible to achieve simultaneous improvements in properties such as stiffness, compression strength, damping, electrical conductivity, and fracture toughness. For example, among the most important nonstructural functions that a structure may need is electrical conductivity, but many composites have poorly conducting polymer matrix materials. However, very small concentrations of carbon nanotubes or other conducting nanoreinforcements in polymers lead to disproportionately large improvements in the electrical conductivity of the nanocomposite. Recently developed composites with piezoelectric structural fiber reinforcements are capable of vibration control, damping, energy harvesting/storage, or structural health monitoring. Recent research has also demonstrated the feasibility of self-healing polymers and polymer composites based on the use of a microencapsulated healing agent and a catalyst for polymerizing the healing agent. Structurally integrated electronics such as batteries for energy storage are another recent application of multifunctional structure design. Other important nonstructural functions in multifunctional composite structures include electromagnetic interference shielding, optical transparency, and thermal conductivity.

8:40

1aSA2. Multifunctional acoustic metamaterial composites. Christina J Naify and Steven Nutt (Dept. of Mater. Sci., Univ. of Southern California, 3651 Watt Way VHE 402, Los Angeles, CA 90089)

Composite structures used in aerospace applications are designed to optimize high bending stiffness to weight ratio. Acoustical performance of composite structures, however, is notoriously poor due to decreased mass law performance and decreased coincidence frequency. Efforts to improve the acoustic properties of composite structures have included optimization of material selection as well as addition of acoustic treatments. Additionally, development of composite structures to include multifunctional properties has increased. In this study, two types of multifunctional approaches were examined using acoustic metamaterials. Locally resonant acoustic metamaterials (LRAM), which have sound insulation performance of 500% increase over the mass law prediction, were integrated into a high-strength sandwich structure array to provide increased acoustic performance without increase in mass. LRAM were constructed of thin, tensioned membranes with centrally located masses in the form of small magnets. Varying the geometry of the metamaterial to tune effective frequencies optimized acoustic performance. Acoustic excitation of the central magnet was used to harvest energy by positioning a wire coil around each cell such that the displacement of the magnet induced a voltage. Combination of energy harvesting and array configurations produced multifunctional materials with three applications: increased stiffness, optimal acoustic performance, and harvesting of acoustic energy.

9:00

1aSA3. Wave number and damping characterization for sound and vibration mitigation in sandwich composite structures. James J. Sargianis and Jonghwan Suhr (Dept. of Mech. Eng., Univ of Delaware, 130 Acad. St., Newark, DE 19716, James.Sargianis@gmail.com)

With the rising demand for high performance composite materials, there is a great interest for multi-functional materials. Our research focuses on materials with high noise mitigation and passive structural damping with minimal sacrifice in bending stiffness. Specifically, we seek to understand the vibrational properties of carbon-fiber face sheet sandwich beams, with interest in wave number and damping characteristics. Both experimental and analytical methods were applied to characterize the wave number response of sandwich beams, from which coincidence frequencies were obtained. From the same frequency response function, structural damping ratios were calculated using the half-power bandwidth method. Results showed that for low frequency applications, decreasing bending stiffness has the greatest effect on increasing the coincidence frequency. However for higher frequency applications, an improvement in coincidence frequency can be seen if one reduces the core's shear modulus. Moreover it may be concluded that the high amplitudes seen in the wave number domain may be contributed to the lower structural damping values at low frequencies. With both wave number and damping properties being frequency dependent, one can optimize design parameters pertaining to a certain frequency range which will improve damping and acoustic properties while maintaining necessary flexural stiffness.

9:20

1aSA4. Overview of cumulative results of characterization studies of composites. Bernhard R. Tittmann (Dept. Eng. Sci. & Mech., Penn State, University Park, 212 EES, PA 16802)

After an introduction into design concepts of composites, this paper presents a survey of material characterization investigations of a variety of composites ranging from graphite/epoxy, to metal-matrix, to carbon/carbon to CF/PEEK, and to biological material such as plant cell walls. Discussed are the results of a series of studies in anecdotal form, which include the characterization of soft-body impacts with gelatin projectiles as simulation due to bird strike, the healing of impact damage as imaged and followed by increasing temperature (5000 °C) in acoustic microscopy, finite element simulations of embedded piezoelectric element transducers, acoustic shear velocity measurements during the curing of graphite/epoxy; acoustic emissions during carbonization of carbon/carbon by the use of guided wave techniques inside an autoclave; and acoustic microscopy imaging in the characterization of metal-matrix composites. Always a factor is the resolution requirement. At the extreme end of the resolution capability of current instrumentation is the sub-nanometer resolution of the atomic force microscope which has been demonstrated by the imaging of plant cell walls (onion skin), which are a form of nature's composite structure. These topics have given insight into the many facets of problems as well as opportunities for the use of the characterization of composites.

9:40

1aSA5. Transmission loss of curved sandwich-composite panels with attached noise control materials. Noureddine Atalla (Dept. of Mech. Eng., Univ., de Sherbrooke, Sherbrooke (QC), J1K 2R1, Canada) and Franck Sgard (Inst. de recherche Robert-Sauvé en santé et en sécurité du travail, Montréal (QC), H3A 3C2, Canada)

This paper discusses the modeling of the transmission loss of curved sandwich-composite panels with attached noise control treatments using both analytical and numerical methods. Special attention is devoted to the modeling using the Transfer Matrix Method (TMM) and Statistical Energy Analysis (SEA) of these structures in various mounting conditions (single wall and double wall) under diffuse acoustic field excitation with a systematic comparison with an efficient FEM/VBEM formulation of the problem. Classically, in SEA models the sound package is unwrapped and the TMM is used to calculate its effects in terms of added damping, absorption, and insertion loss. Examples are presented to examine the validity of this practice and demonstrate its range of applicability and usefulness.

10:00–10:15 Break

10:15

1aSA6. Active control of a composite panel utilizing piezoelectric patches connected to negative capacitance shunts. Kenneth A. Cunefare, Benjamin S. Beck (The Georgia Inst. of Technol., School of Mech. Eng., Atlanta, GA 30332, ken.cunefare@me.gatech.edu), and Francisco Mariano Badea Romero (Madrid Polytechnical Univ., INSIA, Carretera de Valencia, 28031 Madrid)

Many industries are implementing the use of thin, lightweight carbon fiber panels to increase stiffness and decrease weight. Yet, vibrations of the panels can reach large amplitudes causing an increased acoustic noise field. The use of piezoelectric actuators bonded on or within the composite panels can be used to decrease the vibration. Negative capacitance shunts have been shown to decrease flexural vibrations over a broadband frequency range. By using the negative capacitance shunts connected to piezoelectric patches, the noise field can be reduced. Yet, the effect of the control may also cause increased acoustic coupling causing more of the flexural modes to create propagating acoustic waves. The acoustic effects of the control system on a clamped carbon fiber composite panel are analyzed by investigating the flexural amplitude and wave number decomposition. The results of the numerical model of the panel are experimentally validated.

10:35

1aSA7. Resonant frequencies of rectangular plates immersed in fluids. Kai Ming Li (Dept. of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031)

The response of a fluid-loaded structure is a subject of considerable interests in many areas of engineering. This includes applications in aerospace industry for composite structures, design of offshore and building structures partially immersed in water, and the design of rectangular micro-plates for bio-sensing devices. For many decades, it has been well recognized that the resonant frequencies of structures in contact with fluid decrease significantly if the fluid-loading on the structure is very heavy. For somewhat lighter fluid-loading, the structural properties are important, and the in vacuo resonant frequencies are usually needed in determining the response due to an external driving force. This paper exploits a simple empirical model to estimate the effect of fluid-loading on the resonant frequency of a rectangular plate. Simple formulas were developed to rapidly compute the natural frequencies of rectangular plates loaded with different fluids on both sides. The rectangular plates with simply supported or clamped edges were considered in the present analysis. Particularly, a square plate with a general boundary condition varying from simply supported to clamped edges was examined. Empirical formulas were derived to calculate the change in the resonant frequencies for baffled plates in all cases.

Contributed Papers

10:55

1aSA8. Acoustic radiation from finite bilaminar submerged plates: Three-dimensional elasticity solution. Sabih I. Hayek (Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802) and Jeffrey E. Boisvert (NAVSEA Div. Newport, Newport, RI 02841)

The vibration and radiation of a submerged baffled finite rectangular bilaminar plate is obtained analytically using the three-dimensional equations of elasticity. The plate is composed of two perfectly bonded finite layers of identical size. The boundary conditions on the perimeter of the plate are free of shear stresses and pinned on the in-plane displacements. The plate is coupled to acoustic media on both sides and is baffled by an infinite rigid plane. The solution for the vibration response due to normal surface forces is found in terms of the bilaminar plate eigenmodes. The vibratory response and the associated near- and far-field radiated acoustic pressures are computed for various ratios of thickness to plate dimensions over a broad frequency range. [Work supported by the ASEE Summer Faculty Research Program.]

11:10

1aSA9. Vibrational characteristics of wood, aluminum, and composite hockey sticks. Linda J. Hunt (Dept. of Phys., Kettering Univ., 1700 W. University Ave., Flint, MI 48504) and Daniel A. Russell (Penn State Univ., University Park, PA 16802)

The stick used by ice hockey players consists of a long straight shaft attached to a curved blade. Composite materials have replaced wood and aluminum shafts for the vast majority of current professional and amateur players. The stiffness of the shaft plays a crucial role in the amount of potential energy that can be stored and released during a slap shot. Shafts are available in a wide range of stiffness and flex ratings in order to match player preference. Several wood, aluminum, and composite shafts were tested using a roving hammer experimental modal analysis with and without their blades. Mode shapes of the shaft tested alone were found to be those of a free-free beam. For shafts of similar lengths, aluminum shafts had the highest resonance frequencies and the lowest damping coefficients, while wood shafts had the lowest frequencies and the highest damping. Vibrational characteristics of whole sticks, including one-piece, two-piece sticks (shaft and blade from same material), and hybrid (shaft and blade of different materials) show that the presence of the blade significantly lowers the frequencies of the torsional modes of vibration. These results, especially damping coefficients, suggest reasons for the anecdotally preferred "feel" provided by composite sticks.

Session 1aSP

Signal Processing in Acoustics, Noise, Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Sampling Methods for Model-Based Parameter Estimation

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

Paul M. Goggans, Cochair
Electrical Engineering Dept., Univ. of Mississippi, University, MS 38677

Chair's Introduction—8:00

Invited Papers

8:05

1aSP1. Using the Markov chain Monte Carlo method to estimate model order. Paul M. Goggans and Chung-Yong Chan (Dept. of Elec. Engr., Univ. of Mississippi, Anderson Hall Rm. 302B, University, MS 38677, goggans@olemiss.edu)

Markov chain Monte Carlo (MCMC) methods are widely used in the solution of parameter estimation problems arising in acoustics and other applications. The use of MCMC to estimate the parameters of a single model is well established. However, in many applications, there is not a single model for the data but rather a number of competing models. A common method of dealing with multiple models is to use MCMC to compute the posterior probability and estimate the parameter values of each model in turn. However, for problems with many models, it is more efficient to combine the parameter spaces of all models into a single space and use MCMC to perform across-model sampling of the joint space. Although the development of an MCMC algorithm of this sort is sufficiently difficult so as to be unprofitable for the non-specialist, the acoustician wishing to solve their multi model parameter estimation problem using MCMC can still do so using an existing algorithm. This presentation gives an overview and brief tutorial of MCMC for parameter estimation and then discusses and gives an example of using the open source computer program BayeSys [Skilling, 2004] to determine the model order of a simple atomic model.

8:25

1aSP2. Bayesian inversion of seabed reverberation and scattering data via parallel tempering. Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria BC, Canada, V8W 3P6) and Charles W. Holland (The Penn. State Univ., State College, PA 16804-0030)

This paper describes Bayesian inversion of reverberation data for seabed scattering and geoacoustic parameters using the method of parallel tempering. The seabed is modeled as a sediment layer over a semi-infinite basement, with interface scattering occurring at the rough upper and lower boundaries of the sediment and volume scattering within the layer. The scattering mechanisms are considered to be independent and are modeled using perturbation theory and the Born approximation. Unknown parameters include seabed geoacoustic properties (sediment thickness and sound speeds, densities, and attenuations for the sediment and basement) and scattering properties (roughnesses and scattering strengths for upper and lower layer boundaries and volume scattering strength for the sediments). The reverberation inversion problem is found to be strongly nonlinear with a highly multi-modal posterior probability density (PPD). Standard Markov-chain Monte Carlo (MCMC) methods, such as Metropolis–Hastings sampling, are ineffective at sampling the complicated parameter space. However, parallel tempering, which runs multiple MCMC chains at a series of increasing temperatures with probabilistic transitions between chains, effectively samples the multi-modal PPD. Methods to increase efficiency using multiple parallel chains at each temperature and exploiting improved mixing with temperature are considered.

8:45

1aSP3. Trans-dimensional strategies for geoacoustic posterior probability estimation. Jan Dettmer and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria BC, Canada V8W 3P6)

Choosing appropriate model parametrizations is a fundamental aspect of Bayesian parameter inference. Trans-dimensional inference is based on a hierarchical Bayesian model, where the number of unknowns in the problem is itself unknown. The methodology extends the state-space to the union of subspaces of a group of models of interest. This allows the data to estimate the support for the parametrizations under consideration while accounting for the limited knowledge about the parametrization in the posterior probability density estimate. Algorithms, such as reversible-jump Markov chain Monte Carlo, can be used to sample from the trans-dimensional posterior by proposing and accepting/rejecting dimensional transitions (jumps) according to a generalized Metropolis-Hastings criterion. Several challenges exist for the efficient application of trans-dimensional methods to geoacoustic inference including defining proposal distributions for efficient transitions between dimensions and within dimensions, and specifying (or quantifying) data errors distributions (or statistics). Example1109; of applying the trans-dimensional approach are demonstrated for several geoacoustic inverse problems, using trans-dimensional partition modeling and hierarchical data-error models.

9:05

1aSP4. Estimation of the conditional variance of a broadband source signal employing a repetition code through shallow water waveguides. Michael Pfetsch and Paul Gendron (Maritime Systems Div. SSC-Pacific, 53560 Hull St., San Diego, CA 92152)

A repetition code can be employed as a computationally fast suboptimal means to exploit available time-bandwidth product for improved bit error rate performance. Applications include synthetic aperture communications. Underwater acoustic response functions can vary over the duration of the transmission and thereby degrade system performance. A measure of precision of estimation of the source signal is the source posterior variance given the receiver pressure field time series. Presented here is the computation of the covariance of a source signal accounting for the covariance of the acoustic response function. A Gibbs sampling scheme is employed as a computational method to computing these variances to an acceptable precision. The resulting posterior variance as a function of SNR gives insight into the contributing phenomena that impart ambiguity in synthetic aperture communications through underwater acoustic channels. The method is tested on the very shallow water environment of Keyport WA at a center frequency of 20 kHz.

9:25

1aSP5. Comparison of particle filtering and extended Kalman filtering for acoustic-based tracking of low-flying aircraft. Wm. Garth Frazier (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, frazier@olemiss.edu) and Chad Williams (Hyperion Technol. Group, Tupelo, MS 38804)

This presentation summarizes a comparative study of particle filtering and extended Kalman filtering applied to acoustic-based tracking of high-speed, low-flying aircraft. A distributed network of small acoustic arrays, each reporting real-time azimuthal bearings and elevation angles to acoustic sources to a central processing unit, are synthesized into real-time three-dimensional tracks. The primary challenge of the problem is the significant propagation delay between the source and the receivers. Both tracking methods are applied to simulated and field test data and reveal nearly equivalent tracking performance in all cases as along as a sufficient number of particles are utilized. Computational requirements of the extended Kalman filter are significantly less than the particle filters.

9:45

1aSP6. Sequential Bayesian filtering for a varying model-order passive fathometer problem. Caglar Yardim (Marine Physical Lab., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, cyardim@ucsd.edu), Zoi-Heleni Michalopoulou (New Jersey Inst. of Technol., Newark, NJ 07102), and Peter Gerstoft (Univ. of California, San Diego, CA 92093)

Sequential model selection is demonstrated for a drifting passive fathometer case. It has been shown that, by processing noise data at a specific array location, a reflector sequence can be extracted, consisting of a summation of sinc pulses. The center of each pulse identifies the depth of a reflector in the ocean environment at that location. Extracting the number and peak/depth locations of these pulses in the reflector sequence provides insight in the sediment structure of the medium. Collecting data in multiple ranges, a process facilitated by the drifting array, allows the study of multiple reflector sequences. Similarly to spatial time delay tracking with Bayesian filters that sample from posterior density functions, we treat sequences obtained at different ranges as data arriving sequentially into a particle filter that extracts at every range (state) the number of pulses and their corresponding depths using an observation equation. A state equation then predicts reflectors at the next range and updates estimates accordingly. The number of pulses varies with range, following changes in sediment layering. Probability density functions of the number of layers and their depths are calculated and demonstrate the successful tracking of changes in the structure of the ocean environment and the uncertainty therein.

10:05–10:30 Break

10:30

1aSP7. Passive sonar tracking using sequences of received signal amplitude fluctuations: Dependence on environmental sampling. R. Lee Culver, Brett E. Bissinger, and Alex W. Sell (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804)

We have developed a passive sonar tracking algorithm that makes use of variations in the amplitude of the signal received from a source in motion. Originally the likelihood function (probability density function (pdf) of received signal amplitude for a given source location) was calculated by Monte Carlo sampling of the environmental variables and running the acoustic propagation model to predict transmission loss (TL). For each possible source location, the pdf of TL was constructed. To calculate the posterior pdf using received signal amplitude data, Bayes' rule was used sequentially such that for each possible source location, the likelihood function evaluated at the value of the new received amplitude was multiplied by the prior probability associated with that source location. This approach depends fundamentally on how the environment is sampled and on the acoustic propagation model. More recently, the likelihood function has been expanded into the joint pdf of sequences of received signal amplitudes. The environment sampling and use of the acoustic propagation model are the same, but now the likelihood functions are multidimensional. Implications for environmental sampling and selecting the proper model order are discussed. Work supported by the Office of Naval Research Undersea Signal Processing.

10:50

1aSP8. Sequential Bayesian filtering for seismic tremor location. Caglar Yardim, Peter Gerstoft, William S. Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093, gerstoft@ucsd.edu), and Eliza Michalopoulou (Univ. Heights, Newark, NJ 07102-1982)

Sequential filtering provides a suitable framework for estimating and updating the unknown parameters of a system as data become available. The foundations of sequential Bayesian filtering with emphasis on practical issues are first reviewed focusing on particle filtering and particle smoothing. Filtering is demonstrated to be a powerful estimation tool, employing prediction from previous estimates and updates stemming from physical and statistical models that relate seismic measurements to the unknown parameters. Particle filtering and particle smoothing are discussed for tracking multiple tremor locations.

Contributed Paper

11:10

1aSP9. Importance-based sampling for porous material physical parameter estimation. Cameron Fackler and Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, facklc@rpi.edu)

Bayesian inference is applied to the problem of determining the physical parameters of a rigid-frame porous material. Such materials may be characterized by the parameters of flow resistivity, porosity, and tortuosity. The Biot-Allard and Attenborough models predict the propagation of sound in rigid-frame porous materials by modeling the characteristic impedance and propagation coefficient of the material in terms of these physical parameters and a shape factor. In this work, model-based Bayesian analysis is formulated to estimate the values of these parameters from experimental measurements of the characteristic acoustic impedance or propagation coefficient of a material under test. Importance sampling and related methods are used in the parameter estimation procedure as Monte Carlo approaches to characterize the posterior probability distribution which is used to represent the likelihood of parameter values. To increase the efficiency of the sampling, adaptive strategies are employed to extend the classical importance sampling. This analysis provides quantitative estimates of the parameter values as well as estimates of the uncertainty associated with each parameter and the interrelationship between parameters.

MONDAY MORNING, 31 OCTOBER 2011 — PACIFIC SALON 3, 8:00 TO 11:45 A.M.

Session 1aUW

Underwater Acoustics and Structural Acoustics and Vibration: Finite Element Modeling of Acoustic Scattering from Objects in a Heterogeneous Medium

Ahmad Abawi, Cochair
Heat Light and Sound Research, Inc., 3366 Torrey Pines Ct., La Jolla, CA 92037

David S. Burnett, Cochair
Naval Surface Warfare Center, 110 Vernon Ave., Panama City, FL 32407

Invited Papers

8:00

1aUW1. Azimuthal Fourier mode decomposition of generic incident fields in the frequency domain: Computational model and results. Mario Zampolli (TNO, Sonar Group, Oude Waalsdorperweg 63, 2597 AK The Hague, The Netherlands, mario.zampolli@tno.nl), Aubrey L. Espana, Kevin L. Williams (Univ. of Washington, Seattle, WA 98105), and Philip L. Marston (Washington State Univ., Pullman, WA 99164)

The 3-D scattering from axisymmetric objects illuminated by nonaxisymmetric incident acoustic fields can be computed efficiently using finite element models, in which the field variables are decomposed via azimuthal Fourier series expansions. The number of azimuthal modes needed is determined by the convergence of the decomposition of the incident field. For simple cases, the field decomposition can be described analytically. For more complex incident fields, however, a closed form azimuthal Fourier series representation of the incident field is often not possible. A pre-processing step is presented, in which the incident field is decomposed numerically at the Gauss points on the wet surface of the target. This approach makes it possible to treat general cases, such as scattering from objects included inside heterogeneous media, at shallow grazing angle, when the symmetry axis of the object is not perpendicular to the interface between the two media. Other applications include decomposing the boundary scattered field re-incident on an object. The model is applied to scattering from a 2:1 aluminum cylinder, included in a fast fluid medium above a slow fluid medium, with the source in the slow medium, and the results are compared to ultrasonic tank measurements. [Work supported by ONR/ONRG.]

8:20

1aUW2. Acoustic scattering from unexploded ordnance in contact with a sand sediment: Mode identification using finite element models. Aubrey L. España, Kevin L. Williams, Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aespana@apl.washington.edu), Mario Zampolli (TNO Defense, Security and Safety, The Hague, Netherlands), David S. Burnett (Naval Surface Warfare Ctr., Panama City, FL 32407-7001), and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

Previous work has illustrated the potential benefit of using low frequency sound as a means for detecting and classifying objects in contact with a sand sediment. In these situations, the wavelength of sound is on the order of the object dimensions, thus coupling to the objects resonant modes. This leads to an acoustic signature rich in physical phenomena unique to the object shape and elastic properties. Hybrid 2-D/3-D finite element models have been developed for unexploded ordnance in contact with a sand sediment. Previous work has demonstrated these models are in good agreement with data collected during experiments conducted in a test pond in 2010 [A. L. España *et al*., J. Acoust. Soc. Am. **129**, 2685 (2011)]. In this paper, the finite element models are used as a means for mode identification and physical interpretation. These modes are visualized through plots of the pressure amplitudes and displacements along the UXO exterior and are explained using insights derived from physical acoustics. Finally, a full 3-D finite element model was developed to investigate the changes to the acoustic response in situations where the symmetry of the problem is broken and the hybrid 2-D/3-D method is no longer viable. [Research supported by ONR and SERDP.]

8:40

1aUW3. Time-domain finite element modeling of interaction of heterogeneous solids with acoustic waves. Petr Krysl (Dept. of Structural Eng., Univ. of California, San Diego, 9500 Gilman Dr., #0085, La Jolla, CA 92093, pkrysl@ucsd.edu)

The vibroacoustic finite element toolkit (VATK) utilizes a Lagrangian finite element method specifically developed for analyzing the interaction of biosolids immersed in fluids with acoustic waves. It employs a superposition principle to separate the incident acoustic wave from the scattered and radiated waves in a displacement-based finite element model. An absorbing boundary condition is applied to the perturbation part of the displacement. Linear constitutive equation allows for inhomogeneous, anisotropic materials, both fluids and solids. Displacement-based finite elements are used for all materials in the computational volume, which can be fluids, solids, and voids in arbitrary combination. Robust performance for materials with limited compressibility is achieved using incompatible-mode brick elements. A centered-difference time-stepping algorithm is formulated to handle general damping accurately and efficiently. The VATK uses a voxel-based modeling scheme for complex geometries. The modeling methodology comes with some challenges. Here we discuss the issues of verification and validation, convergence and error control, and performance of the present technique in application areas for which it was not originally intended. [Work supported by the Office of Naval Research and the Chief of Naval Operations, Environmental Readiness Division.]

9:00

1aUW4. New strategies for full-wave simulations of large acoustic models. Tomi Huttunen (Dept. of Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, FI-70211 Kuopio, Finland, tomi.huttunen@uef.fi)

The simulation of large acoustic models using conventional low-order finite element or boundary element (BE) methods can be computationally demanding due to the requirement of wavelength dependent mesh density (5-10 elements per wavelength is a typical rule of thumb). A recent improvement of finite element type methods has focused on the efficient parallelization of problems to even thousands of processors. The discontinuous Galerkin (DG) methods, in time- and frequency-domain, have been particularly popular since DG methods retain the geometric flexibility of FE methods while allowing a more flexible choice of basis functions. The locality of DG basis makes them ideal for large-scale parallelization and the choice of basis can be easily extended to non-polynomial functions such as plane waves or Bessel functions. This study focuses on the three new strategies for solving large acoustic problems. First, the non-polynomial basis functions are used with the DG method in frequency-domain problems. Second, from element to element varying polynomial order is used for the wave equation in the time-domain. And third, cloud computing is utilized for efficient resource allocation with a fast multipole BE method for the acoustic Helmholtz equation.

9:20

1aUW5. Scattering by an elastic object in the time domain for underwater acoustic applications by means of the spectral-element method. Paul Cristini and Dimitri Komatitsch (CNRS-LMA, 31 Chemin Joseph Aiguier, 13009 Marseille, France)

The increase in computational power which has been occurring during the past years makes it possible to consider the full wave simulation in the time domain of the propagation of acoustic waves in more and more complex configurations in underwater acoustics. Among the various derivations of the finite element method, the spectral-element method has proven to be an efficient and robust tool in seismology. This method combines the accuracy of the pseudospectral method and the flexibility of the finite element method. Its intrinsic properties lend itself very well to numerical simulations on parallel computers, which nowadays is a very big advantage. We will present some results obtained by means of the SPECFEM software, a freely available code which implements the spectral-element method. Its use in underwater acoustics is natural since all types of media which can be encountered in the oceanic environment have already been implemented: fluid, elastic, viscoelastic, anisotropic, and poroelastic. Some illustrative examples of the diffraction of an acoustic wave by an elastic object in various situations will be presented. In particular, a special attention will be paid to the nature of the medium in which the object is embedded.

9:40

1aUW6. Finite-element modeling of sound transmission blocking materials realized using pancake voids in a soft elastomer. David C. Calvo, Abel L. Thangawng, and Christopher N. Layman Jr. (Acoust. Div. Naval Res. Lab., Washington, DC 20375)

The fact that gas-filled voids in a soft elastomer acoustically resemble gas bubbles in a liquid (both having strong low-frequency monopole resonances) was pointed out by analysis and experiment by V. Leroy *et al*. [Appl. Phys. Lett. **95** (2009)]. The present work reports on the modeling, design, fabrication, and testing of specially designed voids in the soft elastomer polydimethylsiloxane (PDMS) with the aim of creating high performance sound transmission blocking materials. The techniques of soft lithography allow the microfabrication of custom cavity shapes in PDMS to obtain desired microstructural resonances. Results of a finite-element modeling analysis are reported that determine the monopole resonance frequencies of what we term pancake voids embedded in layers of PDMS. Because of the high aspect ratio of these voids, the low-frequency monopole (Minnaert) frequencies are between 3 and 6 times lower than that for a spherical cavity of equivalent volume inside PDMS. In terms of the potential for thin sound blocking slabs, this allows a lowering of the frequency at which blocking begins to take place. We also present modeled transmission loss results for normal incidence on an infinite array of pancake voids (a sonic crystal). [Work sponsored by the Office of Naval Research.]

10:00–10:15 Break

10:15

1aUW7. Design of inhomogeneous pentamode metamaterials for minimization of scattering. Jeffrey Cipolla, Nachiket Gokhale (Weidlinger Assoc., Inc., 375 Hudson St, New York, 10014), Andrew Norris, and Adam Nagy (Rutgers Univ.)

Acoustic metamaterials use sub-wavelength, anisotropic, and inhomogeneous microstructures. Macroscopic properties can be related to the microstructure using homogenization theory Hassani and Hinton [Comput. Struct. **69**, 719–738 (1998), which allows an analyst to confirm the extent to which a candidate metamaterial microstructure meets the requirements for a pentamode cloaking material. Norris [Proc. R. Soc. Ser. A, **464**, 2411–2434 (2008)] presented a theory of transformation acoustics that enables the realization of inhomogeneous pentamode acoustic materials having anisotropic elastic tensors, isotropic density and finite mass. This theory describes the spatially varying material properties in terms of a mapping, which for separable geometries may be generated using a scalar function. This function, the constraints on its behavior implicit in the Norris theory, and the material equations constitute the defining relations for pentamode transformation acoustics. Previously, analytic work in transformation acoustics developed the material properties after having fixed a transformation. By reversing the process, we create a number of new families of pentamode cloaking materials. We validate the concept with three-dimensional explicit transient finite element simulations.

10:30

1aUW8. Modeling of offshore wind turbine noise radiation and propagation. Huikwan Kim, Gopu R. Potty, James H. Miller, and Christopher Baxter (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882, hkkim524@my.uri.edu)

Noise generated by offshore wind turbine and support structure can radiate and propagate through the air, water, and sediment. Predicting noise levels around wind turbine structures at sea is required for the estimation of effects of the noise on marine life. To predict radiated noise, we used a finite element analysis (FEA) of a cylindrical shell model of a monopile structure. In the finite element modeling, transient modal dynamic analysis and steady state dynamic analysis (direct and modal) were implemented to simulate both construction and operational noise. The effect of various sediment types and foundation designs are investigated. The FEA package used was ABAQUS version 6.10. The output of the FEA analysis is used as starting field for acoustic propagation models such as PE to produce long range predictions. We present predictions of particle velocity at the structure-acoustic medium interface and sound pressure level as function of frequency at various distances from the structure. Laboratory experiments using scale models of the cylindrical shell have been carried out to verify the noise predictions. Comparison of the FEA model results and experimental data will be presented.

10:45

1aUW9. Acoustic scattering from a metallic pipe: Mode isolation and visualization via finite element analysis. Aubrey L. España, Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aespana@apl.washington.edu), Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814), and Mario Zampolli (TNO Defense, Security and Safety, The Hague, Netherlands)

Understanding how sound couples to, and radiates from targets, aids in finding frequencies and angles advantageous for target detection and classification. Finite element models, coupled with physical acoustics models, have been used to identify resonant modes of truncated, solid cylinders [D. B. Thiessen *et al.*, J. Acoust. Soc. Am. **129**, 2686 (2011)]. This technique is extended here to an aluminum pipe. Scattering from the pipe in the freefield is studied with a finite element model based on the decomposition of the elastic displacements and fluid pressure in a series of azimuthal Fourier modes. To facilitate elastic mode identification, the calculations for the scattered pressure from a nearly rigid pipe of identical size are subtracted coherently from the elastic model results. Plotting these "subtracted" pressure amplitudes and particle displacements on the wet surface of the pipe allows for visualization of the elastic modes. Subsequently, the model is extended for the pipe in contact with sand, via a superposition of freefield results along with the water/sand reflection coefficient. The method is used to study the modal scattering in the presence of the interface. Where applicable, finite element results are validated by predictions from ray theory and experimental results. [Work supported by ONR.]

11:00

1aUW10. High frequency scattering properties of focused acoustic beams helpful for finite element research. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814)

It can be helpful to understand apparent anomalies in the scattering when developing or testing finite element computations for the scattering of sound by objects in water with the wavenumber-radius product ka of approximately 10 or greater. In the case of focused acoustic beams incident on a sphere centered in the axis of the beam, analytical results are available using the method of superposition of Bessel beams [P. L. Marston, J. Acoust. Soc. Am. **129**, 1773–1782 (2011)]. High ka analytical results can manifest an unusually large forward scattering lobe when the focal width of the beam becomes similar to the size of the sphere. (This is in contrast to the examples of scattering at relatively small ka illustrated in the aforementioned reference.) This new behavior can be understood by considering the definition of the total wave field as the superposition of the incident and the scattered fields combined with reasoning based on geometry, the localization principle, and other high frequency approximations. It is likely that high ka finite element computations of the scattering by narrowly focused beams for other shapes of objects will manifest a correspondingly enhanced scattering lobe. [Research supported by ONR.]

11:15

1aUW11. Numerical model of nonlinear acoustic wave propagation in seabed. Rana Arslan A. Khan and Grant A. Ellis (Dept. of Elec. Eng., Univ. Teknologi PETRONAS, 31750 Tronoh, Perak, Malaysia, rarslana@gmail.com)

In this study, a combination of two fundamental frequencies $f_1 = 485$ KHz and $f_2 = 515$ KHz. The source transducers are placed on the surface of the sand 25 cm above the bottom and 20 cm below the water surface in the tank of dimension $74 \times 47 \times 43.5$ cm simulating a seabed with sand of maximum particle size of 5 mm. Burger's equation is the simplest model for describing the second order nonlinear effects in the propagation of high amplitude plane waves and, in addition, the dissipative effects in media. Typically, the interactions of two acoustic waves at a discontinuity in a dispersive but homogeneous medium will generate harmonics and intermodulation terms. This is numerically modeled in MATLAB using the finite difference time domain method with Neumann boundary conditions and practically verified with measurements from the tank. The measured and simulated results have shown significant agreement with the theory. For instance, second order intermod term show 2 dB decreases in amplitude for 7 mm increase in thickness at constant density of plywood. Also if the thickness is kept same, e.g., 5 mm, there is 10 dB drop in same intermod term for change in density from plywood 550 kg/m^3 to copper 8930 kg/m^3.

11:30

1aUW12. Characterization of scattered acoustic intensity fields of finite cylinders in the resonance region. Robert J. Barton III, Geoffrey R. Moss (Naval Undersea Warfare Ctr., code 1522, 1176 Howell St., Newport, RI 02841, robert.barton@navy.mil), and Kevin B. Smith (Naval Postgrad. School, Monterey, CA)

The properties of the scattered acoustic vector fields generated by infinite-length and finite rigid and elastic cylinders are investigated. Analytical solutions are derived from general acoustic pressure scattering models and analyzed for wave numbers in the resonance region. The separable active and reactive components of the acoustic intensity are used to investigate the structural features of the scattered field components. Numerical results are presented for the near and transition regions. A finite element model is

developed for a rigid cylinder and compared to measured results in-air using an anechoic chamber and acoustic vector sensor probes to measure the scattered acoustic vector field. The finite cylinder model and analysis is then extended to include an evacuated thin-walled elastic shell. The vector properties of the time-independent complex intensity components and their relations to field energy density quantities are summarized.

MONDAY AFTERNOON, 31 OCTOBER 2011 SUNRISE, 1:15 TO 5:20 P.M.

Session 1pAA

Architectural Acoustics and Psychological and Physiological Acoustics: Architectural Acoustics and Audio I

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

Alexander U. Case, Cochair
Fermata Audio & Acoustics, P.O. Box 1161, Portsmouth, NH 03802-1161

Chair's Introduction—1:15

Invited Papers

1:20

1pAA1. Bollywood sound stages. K. Anthony Hoover (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362)

Sound for bollywood films is rarely recorded live on location, but is looped or added later, in large part because of poor ambient acoustics. Several new sound stages in Film City, Mumbai, India, were intended for "Hollywood" quality acoustics, but construction ceased shortly after starting. Then, after several years' hiatus, the project was renewed with the directive that the existing fragmentary construction be used in the new design and as the foundation for subsequent construction. This paper will discuss the background, site conditions, encroaching hutments, design issues, concerns for local materials and methods, and the results of post-construction acoustical testing.

1:40

1pAA2. Case study: Active acoustics at the Barbara Streisand scoring stage. Steve Ellison (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com), Shawn Murphy, and David R. Schwind (Charles M. Salter Assoc., Inc., San Francisco, CA 94104)

Scoring stages are recording studios large enough to accommodate an orchestra, and typically used to record music sound tracks for films. The acoustic properties and goals of scoring stages are reviewed, and compared with other venue types. Recently, an active acoustics system was installed in the Barbara Streisand scoring stage on the Sony lot in Culver City, California. The system was used to electronically vary the reflected sound and reverberation during the recording of several film scores. The objectives, design, and performance of the system in the room is reviewed. The range of settings and controls provided to the scoring mixer are described as well as the process used to select parameters for different aspects of the score. These had implications on the artist performance as well as the recording and mixing process. The resultant reverberation times achieved are compared with archetypical orchestral performance venues as well as other scoring stages.

2:00

1pAA3. Case study: Multipurpose venue at Berklee College of Music. Raunak Mukherjee and Eric Reuter (Music Production and Eng./Liberal Arts Dept., Berklee College of Music, Boston, MA 02215, rmukherjee@berklee.net)

This 90 m^2 square meter multipurpose venue was intended to host a variety of programs, including musical performance, lecture, yoga, and dance instruction, etc. Most of these employ a built-in sound reinforcement systems. However, since its construction, the room has suffered from excessive reverberation, making it unsuitable for most of these intended uses, even with reinforcement. Venues of this type demand a balance between speech clarity and a level of reverberation sufficient to support musical performance. The authors will present a detailed acoustical analysis of the room at the start of the project, the goals of the design, and specific recommendations for acoustical treatment.

2:20

1pAA4. Design guidelines for rooms used for music, speech, and teaching. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com) and Christian Nocke (Akustikbüro Oldenburg, Oldenburg, DE D-26121)

Good architectural acoustic design requires an appropriate combination of absorptive, reflective and diffusive surfaces, providing attenuation, redirection, and uniform scattering. Since the terms acoustical and absorptive have become synonymous in common parlance, education of the architectural design community is needed to understand the use of reflective and diffusive surfaces in spaces used for speech, music rehearsal and performance, and teaching. For example, in speech and teaching rooms, where the signal to noise ratio and associated intelligibility are the central issues, designs must not only lower the noise with absorption and isolation but also passively increase the signal with diffusion/reflection. In music education and rehearsal spaces, absorptive low frequency and volume control are important, but reflection/diffusion is equally important for ensemble, tone production and intonation control. In an effort to develop design guidelines, we will review the German standard DIN 18041 on "Acoustic Quality in small and medium sized rooms" (1968, 2004), which standardizes performance objectives in rooms with a volume up to 5000 cbm used for music, speech, and teaching. A frequency and volume dependent suggestion for the reverberation time will be presented as a function of the volume index and examples will be provided.

2:40

1pAA5. Active acoustics used in existing spaces for music practice. Ron Freiheit (Wenger Corp., 555 Park Dr. Owatonna, MN 55060, ron.freiheit@wengercorp.com)

For a number of years, modular practice rooms have offered the option of active acoustics technology as part of their integrated learning environment for music. This approach was effective, in part, because the interior acoustics of each room were well-defined with regard to reverberation time, background noise, and sound isolation. The desire to retrofit existing practice and music teaching spaces with active acoustics introduces numerous challenges. These include addressing environmental limitations (such as unencumbered physical locations for speakers and microphones) and the acoustic environment (issues of sound isolation, reverberation time, and background levels). This paper will explore how these challenges were addressed and how compromises were reached.

3:00

1pAA6. Extending the acoustical versatility of large performing arts classrooms. Robert M. Brenneman and David A. Conant (McKay Conant Hoover, Inc., 7435 E. Stetson Dr., Ste. A, Scottsdale, AZ 85251, rbrenneman@mchinc.com)

To accommodate an extensive array of performing arts programs with limited financial and facility resources, learning institutions often desire a high-degree of acoustical versatility within their larger classroom performance spaces. For the acoustician, achieving quality room acoustics in a singular space suitable for lecture, performance, and application of performance technologies can pose a significant challenge. The large multi-purpose performing arts classroom at Mesa Community College's Red Mountain Campus demonstrates how careful selection of sound absorptive materials, shaping of sound reflective surfaces, material placement, and employment of variable acoustics methods including novel, custom gobos can provide a diverse acoustical environment for addressing the varying acoustical needs of dance, music, and theater departments. This paper presents the cost-effective and practical materials and methods utilized within this multi-purpose performing arts classroom for its use as a teaching, drama, dance, and music performance space for virtually all music genres, from jazz to renaissance choral works.

3:20–3:30 Break

3:30

1pAA7. Subwoofer optimization in rooms for consistancy and efficiency. Todd S. Welti (Corporate RD Dept., Harman Int., 8500 Balboa Blvd., Northridge, CA 91329, todd.welti@harman.com)

Early work on optimizing subwoofer placement in rooms focused on trying to minimize variations in sound power response by optimizing room dimensions. Some consideration has also been given to maximizing overall sound power output, typically by putting subwoofers in room corners. Recent research using a sophisticated computer model has instead focused on trying to minimize variation of magnitude response from seat to seat. Using the same computer model, optimal subwoofer configurations are identified, which give consistent seat to seat responses and maximize low frequency efficiency. Consideration is given to different room dimensions and different seating configurations. The resulting plots may be very useful in designing small rooms with full range audio playback systems and multiple seats.

3:50

1pAA8. On a variable broadband absorption product and acceptable tolerances of reverberation times in halls for amplified music. Niels Werner Adelman-Larsen (Flex Acoust., Scion DTU, Diplomvej 377, Kgs. Lyngby, Denmark, nwl@flexac.com), Jens Joergen Dammerud (Nordic Inst. for Stage and Studio, Oslo, Norway), and Eric R. Thompson (Boston Univ.)

Previous studies have shown that what distinguishes the best from the less well liked halls for pop and rock music is a short reverberation time in the 63, 125, and 250 Hz octave bands. Since quite long reverberation times in these frequency bands are needed in order to obtain warmth and enough strength at classical music concerts, variable acoustics must address these frequencies in order to obtain desirable acoustics in multipurpose halls. Based on the results of a previous study of Danish rock venues as well as reports from three newly built halls, acceptable tolerances of T30 were investigated. The results suggest that T30 can be at least 1.8 times as long in the 63 Hz octave band as in the 125 Hz band and attain values of +/- 15% at higher frequencies compared to the previously determined values. A variable broadband absorption product is also presented. Absorption coefficients are approx. 0.8 in the 125, 250, and 500 Hz bands, 0.6 at 1 kHz and decreasing at higher frequencies, and in the 63 Hz band when in the ON position. In the OFF position the product attains absorption values between 0.0 and 0.2.

4:10

1pAA9. Glass as an aid and as a challenge in acoustics treatment. Sergio Beristain (E.S.I.M.E., IPN, Lab. Acoustics, IMA, Mexico City, Mexico, sberista@hotmail.com)

A new installation for a broadcast center that included recording and transmission studios had to be constructed in an all transparent glass façade building in a major street in Mexico City. Together with all the acoustical specifications requested and needed for proper response of the studios, it was also requested from the very beginning of the project that the building façade could not be modified at all,

i.e., all the glassing could not be obstructed in at least a couple of meters from the edge of the building in order to preserve the look of the building as it was originally constructed. This meant that enough sound insulation for the studios and control rooms as well as the short reverberation times, and resonance control needed by the acoustic conditioning treatment had to be attained with plenty of glass area, which could not be moved, plus all the glassing needed for visual communication between rooms to allow for normal operation of the studios. Construction details and results are presented.

4:30

1pAA10. Cost-effective evaluation of different residential hard-surfaced flooring and underlayment systems. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

There is increasing interest in replacing carpeted flooring with hard-surfaced flooring. Changing from soft-surfaced to hard-surfaced floors always results in an increase in the liveness of hard-flooring rooms and increased impact noise in lower rooms due to reduced impact isolation performance of the revised floor/ceiling assembly. An effective underlayment between the subfloor and the hard-surfaced flooring must be carefully selected and evaluated for ultimate compliance with floor/ceiling assembly performance requirements. In addition, any overall increase in flooring thickness is a major concern. This paper presents optimized field performance testing of several condominiums in a luxury multi-family residential complex for which the homeowner's association had set strict performance and testing requirements. Failure meant restoring the flooring to its original condition. A series of *in situ* and special field tests were undertaken to assist in the selection of qualifying underlayment systems for the desired hard-surfaced flooring. *In situ*, client-preferred, flooring samples were installed by the respective flooring contractors. Special field tests that involved the preferred systems and other flooring/underlayment combinations were conducted in a vacant two-story residence. In the *in situ* and vacant testing locations, the respective variables at each location were the different flooring and multiple underlayment systems.

Contributed Papers

4:50

1pAA11. The practical effects of mixing in an environment closely resembling a home listening environment. Richard King, Brett Leonard, and Grzegorz Sikora (Graduate Program in Sound Recording, Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A1E3)

Two traditional philosophies exist when considering recording studio control room design: the method of closely replicating the listening environment of the consumer, quite often the typical living room, as juxtaposed to the idea that the control room should be uniquely suited toward color-free critical listening. While both of these design philosophies have had their proponents and detractors, very little data have been gathered to show the merits or drawbacks of either when used for critical recording and mixing work. A novel method of task-based testing is developed to determine what, if any, affects can be attributed to particular aspects of control room designs. The method employs highly trained recording engineers and producers to provide real-world feedback on the makeup of a control room's sidewalls. By having these trained subjects to perform basic mixing tasks while altering the reflectivity and diffusiveness of the sidewalls, the effects of these isolated acoustic features are determined. Data showing the final balance levels, as well as total elapsed time per trial, are recorded from more than 20 expert subjects. Despite strong differences in lateral reflected energy, most subjects were able to perform their task with relatively little variance.

5:05

1pAA12. The insertion loss of plenum windows with non-parallel line sources. Y. G. Tong (Dept. of Construction Eng. & Architecture, Univ. Tun Hussein Onn Malaysia, 86400 Parit Raja, Batu Pahat, Johore, Malaysia, ygtong@uthm.edu.my) and S. K. Tang (The Hong Kong Polytechnic Univ., Hong Kong, China)

Opened window for natural ventilation becomes difficult to implement for the buildings in densely populated cities due to traffic noise problem. An experimental investigation of plenum window which was staggered inlet and outlet openings to allow natural ventilation through itself was studied. A 1:4 scale down model was adopted in the study to estimate the acoustical insertion loss of plenum windows in the presence of stimulated non-parallel line source. The orientations of plenum window relative to the traffic roads are found to be significant in protecting from transportation noise due to the staggered design. About 15 dBA of acoustical benefits can be obtained by studied ventilation window system compared to the opened window. The minimum insertion loss was recorded around 6 dBA when the tested window was placed perpendicularly to the line source. [Y.G. Tong sponsored by Ministry of Higher Education, Malaysia.]

Session 1pAOa

Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Van Holliday Memorial Session II

Whitlow W. L. Au, Cochair
Marine Mammal Research Program, Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kailua, HI 96734

Kelly J. Benoit-Bird, Cochair
College of Ocean and Atmospheric Science, Oregon State Univ., 104 COAS Admin Bldg., Corvallis, OR 97331-5503

Michael J. Buckingham, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Timothy K. Stanton, Cochair
Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Inst., Bigelow 201, Woods Hole, MA 02543-1053

Contributed Papers

1:00

1pAOa1. Relationship between volume backscatter from the mesopelagic boundary community and density of organisms in the waters of the Hawaiian Islands. Whitlow W. L. Au, Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744), and Jakob Jung (Bremen Unv. of Appl. Sci.)

Spinner dolphins in the near-shore waters of the Hawaiian islands forage on a mesopelagic boundary community of organisms consisting of myctophids, mid-water shrimps, and small squids. To better understand the foraging ecology of spinner dolphins, it is important to understand the relationship between volume scattering coefficient (Sv) obtained with an echosounder and the density of the organisms in the mesopelagic boundary community. A suite of Simrad Ek-60 scientific echosounders operating at 38, 72, 120, and 200 kHz directed vertically downward was used to collect volume backscatter data from a vessel anchored 1 km from shore. Sv tended to be highest at 38 kHz and lowest at 200 kHz, although at some depths Sv at 70 kHz was slight higher than at 38 kHz data. Immediately after collecting the echosounding data, a profiler with a broadband high-resolution echo ranger projecting a simulated dolphin biosonar signal horizontally was lowered into the same volume examined by the EK-60. The number of organism in the echoranger beam out to a specified range was estimated by counting the number of highlights in the echoes after performing an envelope detection. The relationship between Sv and number of organisms was found to be linearly related.

1:15

1pAOa2. Temporal and spectral feature extraction from fish school using a broadband split-beam echosounder. Masanori Ito, Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 981-3193, Japan), Tomohito Imaizumi, Tomonari Akamatsu (Natl. Res. Inst. of Fisheries Eng., Hasaki, Kamisu, Ibaraki 314-0408, Japan), Yong Wang, and Yasushi Nishimori (Furuno Electric Co., Ltd., Nishinomiya, Hyogo 662-8580, Japan)

For sustainable fisheries, remote discrimination of fish species has been demanded recently. Features for fish discrimination should be extracted from schools. High spatial resolution of broadband split-beam echosounder system enabled tracking individual fish in a school. Temporal and spectral features such as waveform envelopes and target strength spectra were calculated individually. In addition, incident angles of sound were calculated by using positions and velocities of a fish, which were measured by the tracking process of the four channels split-beam system. Because the acoustical features from individual fish depended on the incident angles of sound, all feature parameters were sorted according to the angles to create temporal and spectral averaged pattern. In free ranging condition, echoes of Japanese jack mackerel (*Trachurus japonicus*), chub mackerel (*Scomber japonicus*), and red sea bream (*Pagrus major*) were measured. Results showed clear difference among temporal and spectral averaged patterns, which was consistent with test measurement obtained in an acoustic tank. Broadband split-beam system seemed to be appropriate to extract species specific feature in the ocean. [Work supported by the Research and Development Program for New Bio-industry Initiatives.]

1:30

1pAOa3. Feature extraction for discrimination of fish species by using the cepstral analysis. Ikuo Matsuo, Masanori Ito (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 981-3193, Japan), Tomohito Imaizumi, Tomonari Akamatsu (Natl. Res. Inst. of Fisheries Eng., Ibaraki 314-0408, Japan), Yong Wang, and Yasushi Nishimori (Furuno Electric Co., Ltd., Hyogo 662-8580, Japan)

Identification and classification of fish species are essential for acoustic surveys of fisheries. The echo from the fish contains components from multiple reflections, including the swimbladder and other organs. The target strength (TS) and temporal structure, which were measured and analyzed by using the broadband signal, were changed dependent on the incident angles and fish species. It has been shown that these features were important for discrimination of fish species. The incident angles were calculated by using positions and velocities of each fish, which were measured by the tracking process of the four channels split-beam system, and the target strength was calculated individually by using Fourier transform. In this paper, the cepstral analysis, which was defined as the inverse Fourier transform, was used to extract features for discrimination of fish species from the broadband spectral pattern. In free ranging condition, echoes of Japanese jack mackerel (*Trachurus japonicus*), chub mackerel (*Scomber japonicus*), and red sea bream (*Pagrus major*) were measured and analyzed. It was examined whether cepstral analysis was appropriate to extract species specific features. [Work supported by the Research and Development Program for New Bio-industry Initiatives.]

1:45

1pAOa4. Statistics of echoes from mixed assemblages of scatterers with different scattering strengths and numerical densities. Wu-Jung Lee and Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, wjlee@whoi.edu)

Mixed assemblages are defined to describe the cases in which more than one type of scatterer are present and are randomly located and spatially interspersed among one another in each sonar resolution cell. The probability density functions (pdfs) formed by the echo envelopes in such cases can be highly non-Rayleigh and possess heavy tails. The shape of the pdf curves contains information for characterizing and discriminating the composition of mixed assemblages. A general characteristic-function-based mixed assemblage pdf model is formulated in this study. The model, which takes into account beampattern effects, was validated using numerical simulations. Simulated data of two-component mixed assemblages with different relative scattering strengths, numerical densities, and spatial distributions were used to compare the performance of this new mixed pdf model and the widely used weighted multiple component mixture pdf model. It was found that use of the latter model can lead to orders of magnitude errors in estimating the composition of the mixed assemblages. This study is inspired in the context of acoustic studies of mixed biological aggregations in the ocean, but the theory is generally applicable to other types of mixed assemblages as well.

2:00

1pAOa5. Spatial variation in the small-scale distribution of juvenile walleye pollock (*Theragra chalcogramma*) in the southeastern Bering Sea. Neal E. McIntosh, Kelly J. Benoit-Bird (College of Oceanic and Atmospheric Sci., Oregon State Univ., 104 COAS Admin Bldg., Corvallis, OR 97331, nmcintosh@coas.oregonstate.edu), and Scott A. Heppell (Dept. of Fisheries and Wildlife, Oregon State Univ., 104 Nash Hall, Corvallis, OR 97331)

Juvenile walleye pollock (*Theragra chalcogramma*) is one of the primary prey items for bird and mammal predators in the Bering Sea and supports a large commercial fishery. An understanding of the abundance and distribution of juvenile pollock is needed to estimate the effects that change in these parameters may have on pollock predators and adult pollock abundance and distribution. During the summers of 2008 and 2009, surveys were conducted in three topographic zones (Middle Shelf, Outer Shelf, and Slope) near the Pribilof Islands in the southeastern Bering Sea. Multi-frequency (38, 70, 120, and 200) acoustic sampling occurred during the entire cruise duration with frequent environmental data sampling (e.g., temperature, salinity, dissolved oxygen, and chlorophyll *a* fluorescence) and targeted fish tows. These data showed that juvenile walleye pollock were primarily found in clusters of small, dense aggregations giving them a leopard spot appearance in the acoustical output. In both years, juvenile pollock distribution was highly variable on small spatial scales and was related to biological and physical features of the water column. These differences in juvenile walleye pollock distribution are likely to affect the use of habitat by predators and may have implications for future sampling.

2:15

1pAOa6. Estimating Atlantic Bluefin Tuna number density using the second moment of intensity. Madeline L. Schroth-Miller, Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, mmiller@ccom.unh.edu), and Molly Lutcavage (Univ. of Massachusetts, Amherst)

Fish number density can be estimated from the normalized second moment of acoustic backscatter intensity [Denbigh *et al*., J. Acoust. Soc. Am. **90**, 457–469 (1991)]. This method assumes that the distribution of fish scattering amplitudes is known and that the fish are randomly distributed following a Poisson volume distribution within regions of constant density. It is most useful at low fish densities, relative to the resolution of the acoustic device being used, since the estimators quickly become noisy as the number of fish per resolution cell increases. The method was applied to an acoustic assessment of juvenile Atlantic Bluefin Tuna, *Thunnus thynnus*. The data were collected using a 400 kHz multibeam echo sounder during the summer month of 2009 in Cape Cod, MA. Due to the high resolution of the multibeam system used, the large size (approx. 1 m) of the tuna, and the spacing of the fish in the school, we expect there to be low fish densities relative to the resolution of the multibeam system. Results based on the normalized second moment of acoustic intensity are compared to fish packing density estimated using aerial imagery that was collected simultaneously.

2:30

1pAOa7. Forward scattering at low acoustical frequencies from schools of swim bladder fish. Maria Paz Raveau and Christopher Feuillade (Pontifícia Universidad Católica de Chile, Ave., Vicuña Mackenna 4860, Santiago, Chile)

Low frequency acoustic scattering from swim bladder fish is dominated by the bladder resonance response. At near-resonance frequencies in dense schools of these fish, acoustical interactions between the fish can cause the ensemble scattering behavior to become highly complex. A school scattering model [J. Acoust. Soc. Am. **99**(1), 196 (1996)] has previously been developed to incorporate both multiple scattering effects between neighboring fish, and coherent interactions of their individual scattered fields, in order to realistically describe the collective back scattering behavior of fish schools. In this present work, the school scattering model has been extended to investigate the properties of the acoustic field scattered in the forward direction. In this region, the scattered field and the incident field must be added coherently to obtain the total field. As in the case of back scattering, the field displays marked frequency dependent affects, which are caused by different combinations of the packing density, structural configuration, and resonance frequencies of the individual fish in the school. The results give new insights into the evolution of the acoustic field as it propagates through the school, and the scattering of sound from the incident beam. [Work supported by ONR.]

2:45–3:00 Break

3:00

1pAOa8. Acoustic cross sections and resonance frequencies of large fish schools. Thomas R. Hahn (Great Lakes Ctr., Buffalo State, SUNY, 1300 Elmwood Ave., Buffalo, NY 14222, hahntr@buffalostate.edu) and Orest Diachok (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD 20723)

A prerequisite for stable inversion of bioacoustic parameters of fish schools from large-scale broadband acoustic observations is a theoretical model that permits fast and accurate calculations of acoustic cross sections and school resonance frequencies based on realistic geometrical models of fish schools. The schools of commercially important species, such as sardines, anchovies, and herring, may be characterized by dense nuclei which contain tens of thousands of individuals (N) with separations (S) on the order of one fish length, and diffuse "fuzz" regions with fish at significantly larger separations. Numerical computations of cross sections and school resonances based on the fundamental equations of multiple scattering for point scatterers for these fish school geometries will be presented. Initial results indicate that bubble cloud frequencies of large schools depend primarily on the average spacing between fish in the nuclei and are essentially independent of school size and shape. It will be shown that theoretical calculations of bubble cloud frequencies (based on previously reported values of N and S of sardine schools) are consistent with Diachok's (1999) observations of the average resonance frequency of sardine schools which were derived from broadband transmission loss measurements. [Work is supported by the Office of Naval Research.]

3:15

1pAOa9. Mid-frequency backscatter from spatially organized fish schools. Thomas C. Weber, Madeline L. Schroth-Miller (Crt. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, weber@ccom.unh.edu), Molly Lutcavage (Univ. of Massachusetts, Gloucester, MA 01930), Shachak Pe'eri, and Yuri Rzhanov (Univ. of New Hampshire, Durham, NH 03284)

Schools of Atlantic bluefin tuna, *Thunnus thynnus*, can exhibit highly organized spatial structure. A stochastic simulation has been used to investigate the impact of this spatial structure on the backscattered acoustic field at frequencies below 10 kHz. The simulations are seeded with realizations of schools of juveniles based on field observations from 2009 in Cape Cod

Bay. The field observations, which consist of both aerial imagery and 400 kHz multibeam echo sounder backscatter, have been used to characterize the school morphology, number of fish, and spatial structure within the school. The simulation examines various degrees of structure within the school, starting with fish locations that are constrained by the school boundaries but are otherwise the result of a Poisson process, and gradually incorporating components of school structure such as nearest neighbor distance and quasi-crystalline school sub-structures containing different numbers of fish. Results of the simulation suggest that multiple scattering is negligible except at low frequencies near the swimbladder resonance. Above resonance, even a modest degree of structure within the school (e.g., spatial constraints on pairs of fish) results in appreciable changes to the scattered field. [Work supported by ONR.]

3:30

1pAOa10. Calculations of coherent backscatter enhancement from aggregations of underwater scatterers. Adaleena Mookerjee and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109-2133)

Coherent backscattering enhancement (CBE) is a multiple scattering phenomenon that occurs in optics and acoustics. For plane wave illumination of an aggregation of randomly placed omni-directional scatterers, CBE may lead to as much as a doubling of the scattered field intensity in the direction opposite that of the incident wave. This presentation compares different calculation techniques for CBE for plane-wave harmonic incident sound fields scattered from N underwater bubbles. The first calculation technique is based on the classical field theory of Foldy (1945) for point scatterers and is equivalent to a boundary-element calculation that uses one element per scatterer and requires inversion of a fully populated $N \times N$ matrix. The second calculation is based on the finite element method and involves inversion of a larger but more sparsely populated matrix, but it allows for the possibility of greater scatterer complexity. Results from both approaches are compared and contrasted for idealized underwater remote sensing scenarios involving different incident-wave frequencies, scatterer cross sections, and ranges from the scatterer aggregation to the receiving array. The eventual goal of this effort is to understand and predict CBE from fish schools. [Work supported by the Office of Naval Research.]

3:45

1pAOa11. The potential of bioacoustic absorption spectroscopy for long term monitoring of fish populations in the ocean. Orest Diachok (11100 Johns Hopkins Rd., Laurel, MD 20723-6099)

Bioacoustic absorption spectroscopy (BAS), which exploits frequency selective attenuation due to swim bladder resonances, could be incorporated into ocean observatories for long term monitoring of fish populations. BAS measurements employ environmentally friendly source levels (170 dB) and provide unbiased (by avoidance and proximity to boundaries) estimates of number densities versus fish length (and with ancillary information, species). The BAS II experiment, which was conducted in the Santa Barbara Channel, and employed a fixed ultra-broadband (0.3–10 kHz) source and a fixed vertical hydrophone array separated by 4 km, demonstrated the power of this method. Observed absorption lines in frequency/depth space were consistent with theoretically calculated resonance frequencies of directly sampled year classes of sardines and anchovies, OConnell's (1955) measurements of the dimensions of the swim bladders of these species, and echo sounder measurements of layer depths. Strategies for incorporating BAS measurements into existing fish monitoring methods will be addressed. [This research was supported by the Office of Naval Research.]

MONDAY AFTERNOON, 31 OCTOBER 2011 PACIFIC SALON 4/5, 1:00 TO 5:05 P.M.

Session 1pAOb

Acoustical Oceanography and Underwater Acoustics: Steven Schock Session on Acoustic Bottom Characterization and Subbottom Imaging Including Buried Objects

Altan Turgut, Cochair
Naval Research Lab., Acoustics Div., 4555 Overlook Ave., SW, Washington, D.C. 20375

Dajun Tang, Cochair
Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Chair's Introduction—1:00

Invited Papers

1:05

1pAOb1. The evolution of the chirp sonar high resolution sub-bottom sonar. Lester R. LeBlanc (Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, Fl, lleblanc6@cfl.rr.com) and Larry A. Mayer (Univ. of New Hampshire, Durham N.H. 03824)

Professor Steven G. Schock completed his Ph.D. Graduate program of studies under our guidance in the Department of Ocean Engineering, University of RI. The Chirp Sonar research project that supported Stevens graduate studies was proposed to, and supported by ONR (Dr. Joseph Kravitz) in the early 1980s The PIs were Professors Lester Leblanc (Ocean Engineering) and Professor Larry Mayer (Ocean Engineering and Geological Oceanography). Dr. Mayer and I shared a common interest in improving the quality of sub-bottom sonars so as to provide high quality quantitative data for imaging and classification of the sea floor. It was Stevens hard work at programming the signal processing and assembling and building of components to the chirp sonar that greatly contributed to making this program a success. The initial system, housed in two large racks was tested by Steven in Narragansett Bay RI, and unprecedented high resolution images of the sub-bottom seafloor were generated with the data collected for his research program. This paper will present the theory and technological development of the chirp sonar from the early days to the present transitioned systems that are built, improved, and maintained commercially.

1:25

1pAOb2. Steve Schock and the Narragansett Bay Project. Kenneth M. Walsh (K+M Eng. Ltd., 51 Bayberry Ln., Middletown, RI 02842, kwalsh4@mindspring.com)

One of the first tasks for Steve Schock's sub bottom measurement system was with the Naval Undersea Warfare Center at Newport, RI. A research CRADA was established between NUWC and Precision Signals (Dr. LeBlanc and Dr. Schock) to measure the bottom sediment beneath the torpedo test range. The measurements were successful. The bay at the test range was measured down to bed rock at 64 m. The results indicated that the bay had been a glacial lake from the time the ice receded; until the sand bar that blocked the mouth of the bay eroded and the lake became a salt water bay, open to the ocean. Dr. Schock and Dr. LeBlanc authored a number of technical papers detailing the technology and its application. They founded the Chirp Lab at Florida Atlantic University in Boca Raton, Florida, where the technology has been advancing. Some of the latest techniques were presented in a special session at the May 2011 ASA meeting in Seattle.

1:45

1pAOb3. Now you see it, now you don't: Chirp imaging of the intermittently shelly shoreface ravinement surface on the inner shelf of Panama City, Florida. John A. Goff (Inst. for Geophys., Jackson School of Geosciences, Univ. of Texas at Austin, 10100 Burnet Rd., R2200, Austin, TX 78758, goff@ig.utexas.edu)

Reconnaissance CHIRP data and vibracores were collected on the inner shelf off Panama City, Florida, in April, 2011, for the purpose of providing seabed characterization for an upcoming ONR acoustic reverberation experiment. The seafloor in this region is part of the MAFLA sand sheet: Holocene shelfal marine sands, 0–5 m thick, extending from Mississippi to the Florida panhandle. Coring often samples a thin shelly layer, associated with the shoreface ravinement, at the base of the sand sheet, followed by finer-grained estuarine sediments. Prior CHIRP data collected by Steve Schock off nearby Fort Walton Beach revealed a highly intermittent reflector that could be correlated to the base of the sand sheet; whether the reflector is caused by the estuarine sediments or the shells was uncertain. The new data also reveal an intermittent basal reflector. Estuarine layering can also be identified, and in parts of our survey area the basal and estuarine horizons are distinct. A core at one of these locations sampled a 0.5-m thick shell layer corresponding to the basal reflector with sand both above and beneath. Shells are therefore likely responsible for this intermittent reflector, and thus themselves likely very heterogeneous in concentration at the ravinement surface.

2:05

1pAOb4. Application of chirp technology in earth science: From sediment dispersal to acoustic trenching of faults. Neal Driscoll (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093) and Graham Kent (Univ. of Nevada, Reno, Reno, NV 89557)

CHIRP technology developed and perfected by Steve Schock and others allows scientists to image the earth at the scale processes shape it. Here, we present CHIRP images from a number of different tectonic and depositional settings. One example is from the Salton Sea, where we discovered faults near the southern end of the San Andreas Fault. Rupture on these newly discovered step-over faults has the potential to trigger large earthquakes on the southern San Andreas Fault (M7.5). Using the CHIRP technology to conduct "acoustic trenching" has revolutionized the study of paleoseismology and geohazards. Another example of CHIRP technology is from the continental shelf edge, offshore the US East Coast where large tensional cracks are observed (~4 km long and 1 km wide) and they might mark the location of the next slope failure along the margin. Even though rare, slope failure along the continental margin may lead to tsunami generation along the US East Coast. Sedimentary layers are the recorder of earth history and CHIRP technology allows us to image and decipher the origin of these layers in terms of climate change and tectonic deformation. The development of this technology is clearly one of the big advancements in subsurface geophysical imaging.

Contributed Papers

2:25

1pAOb5. Can slowly varying sediment layers respond acoustically like a discrete target? Charles W. Holland (Appl. Res. Lab., State College, The Penn State Univ., PA 16804) and Dale D. Ellis (DRDC, Atlantic Dartmouth, NS, Canada)

One of the most challenging issues in active sonar is the discrimination of targets from clutter. In bottom-limited areas, one of the primary sources of clutter is from the seabed. An important source of seabed clutter is discrete objects that may lie on or under the sediment interface. It is clear that target-like scattered returns should arise from discrete features that have a spatial scale, or scattered response with time scales of order of the target of interest. The intent of this research is to show that in certain cases, slowly and continuously varying sediment layers can also lead to a target-like response. This is not an intuitive cause of clutter, and the physical mechanisms that lead to the target-like response are expounded.

2:40

1pAOb6. Physics-based inversion of high-frequency multibeam sonar data. Darrell Jackson and Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

This work follows a path begun by other investigators toward physics-based inversion of sonar data. Data were acquired as part of the GulfEx11 experiment using a Reson 7125 multibeam sonar mounted on a vessel in a four-point mooring. One important feature of this effort is the simultaneous acquisition of ground-truth on seafloor properties, including sound speed, attenuation, and roughness. In addition, several frequencies, spanning 150–450 kHz, were used. As typical of this approach, a model is used to generate echo intensity time series including scattering by both seafloor roughness and volume heterogeneity. The model parameters are adjusted to provide a match to data, yielding estimates of acoustic impedance, attenuation, volume scattering strength, and roughness spectral parameters. Volume scattering is treated using an empirical model, while roughness scattering is treated using the small-slope approximation. The narrow beamwidths of sonars of this type facilitate separation of the roughness and volume signals, but pose a challenge with regard to compensation for variations in pitch and roll. A compensation scheme will be discussed and inverted parameters will be compared to ground truth. [Work supported by ONR.]

2:55

1pAOb7. Littoral and deep-seafloor sediment characterization from recent chirp sonar surveys. Altan Turgut, Raymond J. Soukup, Edward L. Kunz (Naval Res. Lab., Acoust. Div., Washington, DC 20375), and Warren T. Wood (Marine GeoSci. Division, MS 39529)

Sediment sound-speed and attenuation were estimated from the chirp sonar reflection data that were collected during recent surveys in the New

Jersey Shelf and northern Gulf of Mexico. Physical properties of the sediment were also estimated using the Biot–Stoll model outlined in [Schock, IEEE J. Ocean. Eng. **29**(4) (2004)]. In addition, independent measurements of sound-speed and attenuation were also conducted using acoustic probes and light-bulb implosions. Results from independent measurements were in agreement with those of chirp sonar data. Measured sound speed and attenuation values are typical of silty-sand sediments in the New Jersey Shelf and silty-clayey sediments in the northern Gulf of Mexico. Finally, frequency-dependence of sound-speed and attenuation was estimated within a wide frequency band using the data from co-located light-bulb implosion (0.5–4 kHz), chirp-sonar, (2–12 kHz), and sediment acoustic-probe (5–120 kHz) measurements. Observed small frequency-dependence of sound-speed and linear frequency-dependence of attenuation are in agreement with those predicted by an extended Biot–Stoll model. [Work supported by ONR.]

3:10–3:20 Break

Invited Papers

3:20

1pAOb8. Techniques in coastal seismic oceanography: An example in the Adriatic Sea. Warren Wood (Naval Res. Lab., 1005 Balch Blvd, Stennis Space Ctr., MS, 39529), Richard Hobbs (Univ. of Durham, United Kingdom), Jeffrey Book (Stennis Space Ctr., MS 39529), and Sandro Carniel (Inst. of Marine Sci., Venice)

The rapidly developing field of seismic oceanography (SO) uses frequencies far lower (10–200 Hz) than tradition acoustic oceanography, and is not a measure of particulate scattering strength, but rather a direct, quantitative measure of vertical temperature gradient. The temperature gradient is typically a very weak signal (reflection coefficients of 0.001 or less) in the presence of higher amplitude coherent noise, such as the direct wave, and the reverberation of the ship noise (in shallow water). Towing the system faster than 4–5 kts to cover greater distances quickly increases streamer noise. Our objective to develop SO into a useful oceanographic tool is, therefore, a signal-to-noise problem, with mostly coherent noise. Using data acquired in the first coastal application of SO (ADRIASEISMIC), we show the magnitude of the noise sources and how they have been mitigated to result in quantitative (albeit band-limited) measures of temperature gradient from a few tens of meters below the sea surface to just meters above the seafloor. The profiles allow the detailed tracking of very small ocean features, among them warm thermohaline intrusions and dense, cold, bottom water masses, both in places only 10 m thick.

3:40

1pAOb9. Modeling backscatter from a series of sediment rough interfaces and mud inclusions by a normal incident chirp sonar. Dajun Tang and Brian Hefner (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

A chirp sonar measures reflection and backscatter of normal incident sound from sediment interfaces and volume heterogeneity. Motivated by using such chirp sonar data to invert for goeacoustic parameters, a forward model has been developed and reported that uses an exact method on which practical models can be based. Further development of the model is presented to (1) investigate spatial resolution of rough interface scatter for given bandwidth and (2) to include the scattering from volume heterogeneity such as mud inclusions.

4:00

1pAOb10. Acoustic imaging and structural acoustic analysis of scattering from buried targets at above-critical grazing angles. Zachary J. Waters (Physical Acoust. Branch - Code 7130, Naval Res. Lab., 4555 Overlook Ave., Washington, DC, 20375)

Laboratory experiments are conducted in order to examine above-critical angle source configurations for the detection and identification of objects fully buried in water-saturated sediments. A stationary broadband spherical source (3–40 kHz) insonifies realistic unexploded ordnance (UXO), as well as objects representing both natural and man-made clutter, at several aspects from above the critical angle. Bistatic returns, received on a two-dimensional synthetic array, are processed to generate volumetric acoustic images of the objects buried in a variety of orientations. Physical acoustics based interpretations are applied in order to identify features attributed to geometric and elastic scattering processes, as well as the interaction of scattered returns with the water-sediment interface. The symmetry of images attributed to cylindrically shaped UXO is suggested as a potential feature for the discrimination of these objects from clutter. The complementary role of volumetric imaging relative to feature-based identification from this same data set is discussed. [Work Supported by SERDP and ONR.]

Contributed Papers

4:20

1pAOb11. Acoustic imaging and structural acoustic analysis of laboratory measurements of scattering from buried targets above critical grazing angles. Harry J. Simpson, Zachary J. Waters, Brian H. Houston (Physical Acoust. Branch, Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, Harry.simpson@nrl.navy.mil), Kyrie K. Jig, Roger R. Volk, Timothy J. Yoder (Sotera Defense Solutions, Crofton, MD 20815), and Joseph A. Bucaro (Excet, Inc., Springfield, VA 22151)

A broadband (3-40 kHz) compact range measurement technique has been developed to obtain the acoustic scattering from buried unexploded ordnance and objects simulating natural and man-made clutter. The targets—two 5 in. rockets with 0, 30, and 60 deg pitch angles, a large rock, and cinder blocks with 0 and 45 deg roll—are buried 10 cm beneath the surface of a water-saturated sandy bottom with a mean grain size of 240 μm. A 2D synthetic array is generated at a height of 20 cm above the sediment–water interface with an element spacing of 3 cm (25 kHz Nyquist). Waveforms collected on the synthetic array are processed to extract the structural acoustic response of the buried targets. A Relevance Vector Machine algorithm applied to the scattered data for target identification, which shows that the target features separate even as the receiver array size is considerably decreased. Similar results are presented for numerical simulations of the bistatic returns for the buried 5 in. rocket and a rock of comparable size. [Work supported by ONR and SERDP.]

4:35

1pAOb12. Buried object imaging using reflection tomography. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

A recent pioneering study of Steven Schock provided high-quality imaging of buried objects using the buried object sensing sonar (BOSS) system in a towed or AUV-mounted configuration [Schock, Proc. IEEE Oceans, 2005]. This paper numerically investigates the performance of compressive sensing (CS) and standard backprojection methods for imaging multiple buried objects using a sonar system similar to the BOSS. Synthetic data are generated using a 3-D pseudo-spectral model for different sediment types and geometries including synthetic aperture sonar. As compared to the standard backprojection methods, the CS-based reflection tomography provided sharper images of buried objects using smaller data sets. Based on the CS performance, several improvements on the BOSS system and data collection schemes are also presented. [Work supported by ONR.]

4:50

1pAOb13. Delay, scale, and sum migration for planar layer imaging. Sean K. Lehman (Lawrence Livermore, Natl. Lab., 7000 East Ave., Livermore, CA 94550)

Wave-based remote sensing and imaging provide methods for investigating structures or objects with minimal or no contact. Delay, scale, and sum migration is one such method (in some circles this is known as ‘‘beam forming’’ or ‘‘migration’’ but must not be confused with beam forming for target location or geophysical migration). Migration assumes coherently scattered fields will sum constructively at a scattering target and destructively elsewhere. There is an additional assumption that multiple scattering can be neglected but this can be relaxed as the forward model sophistication is increased. This presentation summarizes the forward scattering model and derived inverse imaging algorithm as applied to a two planar layer medium. The measurement system operates in a broadband reflection mode in the one layer with the goal of imaging the second layer. Examples are presented using real data in (1) an ultrasonic experiment to measure a flaw in an aluminum/copper multilayer and (2) a ground penetrating radar in an air/sand environment. The examples provide a proof-of-principle using real data and may be scaled to other wavelengths and environments. [LLNL-ABS-485305 This work performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.]

MONDAY AFTERNOON, 31 OCTOBER 2011 TOWNE, 2:00 TO 4:45 P.M.

Session 1pMUa

Musical Acoustics: Acoustics of Mouth Organs

James P. Cottingham, Chair
Dept. of Physics, Coe College, Cedar Rapids, IA 52402

Chair's Introduction—2:00

Invited Papers

2:05

1pMUa1. Eastern and Western free reed mouth organs: Acoustics and history. James P. Cottingham (Phys. Dept., Coe College, 1220 First Ave., NE, Cedar Rapids, IA 52402)

The Asian mouth blown free reed instruments are of ancient origin and typically employ a free reed strongly coupled to a pipe resonator. In these reed-pipes, the same reed often operates on both directions of airflow and behaves as a blown-open or outward striking reed, with playing frequency above both the resonant frequency of the pipe and the natural frequency of the reed. Although the Asian instruments were known in Europe when the Western free reed family originated there about 200 yr ago, the Western free reed instruments use a different mechanism. In these instruments, the reed tongue is offset from the opening in the frame, permitting operation on only one direction of air flow. Pipe resonators are not required and generally not used. These free reeds behave as blown-closed or inward striking reeds, with playing frequency below the natural frequency of the reed. This paper presents some episodes in the history of the Eastern and Western free reed instruments and a summary of the important acoustical properties of each.

2:25

1pMUa2. Chromatical playing on diatonic harmonica: From physical modeling to sound synthesis. Laurent Millot (Inst. d'esthétique, des arts et technologies UMR 8153, CNRS, Univ. Paris 1, ENS Louis-Lumiére, 7 allée du Promontoire, BP 22, F-93161 Noisy-le-Grand Cedex, France)

To understand chromatical playing on a diatonic harmonica one must consider both the instrument and the musician. For each of the harmonica holes, two reeds share a single reed chamber and, theoretically, there is one reed for each direction of airflow (blown or drawn note) while both reeds practically contribute to escaping airflow. The airflow model must take into account all 3D local airflow free jets passing between the plate and the local bent portion of the studied reed to give access to realistic numerical simulations. But to explain bends and overnotes, the vocal tract of the musician must be included and adapted. The vocal tract model must include at least a pressure excitation in the back cavity, the palatal constriction, the front cavity and a mix between lips channel and harmonica chamber. And, the physics involved in the whole model is acoustical nonlinear fluid mechanics rather than acoustical wave propagation. All relevant physical phenomena will be described and illustrated, sound synthesis given to listen. Moreover, a description of the time-variation of the vocal tract model will be proposed for ones thinking about real-time sound synthesis.

2:45

1pMUa3. Alternative method to determine minimal load for reeds within an acoustical flow. Laurent Millot (Inst. d'esthétique, des arts et technologies UMR 8153, CNRS, Univ. Paris 1, ENS Louis-Lumière, 7 allèe du Promontoire, BP 22, F-93161 Noisy-le-Grand Cedex, France)

To understand (free) reed(s) oscillations, a minimal acoustical load must be introduced between the excitation and the reed(s). Classically, the model for this load is either an acoustic admittance/impedance or a reflexion function. But with an existing acoustical flow as found within some experiments, the whole model should better use nonlinear acoustical flow descriptions. Considering electrofluid analogies to model local behavior of either incompressible or compressible elements, an iterative building of the acoustic load can be derived from which, for each studied load, a pure recursive linear filtering of the excitation can be found. With a time-independent excitation, the acoustic load can create instabilities or not according to the filter poles. Using this method one can find again the minimal load to explain oscillations of either single blown-closed or blown-open reed but also the minimal description of the vocal tract needed for chromatical playing on diatonic harmonica. It is also possible to easily derive the numerical approximation of the acoustic load and reed(s) equations using a single numerical scheme, if a highly sufficient sampling frequency is chosen (maybe greater than 44.1 or 48 kHz), to perform numerical simulations of the studied acoustical problem.

3:05–3:15 Break

3:15

1pMUa4. Aeroelastic analysis of a closing reed of the mouth organ (harmonica). James F. Antaki, Jeongho Kim, Abhinav Singhal (Dept. of Biomedical Eng., Carnegie Mellon Univ., 700 Technol. Dr. Pittsburgh, PA 15219, antaki@cmu.edu), Greg Burgreen (Computational Simulation and Design Ctr., MS State Univ.), and Fangjun Shu (Mech. & Aerosp. Eng., New Mexico State Univ.)

A 3-D numerical simulation of the fluid-structure-interaction of a vibrating free reed was conducted representing an isolated closing reed of the mouth organ. Air flow was considered Newtonian and compressible. The reed was considered mildly viscoelastic. Both large eddy and SST turbulent models were evaluated. Independent variables included upstream resonant chamber dimensions (length and cross section), lateral reed clearance, and offset gap. Particular attention was given to the influence of the initial pressure impulse upon initiation of self sustained oscillations. Numerical results were corroborated by flow visualization within a transparent replica of an isolated harmonica cell using high speed videography (3 000 frames/s) with smoke tracer. These results provide guidelines for improving the responsiveness of the initial transient attack and avoidance of the phenomenon known as "reed choking."

3:35

1pMUa5. Contemporary composition for the traditional Lao free-reed mouth organ khaen. Christopher Adler (Music Dept., Univ. of San Diego, 5998 Alcala Park, San Diego, CA 92110, cadler@sandiego.edu)

The khaen is a bamboo free-reed mouth organ prominent among people of Lao ethnicity in Laos and Northeast Thailand and is likely related to ancestors of other Asian free-reed instruments as well as to Western free-reed instruments. New Musical Geographies is an ongoing project by composer/performer Christopher Adler to promote the khaen as a concert instrument in the Western contemporary concert music tradition by encouraging the composition of new works for the instrument. To date, the project includes 17 solo and ensemble compositions by six composers. The diverse compositional strategies of these composers and their relationships to traditional performance techniques and musical structures will be discussed. The presentation will include the performance of selected compositions and examples of traditional-style improvisations.

3:55

1pMUa6. Chromatic playing on a diatonic harmonica. Howard Levy (P.O. Box 5010, Evanston, IL 60204, harpkeys@hotmail.com)

The purpose of my demonstration is to show how a diatonic harmonica player can get all the chromatic pitches on an instrument not intentionally designed for this purpose. I will recount my participation in the research done by Henry Bahnson and James Antaki in the 1990s that led to a more complete understanding of this phenomenon, and will describe my discovery of overblows and overdraws which enable a diatonic harmonica player to get the so-called missing notes, not obtainable by conventional draw and blow bending techniques. I will demonstrate conventional note-bending techniques as well, and play samples of my playing in different styles making use of these techniques—Blues, Jazz, Ethnic music, Classical, etc., on the ten hole diatonic harmonica.

Contributed Papers

4:15

1pMUa7. Wall vibrations in air-driven free reed bamboo pipes. Miles Faaborg and James P. Cottingham (Phys. Dept., Coe College, 1220 First Ave., NE, Cedar Rapids, IA 52402, jmfaaborg@coe.edu)

In previous investigations of wall vibrations in bamboo pipes from Asian free reed mouth organs, modal frequencies and mode shapes of a number of pipes were measured. Measurements of pipe input impedance were made, some of which suggested possible changes occurring as a result of damping the pipe vibrations. [Cottingham, J. Acoust. Soc. Am. **114**, 2348 (2010)]. The goal of this current work is to study the vibration of the pipe walls for the mechanically blown reed-pipe combination. This was done for undamped pipes and pipes heavily damped with sand or other damping material. Measurements of the internal sound field for both damped and undamped pipes were made as well as measurements of pipe impedance. Effects of the damping of wall vibrations on the radiated sound and the sounding frequency were also explored. [Work partially supported by National Science Foundation REU Grant PHY-1004860.]

4:30

1pMUa8. The diatonic harmonica, pipe resonators, and the siren. Casey N. Brock (Austin Peay State Univ., Clarksville, TN 37044, casey.brock@hotmail.com) and James P. Cottingham (Phys. Dept., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402)

A number of measurements of reed motion and sound field have been made on a diatonic harmonica mounted on a fixed volume wind chamber. These include variation of sounding frequency with blowing pressure and the degree to which the sounding frequency and sound spectrum can be altered by attaching external pipe resonators. Differences were observed between the behavior of blow and draw reeds as well as the dependence of the results on whether the secondary reed in the reed chamber is allowed to vibrate. As noted by Helmholtz, at a simple level of analysis, the sound production of a free reed is similar to that of a siren, in both cases involving an air stream that is periodically interrupted. Our current results are compared with the results of measurements made in an earlier study of a siren in similar experimental configurations. [Work partially supported by National Science Foundation REU Grant PHY-1004860.]

MONDAY AFTERNOON, 31 OCTOBER 2011 — GARDEN SALON 1, 5:00 TO 6:00 P.M.

Session 1pMUb

Musical Acoustics: Mouth Organ Concert

James P. Cottingham, Chair
Dept. of Physics, Coe College, Cedar Rapids, IA 52402

Chair's Introduction—5:00

Following the lecture session on Acoustics of Mouth Organs, a concert will be held featuring Howard Levy on the harmonica and Christopher Adler on the khaen.

MONDAY AFTERNOON, 31 OCTOBER 2011 — SUNSET, 1:00 TO 2:05 P.M.

Session 1pSAa

Structural Acoustics and Vibration: Structural Acoustics and Vibration Distinguished Lecture

Dean E. Capone, Chair
Applied Research Lab., Pennsylvania State Univ., State College, PA 16804

Chair's Introduction—1:00

Invited Paper

1:05

1pSAa1. Modeling of vibration, sound, and stresses using the virtual source approach. Goran Pavic (Vib. and Acoust. Lab., Nat. Inst. of Appl. Sci. 20 Ave., A. Einstien, 69621 Villeurbanne, France, goran.pavic@insa-lyon.fr)

Analytical solutions of acoustical, vibration, or stress problems are available only for some simple structural systems, e.g., a rectangular simply supported plate or a parallelepipedic cavity. An usual way of finding a solution where complex systems are concerned is to apply field discretization in conjunction with some numerical methods such as FE or BE. Yet the use of analytical models is desirable for its simplicity and better physical understanding of phenomena concerned. On top of this, analytical models give the possibility of

assessing quantities like energy flow which are linked to higher spatial derivatives, the latter being difficult to model numerically. The method of virtual sources enables one to obtain analytical solutions of systems of rather simple but non-trivial geometry. A key advantage of the method is the full control over the computation error. The method consists in applying a layer of virtual sources to a simple mother system of known analytical solution. These sources are adjusted in such a way as to produce particular boundary conditions on a target part of the mother system. The target part can be given a complex geometry which cannot be directly treated analytically. The paper will be accompanied by a number of examples which illustrate the approach.

MONDAY AFTERNOON, 31 OCTOBER 2011 SUNSET, 2:20 TO 5:35 P.M.

Session 1pSAb

Structural Acoustics and Vibration: Assorted Topics on Structural Acoustics and Vibration

James E. Phillips, Chair
Wilson Ihrig and Associates, 6001 Shellmound St., Emeryville, CA 94608

Chair's Introduction—2:20

Contributed Papers

2:25

1pSAb1. Aperture extension for near-field acoustical holography applied to jet noise. Alan T. Wall, Kent L. Gee, David W. Krueger, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, alantwall@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC 28801)

Near-field acoustical holography (NAH) techniques are used to investigate noise source characteristics of high-power jets on military aircraft. Sound field reconstruction of large sources, measured with an aperture of limited size, may generally be performed with the use of patch NAH methods. Patch methods, such as statistically optimized near-field acoustical holography (SONAH), help to mitigate the effects of a truncated measurement aperture by avoiding the use of the spatial discrete Fourier transform operation. However, the lack of information outside the measurement aperture may lead to other errors, particularly when large propagation distances are required. Some missing data must be recovered to propagate beyond the immediate measurement region. Numerical aperture extension methods, in conjunction with SONAH, are employed to characterize high-power jet noise sound fields. These methods include complex interpolation and extension, analytic continuation, and in-plane holographic projection. [Work supported by Air Force SBIR.]

2:40

1pSAb2. Explicit modeling of cubic stiffness in large amplitude dynamic response of metal plate under impingement of solid rocket motor plume. Hvard Vold, Paul Blelloch, Allison Hutchings, and Nate Yoder (11995 El Camino Real, Ste. 200, San Diego, CA 92130, havard.vold@ata-e.com)

Determining the damping characteristics of launch vehicle structures has risen to a high priority issue, since dynamic response in modeling and simulations is directly related to assumed damping. An experiment was conducted wherein an instrumented flat plate was excited by hydrodynamic fringes of the plume of a solid rocket motor. Estimation of damping was performed by variations of maximum entropy parametric spectrum estimation. Equivalent damping ratios of 5% were observed, even though the damping was observed to be amplitude dependent, with lower damping ratios associated with lower amplitudes. Analytic simulations, with fixed damping ratios and large displacement and rotation geometric effects, showed similar characteristics to the measured test responses. The hypothesis was formed that the structure is dominated by cubic stiffness effects under the observed impingement conditions, and that the damping actually remains constant, effectively limiting the dynamic response and giving the appearance of an increased damping ratio. Since one can estimate velocity and displacement time histories by filtering measured acceleration time histories, a time domain maximum entropy model can be formulated. The cubic stiffness terms are linear parameters and all nonlinear terms being computed from the measured kinematic quantities. The estimated results will be compared with analytic simulations.

2:55

1pSAb3. Shock transmission and eigenvalue veering within a coupled system. Kiran Vijayan and Jim Woodhouse (Dept. of engineering, Univ. of Cambridge, Cambridge, CB2 1PZ, United Kingdom, kv247@cam.ac.uk)

The operation of dynamical systems in harsh environments requires continuous monitoring. Internal sensors may be used to monitor the conditions in real time. A typical example is the sensor and electronic components used in space structures which, especially during launch, are subject to huge *g* force. The paper will present an experimental and theoretical study on a simplified model used to analyze the possible cause of high acceleration on the enclosed sensors and equipments due to impulsive loading. The model system consists of two beams coupled using compliant connections. An impulse hammer excites one beam, and vibrations are transmitted to the indirectly driven beam. A theoretical model is developed using a Rayleigh–Ritz approach and validated using experimental results in both the frequency and time domains. Monto Carlo simulation was done with random masses positioned on the indirectly driven beam to determine the worst-case conditions for maximum peak acceleration. Highest acceleration levels were found when mode matching in the two beams led to veering behavior in the coupled modes. The results suggest guidelines for the detailed design of internal components of a structure exposed to shock loading from its environment. [The authors thank Schlumberger Cambridge Research for financial support.]

3:10

1pSAb4. Experiences in performing a high-intensity, direct-field acoustic test on a contamination sensitive system. Eric C. Stasiunas, Vit Babuska, Troy J. Skousen, and David J. Gurule (Eng. Sci. R&D, Sandia Natl. Labs., P.O. Box 5800, MS-0557, Albuquerque, NM 87185, ecstasi@sandia.gov)

A direct-field acoustic test (DFAT) was performed on a Sandia system in order to verify survival due to an acoustic environment of 146.7 dB OASPL. The DFAT technique' performed by surrounding a test article with a wall of speakers and controlling the acoustic input with a closed-loop control system' was chosen as the test method in order to meet a

critical schedule. In choosing this test method, other challenges became apparent, such as how to obtain the high-intensity acoustic levels and what occurs to that environment inside the bagged frame constructed to maintain a contamination-free system. In addition, the vast amounts of data measured during a single test necessitated a way for the test director to quickly visualize the acoustic environment, saving time and provide insight for input adjustments if necessary. Finally, even though the specified acoustic environment was successfully obtained, the results illustrated some drawbacks of the current DFAT method. This paper will detail the DFAT setup used to obtain the test specification, the effects of the contamination frame on the acoustic environment, the quick-look data program created for visual analysis of the acoustic field, and ideas for performing more diffuse DFAT tests in the future.

3:25

1pSAb5. The effect of ribbing and pressurization on the vibro-acoustic response of a turbulent boundary layer excited panel. Micah R. Shepherd and Stephen A. Hambric (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

One of the largest contributors to interior aircraft noise is the direct radiation from the wall panels excited by turbulent boundary layer (TBL) flows. The wall panels are designed with ribbing to increase their stiffness and strength while maintaining a relatively low weight. The effect of the ribbing on the vibro-acoustic response is examined for TBL flow excitation. Normal modes of a typical aluminum aircraft panel are computed and compared with and without the ribs. Wavenumber transforms of the mode shapes reveal increased sensitivity of the ribbed panel to high wavenumber excitation. TBL forcing functions are then converted into modal space and used to compute the radiated sound power of each panel. The increase in radiated sound power will be discussed in terms of wavenumber sensitivity. A preload due to pressurization is then applied to the ribbed panel and the modes, wavenumber spectrum and radiated sound power are recomputed. The pressurization causes a significant change in the modal content and subsequently the wavenumber spectrum and radiated power. The general effect of preload will be discussed in the context of interior aircraft noise predictions.

3:40

1pSAb6. Lab vibration complaints due to secondary or questionable indicators. Jack B. Evans and Chad N. Himmel (JEAcoust., 1705 W Koening Ln, Austin, TX 78756, Evans@JEAcoust.com)

Researchers complained about lab vibration in a new multi-floor academic laboratory building after observing surface ripples in water glasses on lab shelves. Structural vibration and low frequency noise can effect sensitive laboratory equipment, degrade experimental specimens and reduce staff efficiency and productivity. Investigatory observations and measurements were undertaken to determine vibration and noise conditions and develop a plan of mitigation. Permissible 1/3 octave Vibration Criteria (VC) and full-octave background noise Room Criteria (RC), including maximum limits for low frequency noise, had been used for building design. Central plant equipment, including pumps and compressors are on ground level. The facility is served by remote campus chillers. The building air handlers, exhaust fans, and boilers are in a penthouse mechanical equipment room and on roof. Structure borne vibration and airborne noise may be transmitted to laboratory spaces via building columns and beams, in pipes and ducts and through vertical duct and pipe shafts. Floor vibration and airborne sound spectra were measured during normal business hours (while the facility was occupied) at various locations for comparison with vibration and noise criteria. Results will be graphically shown on charts. Mitigation measures implemented by the facility management will be enumerated with subjectively determine results.

3:55–4:05 Break

4:05

1pSAb7. Vibrational and acoustical response of a railroad bridge with vehicle loading. R. Daniel Costley, Henry Diaz-Alvarez, and Mihan H. McKenna (U.S. Army Engineer Res. and Development Ctr., Geotechnical & Structures Lab., 3903 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil)

A finite element model has been developed for a Pratt truss railroad bridge located at Ft. Leonard Wood, MO. This model was used to investigate the vibration responses of the bridge under vehicle loading. Modeling results have been obtained for a single axle with two wheels traversing the bridge at different speeds. Superposition of multiple axles have been used to represent various combinations of locomotives and flatcars transiting the bridge. The analysis includes examining the vibrational response of the bridge. The output of the vibration response is used as an input to an acoustic FE model to determine the vibrational modes that radiate infrasound. The vibration and acoustic models of the railroad bridge will be reviewed and results from the analysis will be presented. Experimental results will also be presented.

4:20

1pSAb8. Effective control of machinery noise on offshore platforms. Arindam Ghosh (KBR, 601 Jefferson, Houston, TX 77002, arindam.ghosh@kbr.com)

Offshore noise control must consider operational noise levels in the Topsides work areas from the perspective of hearing conservation. At the same time, for human comfort, it must consider both direct and structure-borne noise transmission from the Topsides sources to the occupied spaces. This paper will summarize design stage case studies for controlling centrifugal compressor and water injection pump noise integrating commercially available outdoor and indoor noise modeling packages and statistical energy analysis software. Practicality and economics of achieving the specified noise limits will be demonstrated based on cost effectiveness and noise reduction capacity of mitigation measures such as acoustic insulation, acoustic blankets, in-line silencers, enclosures, anti vibration mounts, and damping cassettes. For centrifugal compressor noise control, controlling the piping radiated noise by acoustical insulation provides the greatest benefit. For water injection pump noise control, controlling the base plate radiated noise through viscoelastic damping proves most effective. Economic and technical barriers to effective employment of the advanced analysis tools to the field of offshore noise control will be discussed.

4:35

1pSAb9. Analytical vibration model for beam reinforced plate. Alexandre Sarda (Dept. of Mech. Eng., Centro Politécnico, P.O. Box 19011, Curitiba, PR 81531990, Brazil, pescador@ufpr.br)

Beam reinforced plates are typical components of offshore structures, as used in oil prospecting and production. The typical problem associated with this type of structure is the noise and vibration generated by machines and transmitted through the low damped structure. This vibration can propagate to the accommodation area and generate noise, which can generate stress to the crew. This work leads to develop deterministic models and solutions to be used on vibration levels estimation to beam reinforced plates. These models are used in several configurations, as an example L or T joined plates. The obtained results are then compared with finite element model for solution validating. This model will be used for calculating the coupling loss factors to be used in an statistical energy analysis.

4:50

1pSAb10. Energy scattering in weakly non-linear systems. Graham Spelman, Jim Woodhouse, and Robin Langley (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom)

The Chinese Tam-Tam exhibits non-linear behavior in its vibro-acoustic response. The frequency content of the response during free, unforced vibration smoothly changes, with energy being progressively smeared out over a greater bandwidth with time. This is used as a motivating case for the general study of the phenomenon of energy cascading through weak nonlinearity. Numerical models based upon the Fermi-Pasta-Ulam system of non-linearly coupled oscillators, modified with the addition of damping, have been developed. These were used to study the response of ensembles of systems with randomized natural frequencies. Results from simulations will be presented here. For un-damped systems, individual ensemble members exhibit cyclical energy exchange between linear modes, but the ensemble average displays a steady state. For the ensemble response of damped systems, lightly damped modes can exhibit an effective damping which is higher than predicated by linear theory. The presence of a non-linearity provides a path for energy flow to other modes, increasing the apparent damping spectrum at some frequencies and reducing it at others. The target of this work is a model revealing the governing parameters of a generic system of this type and leading to predictions of the ensemble response.

5:05

1pSAb11. The plane wave expansion method applied to thin plates. Ismael Oviedo-de-Julián (Facultad de Ciencias, Universidad Nacional Autónoma de México, México, D.F., México and Universidad Autónoma Metropolitana-Azcapotzalco), Rafael A. Méndez-Sánchez (Universidad Nacional Autónoma de México, 62210 Cuernavaca, Mexico), Betsabé Manzanares-Martínez (Universidad de Sonora, NAvojoa, México.), Felipe Ramos-Mendieta (Universidad de Sonora, Hermosillo, Sonora, Mexico), and Elsa Báez-Juárez (Universidad Autónoma Metropolitana-Cuajimalpa)

The plane wave expansion method refers to a computational technique in electromagnetics to solve Maxwell's equations. This method is popular in the photonic crystal community, because it is possible to obtain the band structure (dispersion-relation) of photonic crystals with specific forms. Since photonic crystals are periodic systems, the method expands all parameters crystal in Fourier series (plane waves). These expansions transform the wave equation in an eigenvalue equation. In this work, we will follow the line of previous investigations where the plane wave method was applied to the classical theory of thin plates. In particular the plane wave expansion method will be applied to the Mindlin equation. This equation describes the out-of-plane vibrations of thin plates in a better way—the classical theory of thin plates.

5:20

1pSAb12. Doorway states in elastic one dimensional systems. Rafael A. Méndez-Sánchez, Alfredo Díaz-de-Anda (Inst. de Ciencias Físicas, Univ. Nacional Autónoma de México, P.O. Box 48-3, 62251 Cuernavaca, Mor., Mexico, mendez@fis.unam.mx), Jorge Flores (Inst. de Física, Univ. Nacional Autónoma de México, México D. F., Mexico), Luis Gutiérrez (Inst. de Ciencias Físicas, Univ. Nacional Autónoma de México, Cuernavaca, Mor., Mexico), Guillermo Monsivais (Inst. de Física, Univ. Nacional Autónoma de México, México D. F., Mexico), and Alejandro Morales (Inst. de Ciencias Físicas, Univ. Nacional Autónoma de México, Cuernavaca, Mor., Mexico)

Whenever a state of a "distinct" and simple nature is immersed in a sea of states of a different, more complicated, structure a strength function phenomenon appears: the amplitude of the distinct state is spread over the complicated eigenstates with a Lorentzian-like shape. The distinct state acts as a doorway state whenever this state is coupled to the continuum and to the sea of complicated states, but the latter are coupled to the continuum only through the distinct state. We present numerical and experimental results for an elastic system which presents a doorway state and the temporal evolution of the phenomenon is measured directly for the first time. The delayed and stationary responses of the system are discussed.

MONDAY AFTERNOON, 31 OCTOBER 2011 ROYAL PALM 1/2, 1:00 TO 3:00 P.M.

Session 1pSP

Signal Processing in Acoustics: Underwater Acoustic Communications

Hee Chun Song, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 8820 Shellback Way, La Jolla, CA 92093-0238

Caglar Yardim, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

1:00

1pSP1. Sparse acoustic response function estimation with a mixture Gaussian model. Paul Gendron (Maritime Systems Div. SSC-Pacific, 53560 Hull St., San Diego, CA 92152)

Underwater acoustic response functions at high frequencies and large bandwidths exhibit significant spatio-temporal variability that depends greatly on volume, boundaries, and the source–receiver motion. Considered here is a mixture Gaussian assignment over Doppler, beam-angle, and channel bandwidth employed to describe the behavior of the sparse acoustic response function over received signal duration, aperture, and bandwidth. Accurate modeling of the dependence between the mixture components can be handled by considering dependence among neighboring indicator variables of the mixture assignment. This allows for a more accurate and adaptable description of the natural persistence that acoustic paths exhibit and improve channel estimation quality. This adaptive structure is applied to underwater M-ary spread spectrum acoustic communication transmissions during the MACE10 experiment of the coast of Martha's Vineyard at near 10 kHz of bandwidth and at ranges of 1 and 2 km. Posterior conditional expectations of the acoustic response are compared with least squares type estimates and performance is quantified in terms of the observed bit error rate (BER) as a function of received SNR. A BER = E-6 at rSNR = -16 dB for 12 element combining is demonstrated. [This work is supported by the Office of Naval Research and by NISE BAR.]

1:15

1pSP2. Multi-input multi-output multicarrier acoustic communications in shallow water. Taehyuk Kang, Heechun Song, and William Hodgkiss (Scripps Inst. of Oceanogr., 9500 Gilman Dr. La Jolla CA 92093-0238, tedkang@ucsd.edu)

Recently multi-carrier orthogonal frequency division multiplexing (OFDM) communications, popularly used in wireless channels, has been introduced in underwater acoustic channels with a large delay spread, as an alternative to the typical single carrier approaches. In this paper, we investigate multiple-input/multiple-output (MIMO) OFDM communications which can effectively increase the data rate in band-limited underwater channels. The performance of MIMO OFDM communications will be illustrated using the data collected from the KAM11 experiment conducted in shallow water, west of Kauai, Hawaii, which involved multiple transmit and receive arrays with different bandwidths, inter-element spacings, and apertures at various ranges between them up to 10 km.

1:30

1pSP3. Estimation of acoustic communication channel capacity of an ocean waveguide disturbed by surface waves. Thomas J. Hayward (Naval Res. Lab., Washington, DC 20375)

The ocean environment is known to present significant challenges to underwater acoustic communication, including noise-imposed bandwidth constraints and large-scale temporal dispersion due to macro-multipath propagation. In addition, acoustic interaction with the rough, moving ocean surface degrades the spatial and temporal coherence of acoustic communication signals, thereby diminishing the achievable data rates. In this work, the effects of the ocean surface on acoustic communication channel capacity are examined in a computational study for a shallow-water waveguide bounded by an ocean surface disturbed by gravity and capillary waves. Surface wave spectra are derived from empirical models [Donelan and Pierson, J. Geophys. Res. **92**, 4971] and are used to construct realizations of the 2-D moving surface. Then, 2-D and 3-D propagation models are applied to derive the space-time correlation properties of the acoustic channel, from which stochastic channel models are constructed. The adequacy of $N \times$ 2-D propagation modeling for this construction is also assessed. The channel models are then used as a basis for the estimation of channel spectral efficiency (achievable data rate per unit of frequency increment). Extensions to estimates of channel capacity are then discussed. [Work supported by the Office of Naval Research.]

1:45

1pSP4. Time reversal communications in a time-varying sparse channel. Hee-Chun Song (MPL/SIO, La Jolla, CA)

Recently, time reversal (TR) communications has been extended to time-varying channels. The basic idea is to implement it on a block-by-block basis such that within each block the channel remains time-invariant and subsequently is updated using detected symbols (decision-directed mode). Using experimental data (12–20 kHz) collected in shallow water, this letter investigates three different block-based TR approaches: (1) without explicit phase tracking, (2) with phase tracking, and (3) exploiting channel sparsity. The TR approaches then are compared to a conventional adaptive multichannel equalizer. It is found that approach (3) generally provides the best performance and robustness.

2:00

1pSP5. High frequency multiple-input/multiple-output time reversal acoustic communication. Aijun Song and Mohsen Badiey (114 Robinson Hall, Univ. of Delaware, Newark, DE 19716)

Current acoustic communication technologies using a single transmitter can only provide limited data rates due to the narrow bandwidth available in the underwater acoustic channel. Significant data rate increases can be achieved through the use of multiple-input/multiple-output (MIMO) systems in the underwater environment. However, in addition to the time-varying inter-symbol interference (ISI), co-channel interference (CoI) occurs as a result of multiple data streams sharing the channel at the same time and at the same bandwidth. Removal of both the CoI and ISI is a challenging problem in the underwater channel. This is especially true for high frequency acoustic communication (greater than 10 kHz). In order to achieve high data rate, reliable communication, time reversal MIMO processors have been developed. In the receiver, both the time-varying ISI and the CoI are addressed. Field data from recent high frequency acoustic experiments in the Pacific Ocean are used to demonstrate the receiver performance. Environmental impacts on acoustic MIMO communication will also be shown. [Work supported by ONR Code 322OA.]

2:15

1pSP6. Code division multiple access based multiuser underwater acoustic cellular network. T. C. Yang (Code 7120, Naval Res. Lab., Washington, DC 20375)

A multiple-access underwater acoustic cellular network is considered in this paper using direct sequence spread spectrum techniques similar to the code division multiple access cellular network in RF communications. To keep a reasonable data rate, in view of the limited bandwidth in an underwater (UW) channel, the length and, therefore, the number of orthogonal codes (the number of users) cannot be too large. At the same time, the orthogonality of the codes is severely degraded by the extended multipath arrivals in the UW channel. As a result, communication bit error rate becomes non-negligible when the interferer's signal energy (due to increasing number of users) becomes order of magnitudes higher than the desired signal [Yang *et al*. JASA **126**, 220–228 (2009)]. In this paper, techniques used in multiple-input-multiple-output communications are applied to underwater cellular network for simultaneous communications between the users and the base station, taking advantages of the (rich) spatial degrees of freedom of the UW channel. Simulation and experimental results will be presented. [Work supported by the U.S. Office of Naval Research.]

2:30

1pSP7. Underwater acoustic communication channel simulation using parabolic equation. Aijun Song, Joseph Senne, Mohsen Badiey (114 Robinson Hall, Univ. of Delaware, Newark, DE 19716), and Kevin B. Smith (Graduate School of Eng. and Appl. Sci., Monterey, CA 93943)

High frequency acoustic communication (8–50 kHz) has attracted much attention recently. At these high frequencies, vaHigh frequency acoustic communication (8–50 kHz) has attracted much attention recently. At these high frequencies, various physical processes, including surface waves, subsurface bubbles, and ocean volume fluctuations, can significantly affect the communication channel. While there is an on-going work, however, the research community is still lacking adequate numerical models that can provide realistic representations of both deterministic and stochastic channel properties in the dynamic ocean. Advancements in underwater acoustic communication technologies mainly rely on at-sea experiments, which are very costly. Our studies show that it is possible to simulate realistic communication channels through parabolic equation (PE) modeling. The Monterey-Miami PE model with an evolving sea surface has been used to generate time-varying impulse responses. Data from our recent experiments are used to evaluate the model in predicting acoustic communication performance. [Work supported by ONR Code 322OA.]

2:45

1pSP8. Resource allocation for orthogonal frequency division multiplexing (OFDM) underwater acoustic cooperative transmission with limited feedback. Xiaopeng Huang and Victor Lawrence (Dept. of Elec. and Comput. Eng., Stevens Inst. of Technol., Hoboken, NJ 07030)

In this paper, we first compare the difference between the wave cooperative (WC) protocol (designed for acoustic propagation) and amplify-forward (AF) protocol. In WC protocol, the relay nodes will transmit the received signal from the transmitter to the destination immediately (one time slot), in order to overcome the low transmission data rate problem in acoustic communication systems. Then we will propose the closed-form expression of the WC protocol with single relay. We will adopt the underwater acoustic (UWA) Rayleigh fading channel model and ambient noise in this cooperative transmission system. We propose capacity criterion based power allocation for UWA cooperative transmission (WC model) with the total transmitted power constraint, achieved under different levels of quantized CSI feedback using Lloyd algorithm. Meanwhile, limited feedback general procedure and Lloyd algorithm based codebook design will be presented. Simulation results will compare the system capacity with different bits of feedback, perfect feedback and non-feedback. In addition, the effect of relay locations on the system performance and the convergence speed of the Lloyd algorithm will be explored in the results.

Session 1pUW

Underwater Acoustics: Boundary Scattering From the Ocean Bottom or Surface

Grant B. Deane, Chair
Marine Physical Lab., Univ. of California, San Diego, La Jolla, CA 92093-0238

Contributed Papers

1:15

1pUW1. Reflection and refraction of sound on on smoothed boundaries. Nick Maltsev (1467 Leaftree Cir, San Jose, CA 95131)

Classic problems of reflection and refraction of sound waves on the boundary of two liquid half spaces are reformulated for more complicated and more realistic case, when acoustical properties of media change smoothly. Refraction and reflection of plane, cylindrical and spherical waves on smoothed boundaries is discussed. The propagation of waves in Pekeris waveguide with smoothed boundaries is also formulated and solutions for the point source are discussed. New formulas for rays and modes are derived for both cases.

1:30

1pUW2. Importance of surface forward scattering on reverberation level. Eric I. Thorsos, Jie Yang, W. T. Elam, Frank S. Henyey, and Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

Transport theory has been developed for modeling shallow water propagation at mid frequencies (1–10 kHz) where forward scattering from a rough sea surface is taken into account in a computationally efficient manner. The method is based on a decomposition of the field in terms of unperturbed modes, and forward scattering at the sea surface leads to mode coupling that is treated with perturbation theory. Transport theory has recently been extended to model shallow water reverberation, and the effect of forward scattering from the sea surface is found to be very important in accurately modeling the reverberation level under winter-like (nearly isovelocity) sound speed conditions. At a frequency of 3 kHz and a wind speed of 15 knots (7.7 m/s) with a fully developed sea, the reverberation level from sea surface scattering is found to be up to 20 dB greater than when a coherent reflection loss is used to account for the effect of surface roughness during the propagation out to and back from the primary backscatter source of reverberation. Examples illustrating the importance of sea surface forward scattering on reverberation level are shown. [Work supported by ONR Ocean Acoustics.]

1:45

1pUW3. Observation of ocean surface scattering in the deep ocean using a towed array. Stephen D. Lynch, Gerald L. D’Spain (Marine Phys. Lab., SIO, 291 Rosecrans, San Diego, CA 92106), Kevin D. Heaney (OASIS, Lexington, MA 02421), Arthur B. Baggeroer (MIT, Cambridge, MA 02139), Peter Worcester (SIO, La Jolla, CA, 92093), James Mercer (APL-UW, Seattle, WA, 98105), and James Murray (OASIS, Lexington, MA 02421)

Over the course of an experiment in the northern Philippine Sea in 2009, weather and sea surface conditions varied from calm and smooth to stormy and very rough. A ship with an acoustic source deployed at 15 and 60 m held station while a second ship towed Penn State’s Five Octave Research Array (FORA) at 120 m depth in an arc, maintaining constant range at the first convergence zone. With the source and receivers so near the ocean surface and with zero closing speed, these events offer an opportunity to extract information about the acoustic waves’ interaction with the temporally varying ocean surface. Additionally, the FORA was towed at various depths in a star pattern about the stationary source ship, and a vertical acoustic array was deployed near the source ship. Using meteorological data and a model for ocean waves given weather conditions and fully developed seas, these events offer an opportunity to separate in the data-adaptive beamformer output the arrivals scattered and Doppler shifted by the rough ocean surface while accounting for motions of the receiver and acoustic interactions with the bottom.

2:00

1pUW4. Predicting surface scattering from surface elevation time series. Grant B. Deane (Code 0238, Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu) and James C. Preisig (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Surface reverberation in mid-frequency bands (we are considering 3–15 kHz) can be an important determinant for the performance of underwater communications systems operating in surface scattering environments. At these frequencies and at relatively short ranges (order 10 water depths), gravity waves focus sound energy incident on the surface, creating intensifications, Doppler shifts and phase shifts in the reflected field, all of which impact the performance of underwater acoustic communications systems. Observations of the time-varying arrival intensity structure from an experiment conducted in the Martha’s Vineyard Coastal Observatory will be presented along with model calculations made using the Kirchhoff approximation. The sensitivity of the reflected field to the distribution of energy in the surface wave field and short wavelength gravity waves will will be discussed. [Work supported by the ONR Ocean Acoustics Program.]

2:15

1pUW5. Real time simulation of element level time series for active sonars in deep ocean environment. Sheida Danesh (Dept. of Mech. Eng., Ctr. for Ocean Eng., MIT, 77 Mass Ave., Bldg. 5-223, Cambridge, MA 02139, sdanesh@mit.edu)

The simulation of experiments in deep ocean environments is becoming increasingly important to researches due to expensive and complex modern equipment. Accurate simulations allow for more efficient use of ship time. LAMSS (Laboratory for Autonomous Marine Sensing Systems) at MIT has developed an indispensable tool for simulating real time active sonar experiments using a combination of ray tracing software (Bellhop), MATLAB, and MOOS-IvP (Mission Oriented Operating Suite - Interval Programming). A new key component to this toolbox is a module that models surface reverberation. Experiments in deep ocean environments are subject to environmental effects, such as the scattering of acoustic ray bundles on the ocean boundary due to upward refraction. Therefore, modeling surface reverberation is essential in deep ocean simulations when identifying target signatures in real time through active sonar. The implementation of this model is intended to simulate deep ocean active sonar experiments more accurately, thus optimizing on site data collection and evaluation.

2:30

1pUW6. Three-dimensional surface scattering using a parabolic equation model. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943, kbsmith@nps.edu), Mohsen Badiey, and Joseph Senne (Univ. of Delaware, Newark, DE 19716)

Calculations of acoustic rough surface scattering have previously been performed by numerous researchers using parabolic equation models in two

dimensions. Approaches have varied from exact solutions that re-map the rough interface to the pressure release boundary condition through a generalized method of images, approximate methods based on a perturbation approach valid for small slopes, and extended domain approaches that define the rough surface as a boundary between water and air with an additional absorbing boundary above the rough surface. In this work, we examine extensions of these approaches to the three-dimensional propagation. It is shown that the exact solution based on a generalized method of images does not yield realizable expressions for implementation. The other approaches are relatively easily adapted to three-dimensions, and some simple test cases involving Bragg scatter are analyzed for accuracy. Implications for other rough surface scattering model studies are also discussed.

2:45

1pUW7. Energy conservation in the Kirchhoff approximation. Sumedh M. Joshi and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

The scattering of sound from rough interfaces is frequently modeled using the Kirchhoff approximation. As has been shown by Lynch and Wagner [J. Acoust. Soc. Am. **47**(3)] and others, for the case of a pressure-release surface, the Kirchhoff approximation fails to conserve energy. In particular, Lynch and Wagner derive an analytical expression for the proportion of incident energy conserved for a surface with a Gaussian roughness spectrum. They demonstrate that energy is not conserved near normal incidence due to the failure of the Kirchhoff approximation to multiply scatter rays back into the upper half-space. In this work, a Monte Carlo technique is used to quantify the degree to which energy is not conserved in the three-dimensional Kirchhoff approximation; these results are compared with theoretical prediction of Lynch and Wagner for the Gaussian spectrum. A similar Monte Carlo analysis is undertaken for other roughness types. Finally, it is shown that the integral solution, a model that accounts for multiple scattering and shadowing, conserves energy in the pressure-release case. [Work supported by ONR, Ocean Acoustics.]

3:00

1pUW8. Crosscorrelation of broadband signals scattered by a random pressure-release surface. P. J. Welton (Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX 78713)

An expression for the crosscorrelation of omnidirectional, broadband acoustic signals scattered by the sea surface is derived for both vertically separated and horizontally separated hydrophones using an integral equation approach in conjunction with the Fresnel phase approximation, and the specular point approximation, i.e., the approximation that most of the scattered energy comes from regions close to the specular point. The resulting expressions are simple and have an obvious physical interpretation. In the limit as the surface becomes perfectly smooth, the expressions reduce to the image solution, thus ensuring conservation of energy.

3:15–3:30 Break

3:30

1pUW9. Fast direct integral equation-based analysis for acoustic scattering using non-uniform grid accelerated matrix compression. Yaniv Brick and Amir Boag (School of Elec. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, yaniv.brick@gmail.com)

An approach for the fast direct solution of scattering problems via compression of boundary element method (BEM) matrices is presented. Such an approach is advantageous for large resonant problems where iterative solvers converge poorly, or if the solutions are sought for multiple directions of incidence, so that the computational cost becomes proportional to the number of desired solutions. The compression is achieved by revealing the ranks of interactions between source and observation subdomains via algebraic analysis of the source subdomains' interactions with coarse non-uniform grids (NGs) surrounding the observation subdomains and vice versa. The NG field representation, originally developed for acceleration of iterative solvers [Y. Brick and A. Boag, IEEE Trans. Ultrason. Ferroelectr. Freq. Control **57**, 262–273 (2010)], is used to facilitate the computation of interacting and non-interacting mode sets that serve as a basis for a subsequent transformation of the BEM matrix. Only a highly compressed system linking the interacting modes is solved prior to the solution's extraction for the non-interacting ones. Further compression and acceleration of a multilevel scheme is suggested based on a "layered-localized" field representation via modes sorted according to their radiation, directivity, evanescence, and focusing properties, which are also computed using the NG approach.

3:45

1pUW10. Effects of random bottom bathymetry on temporal and spatial coherence in shallow water propagation. Jennifer Wylie, Felipe Lourenco, and Harry DeFerrari (Div. of Appl. Marine Phys., Univ. of Miami, 4600 Rickenbacker Cswy, Miami, Fl 33149, jwylie@rsmas.miami.edu)

Fixed system measurements for three experimental shallow water sites reveal systematic dependence of temporal and spatial coherence of individual mode arrivals on frequency and mode number that cannot be fully explained with internal wave variability. Here random fluctuations in bottom bathymetry are introduced to propagation models. Mode structures are randomized and coherence times and lengths decrease with increasing amplitude of bathymetry fluctuations for the same internal wave variations. Distinct mode features blur as frequency of pulse signals increased. Mode coupling and eventually a smearing of continuous modes are observed with increasing frequency and magnitude of bathymetry variations. For low frequencies and bathymetry variations constrained a small fraction of a wavelength, the bottom appears flat and nearly perfect discrete modes are formed. Modes are reinforced by specular reflections all along the path of propagation. Coherence depends entirely on internal wave fluctuations. All mode arrivals have nearly the same coherence times. But modes break up as height of bathymetry fluctuations become comparable to acoustic wavelength. Then sensitivity to sound speed fluctuation increases and coherence is reduced. Two dimensional bathymetry fluctuations account for observation out to 10–20 km and three-dimensional effects become important at longer ranges.

4:00

1pUW11. Characterization and scattering measurements from rock seafloors using high-resolution synthetic aperture sonar. Derek R. Olson (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dro131@psu.edu) and Anthony P. Lyons (The Penn State Univ., State College, PA 16804-0030)

Automated target detection systems are known to perform poorly in shallow water environments having high levels of reverberation and strongly varying scattering, such as rock outcroppings. Prediction of the scattering statistics is difficult because scattering from rock seafloors is not well understood. There is a lack of characterization methods, scattering strength measurements, and accurate approximate models for such interfaces. This research aims to measure the scattering strength of rock seafloors using an uncalibrated synthetic aperture sonar (SAS). An effective calibration has been made using a seafloor with a known scattering response. Scattering strength measurements will be used in conjunction with local slope information provided by interferometric SAS to characterize the rough interface.

4:15

1pUW12. Angular dependence of high-frequency seafloor acoustic backscatter (200–400 kHz). Christian de Moustier (HLS Res., Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037 cpm@hlsresearch.com), Gorm Wendelboe (RESON, AS, Denmark), Eric Maillard (RESON, Inc., Goleta, CA 93117), and Barbara J. Kraft (Barrington, NH)

Acoustic backscatter measurements were made with a RESON 7125-V2 multifrequency, multibeam sonar at 50 kHz increments between 150 and 450 kHz. The sonar was hull-mounted on a vessel held in a four-point moor in 17 m of water depth. The resulting constrained ship motion provided thousands of independent samples of the angular dependence of seafloor acoustic backscatter for grazing angles ranging from 90 deg to about 25 deg, over a well defined seafloor patch. Focused beamforming at all frequencies yielded fore-aft beam footprints with nearly constant width across the swath. Calibrated results are presented at 200 and 400 kHz, and relative results at all other frequencies. [Work funded by ONR-Ocean Acoustics (Code 32).]

4:30

1pUW13. A study on subcritical penetration into rough seafloor. Linhui Peng, Gaokun Yu (Information College, Ocean University of China, 238 Songling Rd., Qingdao, PRC), and Jianhui Lu (Ocean Engineering College, Ocean University of China, Qingdao, PRC)

Bragg scattering from rough interface of seafloor is one of the main causes of subcritical penetration of sound into seafloor. In this paper, the mechanism of subcritical penetration into rough seafloor is analyzed by Bragg scattering from sinusoidal fluctuation surface, and the condition that subcritical penetration can be induced is discussed. Because the refraction angle of minus order Bragg scattering waves is always smaller than refraction angle of Snell's penetration wave, the subcritical penetration attributes to the minus order Bragg scattering waves which propagate as normal plane wave below the critical grazing angle. So, in order to obtain the subcritical penetration, the minus order Bragg scattering wave should be considered detailed. The first order perturbation approximation is used for rough surface scattering usually. In this paper the validity of first order perturbation approximation for rough surface scattering is also checked by comparing the first order Bragg scattering wave with high order Bragg scattering waves.

4:45

1pUW14. Estimating seafloor roughness using synthetic aperture sonar image statistics. Anthony P. Lyons (Appl. Res. Lab., Penn State Univ., State College, PA 16804, apl2@psu.edu) and Shawn F. Johnson (Johns Hopkins Univ., Laurel, MD 21043)

In this work, we present a model to predict the impact of intensity scaling caused by random seafloor roughness on SAS image speckle statistics and the possible use of this model to estimate roughness parameters, such as root-mean-square height and slope. The continuous variation in scattering strength produced by roughness (i.e., roughness-induced changes in seafloor slope) is treated as an intensity scaling on image speckle produced by the SAS imaging process. Changes in image statistics caused by roughness are quantified in terms of an effective K-distribution shape parameter. Comparisons between parameter estimates obtained from the scaling model and estimates obtained from high-resolution SAS data collected in experiments off of the Ligurian coast near La Spezia, Italy, are used to illustrate the efficacy of the model. [Work performed under ONR Grants N00014-10-1-0051, N00014-10-1-0047, and N00014-10-1-0151].

Payment of an additional registration fee is required to attend this lecture see page A22

MONDAY EVENING, 31 OCTOBER 2011 — SUNRISE, 7:00 TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Acoustics of Green Buildings

Lily M. Wang, Chair
Architectural Engineering, Univ. of Nebraska, Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Acoustics of green buildings. Ralph T. Muehleisen (Illinois Inst,. of Tech., Civil and Architectural Eng., 3201 S Dearborn, Rm. 228, Chicago, IL 60616, muehleisen@iit.edu)

As the world realizes the considerable environmental impacts of building construction and operation, a strong movement to create more sustainable buildings, i.e., "green" buildings has developed. While energy and resource use of green buildings is reduced and some indoor environmental conditions are improved, the acoustics of green buildings are often ignored and are frequently worse than in conventional buildings. In this tutorial presentation, the sustainable building movement is reviewed, the major design differences between green and conventional buildings are explained, and the impact of green design on acoustics is explored. There will be animations and auralizations to help attendees better understand acoustic impacts of green building design choices. Finally, lest ye think all is bad, the presentation will discuss solutions to the green building acoustic problem and present some of the positive impacts of green buildings can have on acoustics.

Session 2aAAa

Architectural Acoustics and Psychological and Physiological Acoustics: Architectural Acoustics and Audio II

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

Alexander U. Case, Cochair
Fermata Audio & Acoustics, P.O. Box 1161, Portsmouth, NH 03802-1161

Invited Papers

8:00

2aAAa1. When an idea goes in a different direction from that expected. Thomas Plsek (Berklee College of Music, MS-1140 BRASS, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu)

Quite a few years ago when airlines were using the simple headsets with the rubber tubes, it was realized that they could be coupled to a trombone practice mute to make a system that one could use to practice without disturbing others. As it happened, the idea did not for various reasons work as expected. It was realized that a new instrument could be created by placing the headphones on someone other than the performer. By doing so, it was realized that the acoustic feedback normally received by the performer would be eliminated, the performance space and its acoustic qualities was made irrelevant, and the listener was given a unique perspective shared with no one. Over the past few years, this instrument was used to perform for numerous people. This paper explores and summarizes those adventures.

8:20

2aAAa2. Don't get caught with your mics open. Deborah J. Britton (K2 Audio, LLC 4900 Pearl East Circle, Ste. 201E, Boulder, CO 80301)

Designing sound systems for legislative facilities often presents numerous challenges. From creating intelligible sound reinforcement and broadcast feeds in highly ornate, reverberant spaces to using digital signal processing to ensure that "off-the-record" side conversations are not made public, many different elements contribute to the final design. This presentation discusses a few of these specific challenges and how they were overcome.

8:40

2aAAa3. Methods for separating harmonic instruments from a monaural mix. Mert Bay and James W. Beauchamp (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801, mertbay@illinois.edu)

During the past decade, solutions to the problem of musical sound source separation have become more evident. Potential applications include sound editing, enhanced spatialization, music-minus-one, karaoke, music classification/identification, music transcription, and computational musicology. Our current approach is to restrict the input signal to a mix of a limited number of instruments, each comprised of harmonic partials, with known F0 contours. (These can be obtained either by audio-to-midi alignment or multiple-F0 estimation.) Since harmonic frequencies of known F0s are easily predicted, binary mask separation is robust except for frequency regions where harmonics of different instruments collide. Three methods for repairing collisions are compared: (1) F0-informed non-negative matrix inversion using instrument spectral libraries; (2) least-squares estimate of collision frequencies based on a multiple sinusoidal model; and (3) common amplitude modulation (CAM) [Li *et al.*, IEEE Trans. Audio, Speech, Lang., Process. **17**(7), 2009]. Separation examples using these methods will be demonstrated.

9:00

2aAAa4. Would a rose sound as sweet. Sam Ortallono (MediaTech Inst., 3324 Walnut Bend Ln., Houston, TX 77042)

Would a rose sound as sweet? In this presentation, we will explore perception of individual microphones. The same vocalist will be recorded with five different microphones. Volunteers will be presented with two sets of tracks to evaluate. The first set of tracks will be labeled only with letters, microphone A, B, C, D, and E. The second set of tracks will be labeled with the names of the microphones. Then the subjects will be asked to rate subjective qualities of both sets. Each person will be their own control by judging one set blind and one with the knowledge of the name of the microphone. Will preconceived notions of a microphonex's reputation change the outcomes?

9:20

2aAAa5. Binaural room acoustics. William M. Hartmann, Brad Rakerd, and Ryan A. Cook (4208 BPS Bldg., Michigan State Univ., East Lansing, MI 48824)

Room acoustics concerns the geometrical and materials properties of rooms as they determine measurable sound field parameters that are perceptually important to listeners. For example, the volume and absorption in a room determine the reverberation time, which is important to the comprehension of speech and the enjoyment of music. Binaural room acoustics concerns the properties of rooms as they affect measurable interaural parameters, important to the human binaural perceptual system. For example, room properties affect the short-term interaural cross-correlation, which is important to the perception of apparent source width. Our recent work emphasizes room effects on steady-state interaural level and phase differences, important to sound localization in the horizontal plane. Particularly, we have sought to give mathematical meaning to a "binaural critical distance." If a sound source and a listener are separated by less than the binaural critical distance, there is good probability that the interaural differences correctly indicate whether the source is on the listener's left or right. Experimentally, we have focused on sine tones in the range 200–1200 Hz, but we expect the results to be more generally applicable. [Work supported by the AFOSR (grant 11NL002) and by the NSF REU program.]

9:40

2aAAa6. Why reverberation affects different individuals differently. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., 677 Beacon St., Boston, MA 02215)

Any self-respecting architectural acoustician understands that good reflected sound enhances all sorts of important sound attributes. He or she undoubtedly also understands that reflected energy distorts the fine spectrotemporal structure of sound reaching a listener's ears. But what are the perceptual consequences of such minor acoustic distortions of sound detail, and what is the significance of this distortion when designing listening spaces? In psychophysics, recent experiments show that fine spectrotemporal structure is very important for understanding the content of complex, natural sounds, like speech, especially in the presence of background noise or competing sound. This talk will review why spectrotemporal structure is so important for verbal communication in everyday settings. Building from behavioral and physiological measures, this review will explore why even modest amounts of reverberant energy can dramatically impair the ability to communicate for some individuals, even when it causes no such ill effects in others. Understanding such effects is critical when designing listening spaces in which verbal communication is a key activity, like in a classroom.

10:00

2aAAa7. Verb, reverb, and re-reverb. Jim Anderson (Clive Davis Inst. of Recorded Music, New York Univ., 194 Mercer St., New York, NY 10012)

In a contemporary recording, artificial ambience, reverberation of varying length and degree, is added to compliment the various elements in a recording. This mixing condiment comes in many flavors: live chamber, plate, and digital. The application of reverberation is an art, as well as science, and its treatment can be a matter of taste, good, or bad. While it can be difficult to achieve a natural effect with one reverb in a recording, parallel reverbs can be used effectively. As a live performance takes place in a single reverberant hall, its recorded counterpart may take advantage of the application of different multiple reverbs.

10:20–10:35 Break

10:35

2aAAa8. Highly dynamic reverb—The recording studio always has what the performance hall sometimes wants. Alex Case (Sound Recording Tech., Univ. of Mass Lowell, 35 Wilder St., Lowell, MA 01854, alex_case@uml.edu)

The orchestra wishes to perform in a space whose acoustics offers flattering reverberation. That reverb is generally unchanging, from the beginning to the end of the performance. And that reverb must be "one size fits all," as every instrument in the band is offered the same acoustic treatment. Not so in recorded music. When reverb is generated acoustically, mechanically, and digitally and introduced into the music by way of a mixing console or digital audio workstation, under the watchful ear of the creative recording engineer, highly variable reverb becomes possible. Momentary reverb, dynamic reverb, multiple contrasting reverbs, and still more variations not easily realized in a physical space are found common in popular music recordings, demonstrated and discussed here.

10:55

2aAAa9. Stonehenge-like auditory illusion evoked by interference pattern. Steven J. Waller (Rock Art Acoust., 5415-8 Lake Murray Blvd., La Mesa, CA 91942, wallersj@yahoo.com)

Blindfolded participants hearing an interference pattern attributed the dead zones to the presence of acoustic obstructions in an arrangement reminiscent of Stonehenge. Without prior mention of interference or Stonehenge, participants were brought blindfolded to an open field with two English flutes atop thin 1 m tall wooden stands 2 m apart, connected via tubing to an air pump, giving a sustained pitch of C#5 (1 100 Hz). One by one, participants were led blindfolded in a 7.6 m radius circle around the pair of flutes. Afterward, faced away from the setup, they were instructed to remove blindfolds and independently "draw and describe what you thought was between you and the noise." Although there actually were no physical obstacles, the resulting drawings from all three participants represented arrangements of objects having many characteristics in common with Stonehenge; verbal descriptions included "pillars," "solid objects," "openings," "tall vertical slats." These pilot results demonstrate regions of low sound intensity due to destructive interference of sound waves from musical instruments can be misperceived as an auditory illusion of acoustic shadows cast by a ring of large obstructions. Measurements of acoustic shadows radiating out from Stonehenge are consistent with the hypothesis that interference patterns served as blueprints for the design.

Contributed Papers

11:15

2aAAa10. A new generation of time variant electro-acoustic enhancement systems. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

In 1991, Dr. David Griesinger presented a paper at the 90th AES convention [preprint 3014] describing a new method for improved electronic acoustic enhancement, and hundreds of these systems have been installed throughout the world. This paper describes a new approach to both hardware and software that comprises a third generation system. The new system retains the essential features of multichannel time variance with low pitch alteration, but updates the algorithms to utilize the superior DSP power of modern computers. Multiple reverberators, equalization, and matrix distribution can be combined into a single, reliable, and easily replaceable piece of hardware. Overall system gain and equalization can be calibrated automatically anytime; parameters in the hall have changed. The new algorithm combines the efficiency of infinite impulse response filters with direct convolution to produce a time-variant structure with extremely low coloration. The algorithm also includes a novel 32 band antifeedback circuit that further reduces coloration when the system is installed in a room. Unlike much of the competition, this system architecture allows the strength of the reverberation to be adjusted independently of the reverberation time, which makes it possible to simultaneously optimize both clarity and reverberance.

11:30

2aAAa11. Motion simulation in the environment for auditory research. Braxton B. Boren (Dept. of Music and Performing Arts, New York Univ., 34 W. 4th St., New York, NY 10012) and Mark A. Ericson (Army Res. Lab., Aberdeen Proving Ground, MD 21005)

Virtual sound source motion has been implemented in the Army Research Laboratory's Environment for Auditory Research, which contains a 57-channel spherical loudspeaker array located in an anechoic chamber. Using the Psychophysics Toolbox Version 3 low-latency PortAudio API, 57 channels of streaming audio are dynamically updated in real-time using MATLAB for signal processing. Both distance-based amplitude panning (DBAP) and vector base amplitude panning (VBAP) have been implemented in MATLAB for controlling source motion. Sources are defined on a given path, such as a circle, ellipse, or the "dog bonex"; flight pattern often used in aviation. While DBAP works convincingly for virtual sources located on the sphere defined by the loudspeaker array, VBAP is needed to position sources outside the array. Source motion paths are defined parametrically with respect to time, and playback buffers update the panned position every 11.6 ms. Based on the source's instantaneous distance, diffuse-field or free-field amplitude attenuation is added in MATLAB as well air absorption filtering. This system will be used for a variety of audio simulations and auditory experiments.

11:45

2aAAa12. Whales and poets—Two tales of techniques for adapting, enhancing, and fictionalizing acoustic traits. Carl P. Giegold, Kevin T. Watson, Molly K. Norris, and Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd. Ste. 815, Chicago, IL 60604)

Enhancement techniques are employed in two vastly divergent venues in Chicago with the purpose of removing barriers to communication. In the first case, the marine environment scaled to house Beluga whales presents its own acoustical obstacle. In the second, the presence of a microphone can be an obstacle as seen by a poet with a message to share. The Poetry Foundation implemented a voice lift system in a sonorous environment that supports the voice or light amplification, but requires assistance to extend the reach of the unamplified talker. Natural acoustic design, design of traditional public address systems, and the introduction of enhancement techniques are all outlined for these two case studies to illustrate how the roles intersect. Both environments are live in their un-enhanced state. The unusual aspect of these projects is the use of enhancement to increase intelligibility within a naturally reverberant space.

TUESDAY MORNING, 1 NOVEMBER 2011 SUNSET, 8:00 A.M. TO 12:00 NOON

Session 2aAAb

Architectural Acoustics, Noise, and Signal Processing in Acoustics: Accuracy of Absorption, Scattering, and Diffusion Coefficients

Peter D'Antonio, Chair
RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Chair's Introduction—8:00

Invited Papers

8:05

2aAAb1. New developments in the measurement and calibration of absorption, scattering, and diffusion coefficients. Peter D'Antonio and Brian Rife (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Round robin experiments have confirmed that current methods to measure the random incidence absorption coefficient are inaccurate and not reproducible from lab to lab. As a result, various improvements to ISO 354 are being considered. Among them include measures to prevent the coefficient from exceeding unity and achieve reproducibility, using calibration. Since all measurement equipment is calibrated, it seems prudent to calibrate the rev room as well. We will describe approaches used at RPG to replace suspended clouds with boundary diffusors, the use of Eyring instead of Sabine and calibration, using a non-diaphragmatic reflective sample, with the same area, thickness and perimeter as the absorber, as a "zero" or low absorption reference. The new rotating table ISO 17497-1 scattering coefficient apparatus, using dual source/simultaneous impulse response measurements will be described. Since the technique utilizes subtraction of the reflective table as a reference, some of the problems associated with the rev room method are mitigated and reproducibility is achieved. Finally, the new diffusion coefficient goniometer, utilizing 32 simultaneous impulse response measurements, will be described. As with the scattering coefficient, the normalized diffusion coefficient benefits from the use of a reflective calibration reference, thus removing edge diffraction and other effects.

8:25

2aAAb2. The sound absorption measurement according to ISO 354. Martijn Vercammen and Margriet Lautenbach (Peut, P.O. Box 66, Mook ZH-6585, Netherlands, m.vercammen@mook.peutz.nl)

Sound absorption measurements of building materials such as sound absorbing ceilings and other products are performed in a reverberation chamber according to ISO 354. It is known that the interlaboratory reproducibility of these measurements is not very well. At this moment, the differences of results between laboratories are much larger than can be accepted, from a practical point of view for predictions as well as from a jurisdictional point of view. An ISO working group has started to investigate possibilities to improve the method. Due to the insufficient diffuse sound field in a reverberation chamber with test sample, the shape of the reverberation room and the placing of diffusers will influence the result. A round robin research containing ten laboratories is performed to get information on the spread and if it is possible to reduce this by correcting for the mean free path or by application of a reference material. Additional measurements are performed to improve the measurement conditions such as measurements with volume diffusers. Possible improvements of ISO 354 will be presented. These consist of a procedure to qualify laboratories based on the statistical variation of the reverberation time and based on the results of a reference absorber.

8:45

2aAAb3. Diffusivity of diffusers in the reverberation room. Margriet Lautenbach (Peutz BV, Paletsingel 2, P.O. Box 696, 2700 AR Zoetermeer, m.lautenbach@zoetermeer.peutz.nl)

The random incidence absorption coefficient is measured in a reverberation room according to the ISO354 or ASTM C423-09. According to these standards, the diffusivity of a reverberation room is usually obtained with panel diffusers. Besides the fundamental problem that a reverberation room with a highly absorptive specimen is not diffuse, these panel diffusers introduce a number of uncertainties like the acoustical effective volume and the total boundary surface of the reverberation room. This might be one of the causes that some laboratories are structurally able to measure absorption coefficients larger than 1, even if the volume of the specimen, edge absorption, and the absorption of the surface covered by the specimen are taken into account. To reduce the difference in measurement results between different laboratories, the possible use of volume diffusers instead of panel diffusers is investigated. The following criteria are investigated to substantiate the hypothesis that volume diffusers lead to better results: (1) Deviation between microphone-source positions. (2) Comparison to maximum relative standard deviation (ASTM). (3) Comparison to theoretical variance. (4) Influence of place of specimen. The investigations have been performed in a 1:10 scale model. The results are presented in this paper.

9:05

2aAAb4. Measuring absorption: Bad methods and worse assumptions. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., SDP\#8, Elma, WA 98541, audio_ron@msn.com)

Ever since Sabine described the use of absorptive materials to affect reverberation time, the acoustic community has been trying to quantify this effect. Standards like C-423 and ISO-354 have been developed to aid in that process. These standards, describing coefficients ranging from 0.0 to 1.0, have been inadequate to fully describe the actual absorption of the tested materials. It is not uncommon to find actual measurements that result in coefficients exceeding the number 1.0. When this happens, assumptions are made that other properties of the material are not taken into account by the coefficient. The present standards also suffer from inadequacies of methodology. They do not measure the full range of absorptive qualities. This paper describes the incorrect assumptions about absorption and measuring it as well as illustrates the incorrect methodologies that are used in these standards. Because of the flaws in the assumptions and the inherent defects in the methodologies, this author believes the current standards need to be replaced by new standards. Replacement instead of correction is preferable because of the extremely large database of materials already measured and the difficulty explaining how new measurements based upon corrections would be comparable to old measurements.

9:25

2aAAb5. On the reproducibility of measuring random incidence sound absorption. Anthony Nash (Charles M. Salter Assoc., 130 Sutter St., San Francisco, CA 94104, anthony.nash@cmsalter.com)

For over 50 years, the American Society for Testing and Materials (ASTM International) has promulgated a method for the laboratory testing of random-incidence sound absorption coefficients (Test Method C423). This test method falls under the purview of ASTM Committee E33 (Environmental Acoustics). In 1999, the protocols in this test method became significantly more stringent with the goal of improving inter-laboratory "reproducibility" (i.e., quantitative differences among laboratories when testing the same specimen). ASTM calls such an inter-laboratory test a "round robin"; its outcome is an array of computed precision (i.e., uncertainty) values. This paper presents results from several "round robin" evaluations and discusses some possible causes for the range of values. If time permits, the fine points of the test protocols in C423 will be described and compared to those in ISO 354.

9:45

2aAAb6. Energy-based measurements in reverberation chambers. Timothy W. Leishman, Buye Xu, Scott D. Sommerfeldt, and Nicholas J. Eyring II (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, twleishman@byu.edu)

Standardized reverberation chamber measurements, including those of absorption coefficients, scattering coefficients, and sound power, rely on the acquisition and processing of squared acoustic pressure (or potential energy density) from many sound field locations. Kinetic energy density, total energy density, and the newly defined generalized energy density exhibit greater spatial uniformity in enclosed fields (especially at lower frequencies), which makes them more favorable quantities for these measurements. They produce greater accuracy or require fewer source-receiver positions to produce acceptable levels of accuracy. This paper explores their use and provides several analytical and experimental results to demonstrate their benefits.

10:05–10:30 Break

10:30

2aAAb7. Error factors in the measurement of the scattering coefficient in full and small scales. Tetsuya Sakuma and Hyojin Lee (Grad. Sch. of Frontier Sci., Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa 277-8563, Japan, sakuma@k.u-tokyo.ac.jp)

The reverberation room method for measuring the random-incidence scattering coefficient was standardized by ISO 17497-1 in 2004, and now the amendment is being discussed on the specimen mounting and the turntable speed. Regarding the former point, destructive diffraction from an uneven perimeter of a specimen is an error factor to overestimate the scattering coefficient; however, it can be suppressed by setting a border around the turntable. Regarding the latter point, special attention is needed to the combination of the turntable speed and the signal period of MLS in the impulse response measurement. Basically, the turntable speed should be limited by an angular step of 3° to 6° for one signal, thus 60 to 120 signals are required for one revolution. In practice, in order to suppress the time variance, a shorter signal period is preferred as far as the measured reverberation time is guaranteed. However, a best choice of the signal period depends on reverberation rooms in full and small scales, additionally depending on frequency bands. Some data measured in full and small scales demonstrate the effect of the combination on measurement accuracy.

10:50

2aAAb8. Uncertainty factors in determining the random-incidence scattering coefficient. Markus Müller-Trapet and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr 50, D-52066 Aachen, Germany, mmt@akustik.rwth-aachen.de)

When determining the random-incidence scattering coefficient with measurements that have been carried out according to ISO 17497-1, several quantities from four different measurements have to be combined to finally produce the sought-after coefficient. These include the reverberation time and atmospheric conditions to account for air absorption. As in most measurement situations, one has to keep in mind that uncertainties have affected the measurements, so that an absolute statement about the scattering coefficient might not be possible. This work will give an insight as to how each of the mentioned measured parameters and their uncertainty will affect the evaluation of the scattering coefficient. The "Guide to the Expression of Uncertainty in Measurement" will be employed to investigate the effect of both systematic and random errors. Within the limits of applicability, intervals for the necessary accuracy of the measured parameters that lead to an acceptable uncertainty in the final result will be given.

11:10

2aAAb9. Diffusion and scattering: An improved method of measuring both at the same time. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd. SDP#8, Elma, WA 98541, audio_ron@msn.com)

ISO-17497-1 and ISO-17497-2 describe methods of measuring the properties of diffusers and scattering devices. The result of these measurements is a description of the quality of the diffuser and the quantity of energy scattered. This paper will describe an improved method of measurement that enables both properties being simultaneously measured using the same procedure. The hope is that this method will produce a simplified database mimicking the directivity measurements of sound sources for use in simulation programs and in comparing various diffuser designs.

Contributed Papers

11:30

2aAAb10. Predicting scattering at the mid-frequency range. Carl R. Hart and Siu-Kit Lau (Durham School of Architectural Eng. and Construction, Univ. of Nebraska—Lincoln, 1110 S. 67th St., Omaha, NE 68182, carl.hart@huskers.unl.edu)

Predicting scattering characteristics of isolated surfaces is problematic in the mid-frequency range, due to high computational costs. In the mid-frequency range, techniques such as the finite element method or the boundary element method require extremely fine mesh structures, leading to the solution of very large matrices. A gap exists for predicting acoustic scattering in the mid-frequency range with low computational costs. A geometric acoustic time domain method, incorporating diffraction, is proposed to bridge the gap in current prediction methods. The geometric acoustic method utilized is adaptive beam tracing. Adaptive beam tracing is advantageous, compared to other geometric methods, since it does not generate aliasing errors and faulty image sources, which are shortcomings of ray tracing and the image source method, respectively. The scattering characteristics of a periodic rigid geometry are investigated. Relationships between geometry dimensions and frequency of excitation are studied. Furthermore, the importance of accounting for single diffraction or multiple diffractions will be discussed. The spatial, temporal, and frequency characteristics of the scattering geometry are quantified.

11:45

2aAAb11. Repeatability and reproducibility concepts in acoustical laboratories. John LoVerde and Wayland Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

There are many categories of variability in acoustical laboratory testing. The repeatability of acoustical testing is usually defined based on repeated testing of the same assembly over a short time period. Other variability, such as rebuilding an assembly in the same lab or building a nominally identical assembly in a different lab, is generally termed reproducibility. A previously undocumented variability is what may be termed "long term repeatability," the repeated testing of the identical specimen in the same laboratory with the same equipment and operator, but over an extended time period. The authors recently had the opportunity to measure the long term repeatability of a wood joist floor/ceiling assembly that remained in the same test chamber at an accredited acoustical laboratory over 50 days. The results are compared with previously known variability, and the implications for manufacturers and designers are discussed.

Session 2aAB

Animal Bioacoustics: General Topics in Animal Bioacoustics I

John A. Hildebrand, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

8:15

2aAB1. Bird's singing codification using quick response barcode. Hugo Fernando Jácome Andrade (Dept. of Sound & Acoust. Eng., Univ. of the Americas, E12-41 Colimes St. and Granados Ave., Quito, Pichincha, hjacome@udla.edu.ec) and David Parra Puente (Univ. Catholic of Ecuador, Quito, Pichincha)

This work responds to the necessity to objectively describe bird's songs in scientific publications and field guides. Nevertheless, it can also be extended to the study of other animal like marine mammals, bats, humans, and insects. The project consists on a computer application development in MAX/MSP language, that extracts coefficients from an audio signal, those coefficients will characterize the signal, and are codified by QR tool (Quick Response Barcode), created by the Japanese company Denso-Wave in 1994. The following stage of the project, that is in developing process, is the programming of the de-codifier application, that rebuild the audio signal from those numbers, to bring back the signal to the human hearing field, as final stage of the project, it will be developed a similar application with the same intentions for mobile phones with the built-in QR reading tool.

8:30

2aAB2. Correlation detection of bat echolocation calls. Mark D. Skowronski (Dept. Speech Lang. and Hearing Sci., Univ. of Florida, Gainesville, FL 32611, markskow@hotmail.com)

In an experiment comparing detection of echolocation calls between humans and automated methods, an optimal linear detector (matched filter) out-performed humans by 5 dB and model-based automated methods by 9 dB [Skowronski and Fenton, J. Acoust. Soc. Am. **125**, 513–521 (2009)]. While optimal linear performance cannot be achieved in practice, near-optimal performance may be reached if the detection filter nearly matches the target calls. Bat calls from a species are fairly stereotypic, so designing a test call (or bank of test calls) for a correlation detector is feasible. Model-based detectors collect information over short analysis windows and piece that information together to form a detection decision, a bottom-up strategy that is good at collecting local information but poor at arranging local information to form a global detection decision. By contrast, a correlation detector operates at one time scale, the global call duration, and may be considered a top-down strategy of detection. The strengths and weakness of a correlation detector are discussed, including designing filters for a given species, the tradeoff between filter bank size (computational cost) and detector sensitivity, species classification, and the effect of filter bank size on false positive rate.

8:45

2aAB3. Automated extraction and classification of contours in humpback whale vocalizations. Helen H. Ou (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, ohui@pdx.edu), Whitlow W. L. Au, Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kaneohe, HI 96744), Adam A. Pack (Univ. of Hawaii at Hilo, Hilo, HI 96720), and Lisa M. Zurk (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR 97201)

Humpback whales produce songs which consist of a sequence of short, continuous sounds known as units. This paper introduces an automated algorithm to extract the unit contours. An unsupervised classification is developed to provide a set of distinct units of the singing group. The analysis is performed on the vocalization spectrograms, which are normalized and interpolated into a squared time-frequency image. Unit contours are detected using two edge detection filters capturing sharp changes in the image intensities. The algorithm generates a group of rectangular image segments each containing a single contour unit, with the pixels outside the contour edge lines set to zero. The contours are compared with one another to identify distinct units. The comparison is quantified using parameters including the contour pixel intensity correlation, contour area, frequency range, and frequency of the peak pixel. A pairwise comparison provides a coarse division of classes, where each class is then represented by a candidate unit. The candidate units are compared with one another, and the ones with low similarity are advanced to the final set. The algorithm has been tested on humpback whale songs obtained during the winter season in Hawaiian waters in 2002 and 2003.

9:00

2aAB4. Noise reduction for better detection of beaked whale clicks. Yang Lu, Holger Klinck, and David M. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, luya@onid.orst.edu)

We seek to improve the signal-to-noise (SNR) ratio of clicks recorded from Blainville's beaked whales (*Mesoplodon densirostris*). The proposed filter is based on subspace principles and projects the noisy speech vector onto signal and noise subspaces. An estimate of the clean signal is made by retaining only the components in the signal subspace. To test the filter, a detector using the filter output has been designed for the detection of beaked whale calls. Through simulations, the proposed detector is shown to be capable of detecting most of the desired clicks but is not able to differentiate other co-existing species such as Risso's dolphins and pilot whales' that is, it efficiently detects all clicks. By combining the proposed detector with the energy ratio mapping algorithm (ERMA; Klinck and Mellinger (in press), which measures energy differences between different species, higher detection accuracy for beaked whale clicks can be achieved. The filter can also be used to improve the SNR of other marine mammal acoustic signals.

9:15

2aAB5. Spatial orientation of different frequencies within the echolocation beam of a *Tursiops truncatus* and *Pseudorca crassidens*. Stuart Ibsen (Dept. of Bioengineering, Moores Cancer Ctr., Univ. of California San Diego, 3855 Health Sci. Dr. # 0815, La Jolla, CA 92093-0815), Paul Nachtigall (Univ. of Hawaii, Kailua, HI 96734), Jacqueline Krause-Nehring (Alfred Wegener Inst. for Polar and Marine Res., Am Handelshafen 12, 27570 Bremerhaven, Germany), Laura Kloepper, and Marlee Breese (Univ. of Hawaii, Kailua, HI 96734)

A 2D array of hydrophones was created to determine the spatial distribution of frequencies within the echolocation beam of a *Tursiops truncatus*. This *Tursiops* was shown previously to only pay attention to frequencies between 29 and 42 kHz while echolocating. It was found that the 30 kHz frequency was tightly focused, and the spatial location of the

focus was consistently pointed toward the target. At 50 kHz, there was less focusing and less precision in being pointed at the target. At 100 kHz, the focus was often completely lost and was not pointed at the target. This indicates that this dolphin was actively focusing the frequencies it paid attention to toward the target, while the frequencies not paid attention to were left unfocused and undirected. This focusing was probably achieved through morphological manipulations of the melon and nasal air sacs. This explains earlier observations of how the dolphin achieved consistent frequency content only in the 0–42 kHz range with simultaneous variability outside this range in echolocation clicks recorded with a single on-axis hydrophone. Similar results were observed for a *Pseudorca crassidens*, while performing similar target discrimination tasks. [Office of Naval Research Grant No 0014-08-1-1160 to P.E. Nachtigall supported this work.]

9:30

2aAB6. Depth and range tracking of sperm whales in the Gulf of Alaska using a two-element vertical array, satellite and bioacoustic tags. Delphine Mathias, Aaron Thode (Scripps Inst. of Oceanogr., Marine Physical Lab., 9500 Gilman Dr., La Jolla, CA 92037-0238 delphine.mathias@gmail.com), Jan Straley (Univ. of Alaska Southeast, Sitka, AK 99835), and Russel Andrews (School of Fisheries and Ocean Sci., Univ. of Alaska Fairbanks, AK 99775)

A two-element vertical array was deployed between August, 15 and 17 2010, on the continental slope off Southeast Alaska, in 1200 m water depth. The instruments were attached to a longline fishing anchorline, deployed at 300 m depth, close to the sound-speed minimum of the deep water profile. The anchorline also served as a decoy, attracting seven depredating sperm whales to the area. Three animals were tagged with a satellite tag and one of them was tagged with both a satellite and bioacoustic “BProbe” tag. Both tags recorded dive depth information. Relative arrival times of surface- and bottom-reflected paths are used to estimate animal range and depth on a single hydrophone, and compared with tagging results. The two-element array is then used to estimate vertical arrival angles of the direct and surface-reflected paths to determine whether range and depth localization can occur without the use of bottom multipath. This data will be useful in determining whether long-range tracking of sperm whales is possible using a single compact instrument deployment. Potential applications include observing what ranges whales are willing to travel to depredate. [Work conducted under the SEASWAP program, supported by the National Oceanic and Atmospheric Administration and the North Pacific Research Board.]

9:45

2aAB7. Matched-field processing and modal filtered range estimates of bowhead whale calls detected in the Alaskan Beaufort Sea. Aaron M Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238, athode@ucsd.edu)

In 2010, a 15-element autonomous vertical array was deployed roughly 35 km north of Kaktovik alongside a distributed array of directional autonomous seafloor acoustic recorders (DASARs). Matched-field processing and geoacoustic inversion techniques were used to extract the range, depth, bottom sound speed profile, density, and attenuation from one close-range whale call. The inversion localized the call at 1.2 km range and 44 m depth in 55 m deep water. The inverted propagation model was used to derive the group and phase velocities of the normal modes in the region surrounding the array. The vertical array spanned sufficient aperture in the water column to permit isolation of the first and second mode arrivals from any given call. A range- and frequency-dependent phase shift was applied to each modal arrival to remove geometric dispersion effects. The modeled range that time-aligned the modal arrivals was selected as the range estimate. The modal filtering technique is demonstrated on additional whale calls produced at 17.3 km and 35 km range from the vertical array, with the range estimates independently confirmed by triangulating bearings of call detections on surrounding DASARs. [Work supported by the North Pacific Research Board, Shell Exploration and Production Company, and Greeneridge Sciences Incorporated.]

10:00

2aAB8. Bearing estimation and 2D localization of bowhead whale calls using directional autonomous seafloor acoustic recorders. Gerald L. D’Spain, Heidi A. Batchelor, Simon E. Freeman (Marine Physical Lab, Scripps Inst. Oceano., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Katherine H. Kim, Charles R. Greene, Jr., Susanna B. Blackwell (Greeneridge Sci., Inc., Santa Barbara, CA), and A. Michael Macrander (Shell Exploration and Production Co., Houston, TX 77079)

Passive acoustic monitoring of the summer/fall westerly bowhead whale migration in the Beaufort Sea has been conducted by Greeneridge Sciences, sponsored by SEPCO, every year since 2006. The directional autonomous seafloor acoustic recorder (DASAR) packages used in this effort each contain three acoustic sensors that simultaneously measure the two horizontal components of acoustic particle motion and acoustic pressure. A variety of data-adaptive beamforming methods have been applied to a selected subset of these data to examine direction-of-arrival estimation performance, including quantifying bearing bias and variance. A principal component analysis (PCA)-type eigenanalysis of the sensor data cross spectral matrix is used to decompose the received field into orthogonal components having different particle motion polarization and energy flux properties. Appropriate manipulation of these components provides high resolution directional estimates of the primary arriving energy while maintaining robustness to uncertainties in sensor calibration and acoustic propagation conditions. Representative results of beamforming as well as 2D localization performance will be presented. Application of data-adaptive beamforming techniques to bulk processing of these large monitoring data sets will be discussed. [Work supported by Shell Exploration and Production Company (SEPCO).]

10:15–10:30 Break

10:30

2aAB9. A chimpanzee responds best to sine-wave speech with formant tones 1 and 2 present. Lisa A. Heimbauer, Michael J. Beran, and Michael J. Owren (Dept. of Psych., The Lang. Res. Ctr., Georgia State Univ., P. O. Box 5010, Atlanta, GA 30302)

A seminal study by Remez and colleagues [R. E. Remez, *et al*., Science, **212**, 947–949 (1981)] demonstrated that listeners were more successful in identifying sine-wave speech when the first two (T12) or all three formant tones (T123) were present than when either were absent (forms T13 and T23). To determine whether a language-trained chimpanzee (Panzee) with the ability to recognize English words in sine-wave form [L. A. Heimbauer, *et al*., Curr. Biol., doi:10.1016/j.cub.2011.06.007 (2011)] would respond similarly, she and 13 humans were tested with words synthesized in the four forms used by Remez *et al*. Indeed, for each species, perception of speech was significantly better when the first and second tones were both present. Panzee’s performance suggests that she is attending to the same spectro-temporal features of sine-wave speech that are critical to humans. The outcomes further indicate that basic capabilities involved in speech perception could have been present in latent form in the common ancestor of humans and chimpanzees. [Work supported by NICHD.]

10:45

2aAB10. Structure of gray whale calls of San Ignacio Lagoon and their distribution in the water column. Anaid Lpez-Urban, Aaron Thode, Carmen Bazúa-Durán, Melania Guerra, and Jorge Urbán-Ramírez (Univ de Baja California Sur, Carretera al Sur Km 5.5, La Paz, Baja California Sur, Mexico)

During the winter gray whales congregate in San Ignacio Lagoon, Baja California, in order to breed and give birth. The Lagoon population can be roughly divided into two demographic groups: mothers with calves, and single animals. Here the acoustic behavior of gray whales wintering in the lagoon is studied by using bioacoustics tags to determine potential relationships between call type and call structure, relative calling frequency, and position in the water column among demographic groups. Between 2008 and 2010, 24 tags (Bio-Probe) recording tags were attached to gray whales in San Ignacio Lagoon. From 1591 minutes of recordings, 1250 calls were identified and classified into five call types: conga (S1), quejid, purr, croac, and ronroneo. The last call has not been previously reported for this species.

Conga calls were the most common type (88% of calls recorded) and was produced by both demographic groups. Ronroneo calls (5%) were mainly produced by single whales. Differences in call parameters (Call duration, minimum, maximum, and low-maximum frequency, number of pulses, and number of harmonics) were only determined for conga calls, for which statistically significant differences were found between demographic groups. Different call types tend to be produced at different depths.

11:00

2aAB11. N3 call types produced long-term by a killer whale of the northern resident community under controlled conditions: Characteristics, variation, and behavioral context. Juliette S. Nash (Dept. of Mar. & Env. Sci., Univ. of San Diego, 5998 Alcala Park, San Diego, CA 92110, julietten-11@sandiego.edu) and Ann E. Bowles (Hubbs - SeaWorld Res. Inst., San Diego, CA 92109)

A-Clan of the Northern Resident killer whale (*Orcinus orca*) community produces a dialect of discrete stereotyped vocalizations, including N3, a call found in all pods of the clan. Ford (1984) reported that N3 is produced almost exclusively in low-arousal states. A 45-yr old female (BC-F1) collected from the A23 matriline of A5 pod and held under controlled conditions has produced this call throughout her adult life. Her calls were collected in 1985 at Marineland of the Pacific (14.25 h of samples) and every few years from 1987 to 2010 at Sea World San Diego (252 h). Analysis of the data show that BC-F1 uses two stereotyped variations of the N3 call. While temporally and structurally similar, there is a positive and negative frequency inflection distinguishing them. BC-F1 delivers N3 with the positive-inflected call preceding the negative. The N3 call occurs in low-arousal states, but also in high-arousal states, including caller interruption, breaking-up of synchronous swimming, and percussive behavior. N3 is infrequently produced during affiliative synchronous swimming and call-matching bouts. Thus, the call is not simply an indication of the resting state but transitions from one state to another. [Research supported by the author's organizations with in-kind support from SeaWorld San Diego.]

11:15

2aAB12. Scalable distributed ultrasonic microphone array. Tórur Andreassen, Annemarie Surlykke (Dept. of Biology, Univ. of S. Denmark, Campusvej 55, 5230 Odense M, Denmark, thor@biology.sdu.dk), and John Hallam (Maersk Mc-Kinney Moeller Inst., Univ. of S. Denmark, 5230 Odense M, Denmark)

A modular approach to recording airborne ultrasound will be presented, the solution is based as much as possible, on retail components and open source software. The modular design, where each module has approx. 4 microphones, allows modules to be combined to extend the coverage area or get higher recording resolution. The system has been designed to have no inherent scalability limits, i.e., only limited by data storage available and the number of microphones you can get a hold of. Current implementation has been used for recording in field experiments with bats, both short-term (minutes) on BCI Panama and for long-term recording (months) in Denmark. The generic nature of the design and implementation allows us to easily replace old components with new technology as it becomes available, both hardware and software. This means we reap the benefits achieved by the electronics industry and also gain any software improvements/bugfixes by a simple Internet upgrade of the module software. Modular design introduces some extra complexity, but this is only an issue when real-time processing the data, or with regards to synchronization. Although said complexity also has advantages, one of them being the possibility of parallel processing.

11:30

2aAB13. Bat recording under controlled conditions: A replicable chamber for comparable results. Eduardo Romero Vivas, Patricia Cortés Calva, and Braulio León López (Centro de Investigaciones Biológicas del Noroeste, S.C., CIBNOR, Mar Bermejo 195 Col. Playa Palo de Santa Rita, La Paz, B.C.S. 23090, Mexico, evivas@cibnor.mx)

In ecology, bats have become a major subject of study. Some pollinate and disperse the seeds of many tropical plants; some help to control insect populations; a few affect livestock by sucking blood, but in all cases bats are indispensable links in ecosystems. Bats use of sound waves make echolocation calls monitoring a powerful tool for distribution, census, and present studies. This technique depends on having a reliable database of sounds for species identification purposes. Although there are several databases of echolocation sounds; the signal emitted by a bat depends on environmental conditions (vegetation, weather conditions), biotic factors (prey size, movement, defensive measures), and the specific task (seek, flee, pursue, evade, wandering, obstacle avoiding). This variability makes comparisons among databases difficult. A portable chamber made of common materials for *in situ* recording under controlled conditions is presented. The use of the chamber is proposed as a tool which might help to generate databases that could be reliable compared. Nine species of desert bats from the northwest region of Mexico were recorder using the chamber under controlled conditions and compared with field recordings. The performance of the chamber and the utility of the database generated for filed identification are presented.

11:45

2aAB14. Development of dolphin-speaker. Yuka Mishima, Keiichi Uchida, Kazuo Amakasu, Yoshinori Miyamoto (Tokyo Univ. of Marine Sci. and Technol., 4-5-7, Konan, Minato-ku, Tokyo 1088477, Japan), and Toyoki Sasakura (Fusion Incorporation, 1-1-1-806, Daiba, Minato-ku, Tokyo 1350091, Japan, sasakura@fusion-jp.biz)

Dolphin speaks broadband sound of ten octave ranges. The vocalization of dolphin sound is categorized into three types, whistle, burst pulse, and echolocation clicks. The frequency range of whistle is below 20 kHz, while that of echolocation clicks and burst pulse is from about several kHz to 150 kHz. Whistle and burst pulse are used social communication. Whistle is used identifying each other and burst pulse may contain emotive factor. Few burst pulse researches are conducted compared to whistle. Until now, burst pulse researches are mainly collecting the dolphin sound using hydrophone but not speaking their sound from human side. The main reason why is no speaking from human side is no speaker of broadband frequency range for burst pulse. The newly developed Dolphin-Speaker has the broadband frequency response from 15 to 150 kHz. The Dolphin-Speaker is developed by advanced technology using multilayer piezoelectric device. We introduce the frequency characteristics of Dolphin-Speaker and the spoken sound by Dolphin-Speaker. We compared the original burst pulse of dolphin and the playback burst pulse by the Dolphin-Speaker. In near future, we would try to playback the burst pulse to dolphin and observe the behavior of the dolphin.

Session 2aEA

Engineering Acoustics and Structural Acoustics and Vibration: Periodic Structures

Andrew J. Hull, Chair
Naval Undersea Warfare Center, Code 8212, Newport, RI 02841

Invited Papers

8:40

2aEA1. Periodic metal structures for acoustic wave control. Andrew Norris (Dept. of Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854)

Passive control of acoustic waves, such as steering energy around an object (acoustic cloaking), requires non-standard materials. Either the density or the bulk modulus or both must be anisotropic. This explains why the phenomenon is not observed in nature. The focus of the talk is the design of metamaterials that have the desired anisotropic stiffness, with isotropic density. This is achieved by periodic metal lattices with sub wavelength cell size. The simplest example is "Metal Water" which is a foam-like periodic structure with the wave speed and density of water, and low shear modulus. The small shear is important in order to get the material to behave acoustically like water but is large enough to provide structural stability. When "deformed" the structure becomes anisotropic with the desired wave steering properties. The simple idea behind this class of acoustical metamaterial has simple consequence in design of cloaking devices, including the concept of conservation of cloaking space. These ideas will be explained and illustrated through scattering simulations.

9:00

2aEA2. Acoustic behavior of magnetorheological fluids in magnetic fields. Thomas R. Howarth, Frank Fratantonio, Jeffrey E. Boisvert, Anthony Bruno (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841-1703), Clyde L. Scandrett (Naval Postgrad. School, Monterey, CA), William M. Wynn, and Philip S. Davis (NAVSEA Div. Panama City, Panama City Beach, FL)

Acoustic metamaterials are being considered for periodic structures where specific microscopic material properties can be tailored to alter macroscopic acoustic fields. One type of acoustic metamaterial being considered is an active fluid known as magnetorheological (MR) fluids. MR fluids contain magnetic particles dispersed within a host fluid where its viscoelastic behavior is controllable by varying the magnetic field intensity. A series of acoustic experiments has recently been conducted at the National High Magnetic Field Laboratory in Tallahassee, Florida. The acoustic sound speed of MR fluids was measured as functions of applied magnetic field strength, normal and orthogonal field orientations, and acoustic frequency. This presentation will discuss MR fluids, measurement methodology, and preliminary results. [Work supported by NAVSEA Division Newport ILIR.]

9:20

2aEA3. Three-dimensional acoustic scattering by layered media: A novel surface formulation with operator expansions implementation. David Nicholls (Dept. of Math, Stat, and CS, Univ. of Illinois-Chicago, 851 S. Morgan St., Chicago, IL 60607, nicholls@math.uic.edu)

The scattering of acoustic waves by irregular structures plays an important role in a wide range of problems of scientific and technological interest. This talk focuses upon the rapid and highly accurate numerical approximation of solutions of Helmholtz equations coupled across irregular periodic interfaces meant to model acoustic waves incident upon a multi-layered medium. A novel surface formulation for the problem is described in terms of boundary integral operators (Dirichlet–Neumann operators), and a boundary perturbation methodology (the method of operator expansions) is proposed for its numerical simulation. The method requires only the discretization of the layer interfaces (so that the number of unknowns is an order of magnitude smaller than volumetric approaches), while it avoids not only the need for specialized quadrature rules but also the dense linear systems characteristic of boundary integral/element methods. The approach is a generalization to multiple layers of Malcolm & Nicholls' Operator Expansions algorithm for dielectric structures with two layers. As with this precursor, this approach is efficient and spectrally accurate.

9:40

2aEA4. Level repulsion states and cavity modes excited by evanescent waves in sonic crystals waveguides. Vicent Romero García (Instituto Universitario para la Gestión Integrada de zonas Costeras, Universidad Politécnica de Valencia, Gandia 46730, Spain), Luis Miguel Garcia-Raffi (Instituto Universitario de Matemática Pura y Aplicada, Universidad Politécnica de Valencia, Spain), Jérôme Vasseur, Anne Christine Hladky-Hennion (Inst. d'Électronique, Microéléctronique and Nanotechnologie, U.M.R. CNRS 8520, France), and Juan Vicente Sanchez-Prez (Centro de Tecnologías Fsicas: Acstica, Materiales y Astrofísica, Universidad Politécnica de Valencia, Spain)

The relevance of the evanescent modes in sonic crystals is theoretically and experimentally reported in this work. The complex bands structure, $k(\omega)$, calculated using the extended plane wave expansion reveals the presence of evanescent modes in these systems, never predicted by the traditional usual numerical $\omega(\vec{k})$ methods. The interpretation of the evanescent modes introduces novel

explanations of the deaf bands as well as of the level repulsion states in antisymmetric periodic systems. In this work we observe that in the ranges of frequencies, where a deaf band is traditionally predicted, an evanescent mode with the excitable symmetry appears changing drastically the transmission properties. On the other hand, the simplicity of the sonic crystals in which only the longitudinal polarization can be excited is used here to interpret, without loss of generality, the level repulsion between symmetric and antisymmetric bands in sonic crystals as the presence of an evanescent mode connecting both repelled bands. These evanescent modes explain both the attenuation produced in this range of frequencies and the transfer of symmetry from one band to the other. The experimental evidence of the level repulsion and the evanescent coupling are in very good agreement with the theoretical predictions.

10:00–10:20 Break

10:20

2aEA5. Elastic response of a cylinder containing longitudinal stiffeners. Andrew J. Hull (Autonomous and Defensive Systems Dept., Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk develops a 3-D analytical model of a cylinder that contains a longitudinal stiffener. The model begins with the equations of motion for a fully elastic solid that produces displacement fields with unknown wave propagation coefficients. These are inserted into stress and displacement equations at the cylinder boundaries and at the location of the stiffener. Orthogonalization of these equations produces an infinite number of indexed algebraic equations that can be truncated and incorporated into a global matrix equation. Solving this equation yields the solution to the wave propagation coefficients and allows the systems' displacements and stresses to be calculated. The model is verified by comparison of the results of a plane strain analysis example to a solution generated using finite element theory. A 3-D example problem is formulated, and the displacement results are illustrated. The inclusion of multiple stiffeners is discussed.

10:40

2aEA6. Computationally efficient finite element method for predicting wave propagation in periodic structures. Vincent Cotoni, Phil Shorter (ESI Group. 12555 High Bluff Dr., San Diego, CA 92130), and Julio Cordioli (Federal Univ. of Santa Catarina, 88040-970 Florianopolis SC, Brazil)

A generic numerical method for predicting the wave propagation in structures with two-dimensional periodicity is presented. The method is based on a combination of Finite Elements and periodic structure theory. A unit cell of the periodic structure is described with finite elements, and a Craig-Bampton reduction is applied to reduce the number of degrees of freedom. Periodic boundary conditions are then applied and the waves propagating in the structure are obtained by solving an algebraic eigenvalue problem. A number of analytical expressions are then used to derive the vibro-acoustic properties of the finite or infinite periodic structure. The method was recently extended to account for heavy fluid loading and material with frequency-dependent properties (typical of acoustic treatments). A number of examples are presented to validate the formulation and demonstrate the possible use of the method for design.

Contributed Papers

11:00

2aEA7. Reflection reduction by three-dimensional and two-dimensional phononic crystal slabs. Sven M. Ivansson (Swedish Defence Res. Agency, 16490 Stockholm, Sweden, sven.ivansson@foi.se)

A thin rubber coating with scatterer inclusions in a periodic lattice can redistribute sound energy, normally incident on a steel plate in water, in the lateral direction. The scattered energy can be absorbed by the rubber material and the reflection amplitude in the water can be reduced significantly. Coatings with different scatterer material types are here compared: air-filled cavities, high-density inclusions, and high-density inclusions coated by soft silicone rubber (which have attracted much interest in recent phononic crystal research). For each material type, scatterers of spherical (in a doubly periodic lattice) or cylindrical (in a lattice with a single period) shape are considered. Each coating type is optimized by differential evolution, varying a number of material and geometrical parameters to minimize the maximum reflectance within a certain frequency band. The layer multiple-scattering method is used as forward model. Good broad-band reflectance reduction is achieved with cavities (monopole scattering), but even better results are obtained with the coated high-density inclusions (dipole scattering). Combined with mixing scatterers of different sizes, the cylindrical shape, with scatterers in a lattice with a single period, is very powerful. The sensitivity of the performance to different parameters, as well as the incidence angle, is illustrated.

11:15

2aEA8. The appearance and use of Bragg scattering effects when sound is perpendicularly incident on a periodic structure. Jingfei Liu and Nico Declercq (Lab. for Ultrasonic Nondestruct. Eval., Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France)

In the study of diffraction spectra for sound incident on periodic structures, it has often been thought that different diffraction orders can be observed and studied separately. This assumption was probably a result of geometrical considerations where a bounded beam, incident and diffracted, is considered as a straight line. It has been widely used to study different phenomena such as surface wave generation and its physical relation to Wood anomalies. In this paper, we develop a geometrical study based on the finite beam width of transducer. The study reveals that measurement of the zero order reflected beam is always accompanied by the back-scattered beams of higher (Bragg) diffraction orders. Extensive measurements have been performed that precisely show the appearance of such waves in the diffraction spectra. Furthermore, the study reveals how the presence of such waves can be used for nondestructive characterization of corrugated surfaces, a technique much easier to interpret than earlier techniques based on Wood anomalies physically caused by surface wave generation. The most important feature is that the introduced technique can be applied for normal incidence and not necessarily in situations where surface waves can be physically generated. [Thanks to the French Centre National de Recherche Scientifique (CNRS)and the Conseil Rgional de Lorraine (CRL) for their financial support.]

11:30

2aEA9. Influence of periodicity on the reflection properties of finite periodic media. Alejandro Cebrecos, Vicent Romero-García, Rubén Picó, Vctor Snchez-Morcillo (Instituto Universitario para la Gestión Integrada de zonas Costeras, Universidad Politécnica de Valencia, Gandia 46730, Spain), Luis Miguel Garcia-Raffi (Instituto Universitario de Matemática Pura y Aplicada, Universidad Politécnica de Valencia, Spain), Juan Vicente Snchez-Prez (Universidad Politécnica de Valencia, Spain), and Kestutis Staliunas (Universitat Politécnica de Catalunya, Barcelona, Spain)

The dispersion relation of periodic systems, the well-known bands structure, gives the information about the propagating modes inside the media. These bands reveal both propagating and non-propagating ranges of frequencies. It is well known that the transmission properties, with relation to the propagating modes, can be characterized by both the bands structure and the equifrequency contours. Oppositely, in finite systems, one should consider the reflection properties on the interface defined by the host and the periodic media in order to know the spatial distribution of the reflected field. In this work, we analyze the reflected acoustic field by a periodic system. An experimental set up is proposed in this work to analyze the reflection properties of a periodic media. From the exploitation of the properties of both the interface and the inner periodicity of the crystal, fundamental and applied questions can be discussed using this media.

TUESDAY MORNING, 1 NOVEMBER 2011 TOWNE/ESQUIRE, 7:55 A.M. TO 12:00 NOON

Session 2aEDa

Education in Acoustics and Physical Acoustics: Tools for Teaching Advanced Acoustics I

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

Scott D. Sommerfeldt, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N181 Eyring Science Center, Provo, UT 84602

Chair's Introduction—7:55

Invited Papers

8:00

2aEDa1. An advanced version of the vibrating string lab. Andrew C. Morrison (Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, achmorrison@gmail.com)

The vibrating string laboratory is a classic of undergraduate physics courses all over. At the introductory level, a common way of setting up a vibrating string lab is with a mechanical shaker and a non-magnetic string. The analysis usually done in the introductory laboratory usually assumes the vibrating string is a linear system. Real vibrating strings are non-linear and are an appropriate choice for exploring non-linear systems in a lab beyond the introductory level. Variations on the introductory lab and pedagogic approaches which make the lab appropriate for advanced undergraduate labs will be presented.

8:20

2aEDa2. Three approaches to understanding sound radiation from a tuning fork. Daniel O. Ludwigsen (Phys. Dept., Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu) and Daniel A. Russell (The Penn State Univ., University Park, PA 16802)

The simple tuning fork has a remarkable pattern of radiated sound. Measuring and modeling this radiation is the ultimate objective of a module designed for our senior-level laboratory course in acoustics. The primary audience for this course is composed of majors in applied physics and engineering physics, as well as engineering students interested in the acoustics minor, mostly electrical and mechanical engineering majors. To complement the blend of students, the course features a careful mix of theory and analytical modeling, computational modeling, and testing. These approaches run through a series of activities emphasizing practical knowledge, including calibrating microphones, validating a finite element model, measuring sound pressure level to produce polar plots of directivity, and using monopoles, dipoles, and quadrupoles to model and predict behavior of several common sources of sound. With these tools, students examine the sound radiated from a tuning fork driven near its lowest resonance frequency. They collect data to create far field and near field directivity plots and use an intensity probe to map 2-D vector intensity in the plane perpendicular to the tines. These are compared with a quadrupole model, as well as the results of a finite element model of their own creation.

8:40

2aEDa3. High speed video of commonly studied oscillating systems. Edward J. Tucholski (Phys. Dept., US Naval Acad., 572C Holloway Rd., Annapolis, MD 21402)

To allow student visualization of vibrating systems, laboratories and demonstrations have been cleverly designed to include stroboscopes, sand, and cork dust. Sensors in the laboratory are often single point type and must be moved to fully visualize the pattern of a vibrating system. Animations fill in the blanks but are still not real. Standard video recording would be the perfect data gathering and visualization tool except the frame rate is too slow for the most commonly studied oscillating systems. Reasonably priced high speed video cameras provide an alternative. In this paper, simple one and two dimensional vibrating systems (e.g., strings, bars, and membranes) are examined with a camera capable of frame rates up to 16 000 frames per second. Video analysis software is used to strip data from the recordings. More complex oscillating systems are also examined to demonstrate the power of this laboratory tool.

9:00

2aEDa4. Optical imaging of bubbles. Tom Matula (APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@apl.washington.edu)

Slow-motion imaging provides a useful way to image various phenomena, from ballistics to biomechanics to fluid dynamics. Such images lead to discoveries and better understanding of concepts. Our lab uses optical imaging to visualize bubbles under various conditions. The results of some of these images have resulted in paradigm shifts. For example, the famous “Crum bubble” showed clearly the formation and propagation of a liquid jet through a bubble. That image has led to numerous publications attempting to model jetting, and the liquid jet concept is now used extensively to explain various phenomena, including pitting of hardened materials and the destruction of kidney stones. In this paper, I will discuss typical imaging setups and provide examples of imaging bubbles under different conditions. Topics will include high-speed imaging, strobe imaging, and shock wave imaging. Phenomena will include sonoluminescence, snapping shrimp, and ultrasound contrast agents. [Work is supported by NIH NIBIB.]

9:20

2aEDa5. Low cost sound level meters for education and outreach. Ralph T. Muehleisen (Decision and Information Sci. Div. Argonne Natl. Lab., Argonne, IL 60439, rmuehleisen@anl.gov) and Andrew C. Morrison (Joliet Junior College, Joliet, IL 60431)

It used to be that when you wanted a low cost sound level meter for teaching and outreach, you could purchase a Radio Shack 33-2050 analog sound level meter. These meters were fairly accurate as well as rugged and could be used as a microphone with preamp in a pinch. Now that the Radio Shack meter is no longer for sale what other options are available? In recent years a plethora of meters and software apps have become readily available at a low cost. In this presentation, the authors will discuss some of the qualities one should look for in a low cost meter to be used for outreach and education and suggest some equipment that meets these needs.

9:40

2aEDa6. Tablet tools for teaching acoustics. Benjamin M. Faber (Faber Acoust., LLC, 654 Stonebrook Ln., Santaquin, UT 84655, ben@faberacoustical.com)

Recent advances in tablet computing technology have in some ways made tablet devices, such as the iPad, an attractive and viable alternative to the traditional notebook computer in and out of the classroom. Tablets are not only smaller and lighter than notebook computers, but typically employ capacitive touchscreen technology, which enables an unprecedented level of interactivity between user and device. The media-centric nature of the current crop of mobile devices also makes them potentially useful as teaching aids, both for classroom demonstration as well as for hands-on experimentation. The tablet’s utility is further enhanced by wireless and/or mobile Internet connectivity, as well as an increasing amount of available third party software. Potential uses of a tablet computer in teaching acoustics will be discussed.

10:00–10:20 Break

10:20

2aEDa7. Application of active-learning techniques to enhance student-based learning objectives. Tracianne B. Neilsen and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602)

Research in physics education has indicated that the traditional lecture-style class is not the most efficient way to teach science courses at the university level. Current best-teaching practices focus on creating an active-learning environment and emphasize the role of the students in the learning process. Several of the recommended techniques have recently been applied to Brigham Young University’s acoustics courses. Adjustments have been built on a foundation of establishing student-based learning outcomes and attempting to align these objectives with assessments and course activities. Improvements have been made to nearly every aspect of the courses including use of class time, assessment materials, and time the students spend out of the classroom. The progress made in bringing two of the courses, specifically an introductory, descriptive acoustics course for a general audience and a junior level introduction to acoustics course for majors, is described. Many of the principles can be similarly applied to acoustics education at other academic levels. Suggestions are made for those seeking to modernize courses at their institutions.

10:40

2aEDa8. Structural vibration studies using finite element analysis. Uwe J. Hansen (Dept. of Chemistry & Phys., Indiana State Univ., Terre Haute, IN 47809)

Bending wave propagation on a two dimensional structure is usually governed by a partial differential equation the solutions of which represent traveling waves. Imposing boundary conditions generally limits the solutions to standing waves representing the normal modes of vibration of the structure. These deflection shapes can be studied experimentally using holographic interferometry or computer aided modal analysis. Finite element analysis (FEA) provides a tool for calculating such normal modes. Input parameters include geometric variables of the structure along with elastic constants, boundary conditions, and a mesh grid which serves as a calculation basis for the iterative computer calculations. Full functioned FEA programs are generally beyond the means of most University educational laboratories. Among more accessible programs with somewhat limited capabilities is ANSYSED. The bulk of the time allotted for this paper will be spent to demonstrate the operation of this program by doing a detailed calculation of normal modes of a square plate clamped on one edge and also illustrating calculated results for a hand-bell.

11:00

2aEDa9. Using COMSOL multiphysics software to investigate advanced acoustic problems. Andrew A. Piacsek (Dept. of Phys., Central Wash. Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu) and Ralph T. Muehleisen (Argonne Natl. Lab, Argonne, IL 60439)

Numerical simulations provide a valuable tool for students to investigate complicated behavior found in many applications of acoustics. Computational "experiments" can be conducted quickly for a large number of parameter values, enabling students to visualize abstract quantities and to grasp causal relationships not easily recognized when looking at equations. The COMSOL finite-element multiphysics software package provides an integrated workspace in which the user defines a problem, meshes the geometry, and plots the solution(s). A brief overview of COMSOL will be presented, along with three examples of how it can be used to model advanced acoustics problems often encountered by students. The example problems involve musical acoustics, fluid-loaded shell vibrations, and flow resistance in porous materials. Also discussed will be the educational benefit of examining how choices made setting up the model can affect the integrity of the solution.

11:20

2aEDa10. Computer modeling in graduate level underwater acoustics. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, misakson@arlut.utexas.edu)

The more a student can become part of the teaching process, the more engaged he is. In a laboratory setting, this involves physically implementing the learned theories. However, for upper level graduate courses, laboratory experiments are often unrealistic. For example, in an underwater acoustics class, it is impossible for a student to physically realize ray bending in the ocean without a large-scale experiment. However, computer modeling can bridge the gap. In this example, students write their own simulation of underwater propagation based on concepts learned in the lecture. Through the computer simulation, students manipulate the environment to determine the effects on acoustic propagation. As the course progresses, interface and target scattering and array and signal processing are added to the propagation code, leaving the student with a comprehensive ocean acoustics modeling toolkit.

11:40

2aEDa11. Measurements of electrical transfer characteristics with soundcards as classroom activity. Stephan Paul (Undergrad. Prog. Acous. Eng., Federal Univ. of Santa Maria, DECC, CT, UFSM, Av. Roraima 1000, Camobi, 97105-900 RS, Santa Maria, stephan.paul@eac.ufsm.br) and Pascal Dietrich (RWTH Aachen Univ., Neustrasse 50, 52066 Aachen, Germany)

Measurements of transfer functions and impulse responses of different types of systems are an important part in acoustics and should be an integral part in acoustics education. Nevertheless, usually expensive equipment is required to carry out such measurements making it difficult to realize hands-on measurement classes. Contrarily the transfer characteristics of electrical systems can be measured much easier. Electrical systems and acoustics meet in electro-acoustic equipment such as soundcards, which can be found as onboard and external devices and in a wide variety. Input and output of soundcards can be connected by different types of transmission systems that might use different types of electrical circuits adding a variety of transfer characteristics. The inherent transfer characteristics of the given soundcard can be analyzed and compensated before assessing the transfer characteristics of different electrical circuits (black boxes). Thus, the experimental measurement of soundcards together with different black-boxes connected to them can be a good means to perform comprehensive measurements of IR and FRF. Using a MATLAB based measurement software, a measurement set-up has been developed to be used by students in a classroom experience. This contribution discusses the set-up, the results of the different scenarios measured and the feedback of the students.

TUESDAY MORNING, 1 NOVEMBER 2011 SAN DIEGO, 10:30 A.M. TO 1:00 P.M.

Session 2aEDb

Education in Acoustics: Halloween Hands-On Acoustics Demonstrations for Middle-School Students

Joseph R. Gladden, Cochair
Dept. of Physics, National Center for Physical Acoustics, Univ. of Mississippi, 108 Lewis Hall, University, MS 38677

Murray S. Korman
Dept. of Physics, U.S. Naval Academy, Annapolis, MD 21402

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomenon. In this session we will be organizing a set of "Hands-On" demonstrations for a group of middle school students from the San Diego area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Demos will be run in parallel, so students can break into small groups to encourage questions and allow students to spend more time on demos that interest them. The session will have a "Halloween" theme with decorations and costumes! Any acousticians wanting to participate in this fun event should email Josh Gladden (jgladden@olemiss.edu) or Murray Korman (korman@usna.edu).

Session 2aMU

Musical Acoustics: Physical Models For Sound Synthesis I

Edgar J. Berdahl, Chair
Dept. of Music, Stanford Univ., Stanford, CA 94305

Chair's Introduction—8:30

Invited Papers

8:35

2aMU1. Synthesis of new bowed-string sounds using physical models. Philippe Manoury (UCSD Dept. of Music, CPMC 354, 9500 Gilman Dr., MC 0099, La Jolla, CA 92093-0099)

With a virtual bowed-string model, natural aspects of the sound and sound production, such as bow pressure, position, and velocity, control the realization of new synthetic sounds. It is the best way to resolve many problems in synthetic music involving phrasing, transitions, irregularity, and regularity. But what happens when you ask the model to simulate bow movements which are quite impossible for a human being? For example, to produce vertically an exaggerated pressure at the same time as an exaggerated slow motion of the bow across a string? That is even more interesting than to reproduce a simply natural acoustical sound.

9:05

2aMU2. Modeling of a violin input admittance by direct positioning of second-order resonators. Esteban Maestre (CCRMA—Dept. of Music, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, esteban@ccrma.stanford.edu), Gary P. Scavone (CIRMMT—Schulich School of Music, McGill Univ., Montreal, QC H3A 1E3, Canada), and Julius O. Smith (CCRMA—Dept. of Music, Stanford Univ., Stanford, CA 94305)

When approaching violin sound synthesis, the theoretical advantage of body modeling by finite difference or finite element paradigms comes from parameterizing the model by geometry and material properties. However, difficulties in representing the complexity of physical phenomena taking place have kept such approaches from raising more success, due especially to limited modeling accuracy and high computational cost. Conversely, the design of digital filters from admittance measurements, although generally offering a less meaningful parameterization, represents a more affordable technique as it provides significant fidelity at lower computational cost. Within digital filter formulation applied to the violin, modal representations can be considered as among the most physically pertinent, since vibration modes defining the timbre signature are in general observable from admittance measurements. This work introduces a technique for designing violin passive admittances by direct, non-uniform positioning of second-order resonators. Starting from admittance measurements, second-order resonator parameters are designed so that desired modes are modeled from frequency-domain data. Positve real models providing significant accuracy at low order are obtained from second-order resonator parameter fits. As an example, a two-dimensional input admittance is designed from measurements, so that it can be used for a digital waveguide model to include a control for bowing angle.

9:35

2aMU3. An overview of the afterglow phenomena compensation. Carlos Spa (Dept. of Math., Univ. Técnica Federico Santa María Av., Vicuña Mackenna 3939, San Joaquin Santiago de Chile, Chile), Jose Escolano (Telecommunication Eng. Dept., Univ. of Jaén), Toni Mateos (Barcelona Media Innovation Ctr.), and Adan Garriga (Int. Ctr. of Numerical Methods in Eng.)

In many fields such musical and room acoustics, wave propagation is usually simulated by using discrete-time numerical methods. Since 3-D simulations often show an excessive computational cost, proposals exist for extrapolating results from 3-D simulations conducted in equivalent 2-D scenarios. Among the most significant limitations of such extrapolation techniques, recently it has been pointed out the propagation of a point-like impulse in 2-D exhibits the so-called afterglow effect, which leads to obtaining non-null field values after the arrival of the wavefront. In order to compensate for this effect, a filtering process is proposed to overcome this limitation, not only in free space but also on closed spaces. Therefore, three dimensions-like impulse responses can be efficiently computed from 2-D simulations in the context of complex geometries. Recent results using pseudo-spectral time-domain, finite-difference time-domain, and digital waveguide mesh methods demonstrate the potential of this method to extrapolate impulse response from 2-D simulations with similar properties to those coming from 3-D.

10:05–10:20 Break

10:20

2aMU4. Synthesis of harpsichord pluck tones using a physical plectrum and string models. Chao-Yu J. Perng (Dept. of Phys., Stanford Univ., 382 Via Pueblo Mall, Stanford, CA 94305, perng@stanford.edu), Julius O. Smith, and Thomas D. Rossing (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ.)

The harpsichord is a plucked string keyboard instrument with a distinct sound, and previous work has been done to synthesize it [V. Valimaki *et al*., EURASIP J. Appl. Signal. **7**, 934–948 (2004)]. However, these excitation signals are extracted through recorded tones and are not physical models in the true sense. A physical model of the harpsichord plectrum and its interface with the digital waveguide string model has been proposed by us [Chao-Yu J. Perng *et al*., J. Acoust. Soc. Am. **128**, 2309(A) (2010)], and a revised model accounting for the plectrum tip was presented subsequently [Chao-Yu J. Perng *et al*., J. Acoust. Soc. Am. **129**, 2543(A), (2011)]. In this paper, we will demonstrate results of the synthesized tones using our physical plectrum-string interaction model. Changing certain physical parameters alters the synthesized tones, and they are discussed and explored. Lastly, a simple physical model of the harpsichord lute-stop is presented.

10:50

2aMU5. Modalys, a physical modeling synthesizer: More than twenty years of researches, developments, and musical uses. Rene Emile Causse, Joel Bensoam, and Nicholas Ellis (IRCAM, UMR CNRS 9912, Musical acoustics, 1 Place IGOR Stravinsky, 75004 Paris, France)

In the early 1990s, the software was initially created as an open environment with the purpose to serve as a virtual instrument maker workshop. Indeed, the modal formalism offers several interesting advantages, among them the uniform description of numerous mechanical or acoustical systems allowing easy hybridizations. Wildest virtual instruments may well be imagined, played, and manipulated. As there are very few acoustics volumes or mechanical structures tractable by an exact analytical solution, numerical methods, such as FEM, have been programmed for complex geometry for both structures and fluids. The usages are now extending from the virtual reproduction of existing instruments to industrial prototyping. This diversification as the necessity to further improve performance made it necessary to rethink many parts of the software, from the core synthesizer to the numerous interfaces: textual, MAX/MSP, OPENMUSIC, and MATLAB. To control physical modeling synthesis, it is necessary to specify the physical, low-level details of how to play an instrument. Consequently, several recent studies have focused on the study of instrumental gesture and its modeling, thereby increasing the realism of the synthesis. This presentation will review the current possibilities of the software which will be illustrated by examples. Work in progress will also be presented.

11:20

2aMU6. Sound synthesis and musical composition with the physical modeling formalism CORDIS-ANIMA. Claude Cadoz (ACROE & ICA Lab., Grenoble Inst. of Technol., 46 Ave. Flix Viallet, Grenoble, F38000, Claude.Cadoz@imag.fr), Nicolas Castagne (ICA Lab., Grenoble Inst. of Technol., 46 Av. Flix Viallet, Grenoble F38000), and Olivier Tache (ACROE, Grenoble Inst. of Technol., 46 Av. Flix Viallet, Grenoble F38000)

The two historical starting points of Computer Music, in 1957, were digital sound synthesis, founded by Max Mathews (Bell Labs), and automatic composition, founded by Lejaren Hiller and Leonard Isaacson (University of Illinois). Digital sound synthesis and computer aided musical composition then developed and are today essential components of computer music. As said by Jean-Claude Risset, sound synthesis allowed to compose the sound itself. At the end of the 1970s, a new paradigm appeared, no more based on the synthesis of the sound signal, but on the simulation of the physical objects that produce the sound. The CORDIS-ANIMA language, created by Claude Cadoz and his colleagues Annie Luciani and Jean-Loup Florens at the ACROE lab, in Grenoble (France), is one of the most important representatives of this approach. We will present this language, which allows simulating the various components of a musical instrument, but also complex orchestras and dynamic objects with macro temporal behaviors. We will then demonstrate how this language enables linking sound creation and musical composition within a single and global paradigm using the GENESIS environment, created at ACROE for musical applications of CORDIS-ANIMA, and we will play extracts of musical pieces composed with GENESIS.

Session 2aNS

Noise: Noise Impacts in Quiet Residential Communities

Richard D. Horonjeff, Chair
Consultant in Acoustics and Noise Control, 81 Liberty Square Rd., #20-B, Boxborough, MA 01719

Chair's Introduction—8:10

Invited Papers

8:15

2aNS1. Examination of valued attributes of low sound level environments. Richard D. Horonjeff (81 Liberty Square Rd. #20-B, Boxborough, MA 01719) and Herb J. Singleton, Jr. (P.O. Box 90842, Springfield, MA 01139)

The introduction of new noise sources into rural environments presents a number of unique community response challenges not typically encountered in urban and suburban environments. Low ambient sound levels combined with intermittently (rather than continuously) audible anthropogenic ambient contributors create unique soundscapes. Residents appear to value many attributes of these soundscapes and wish to preserve and protect them. Application of existing standards and guidelines (quantitative or nuisance-based) often does not meet resident expectations of protection. This paper examines a number of the valued attributes. It then asks questions about noise policy, resident expectations, adequacy of existing standards, source and ambient noise quantification, and the masking of valued attributes by a new noise source. It concludes with a list of issues that bear close examination in order to set future noise impact guidelines for low-ambient environments.

8:35

2aNS2. Challenges in selecting rural project locations under different types of noise regulations. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

It is sometimes desirable to locate certain types of projects in very rural areas, not only because of noise issues, but also because some may need to be close to electrical grid transmission lines, natural gas pipelines, transportation corridors, open space or special topographical features. Examples of such projects include mining operations, military training facilities, power plants, firing ranges, experimental proving grounds, and rescue dog training facilities, among many others. Whenever planners of such specialty facilities identify an appropriate location, invariably there are noise standards that must also be met before construction and operations can be permitted by the respective local authorities. While some noise requirements are quite specific, others are purposefully vague. Some noise regulations are just copied from other municipalities, while others are carefully crafted to provide for the desired community ambiance. However, a major problem can occur when typical community noise standards are applied to rural locations. This paper describes very different types of projects in quiet rural settings and their difficulties complying with the various types of typical municipal noise ordinances. Some suggestions are given to avoid the obvious noise regulation difficulties.

8:55

2aNS3. Case study: A quiet rural community in a commercial/industrial zone. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

A small New Hampshire town allowed a community to be built in a commercial zone, immediately adjacent to an industrial zone that is home to a long-dormant quarry. When the quarry owner applied for a permit to resume quarrying operations, the nearby residents objected on the grounds of noise impacts and other environmental concerns. A failure to consider noise as an element of community planning put the town in a difficult position. Should residents living in an commercial zone be afforded the same privilege of living in a quiet environment than those in residential zones? Should the industry be held to more stringent standards because of this non-conforming use? Both sides of this issue will be discussed.

9:15

2aNS4. Can a statewide noise rule cover both urban and rural areas? David Braslau (David Braslau Assoc., Inc., 1313 5th St. SE Ste. 322, Minneapolis, MN 55414)

Minnesota adopted a statewide noise rule in 1974. Statistical descriptors L10 and L50 for daytime and nighttime periods were established, along with a state law that no local government can adopt more stringent standards. While these are land use receiving standards, the rule is applied to noise generated by individual sources. The rule has worked well in urban environments but control of existing and new sources in rural environments has been problematical, both for generators and receptors of noise. The intent of the rule is to provide criteria that potential new development can take into account when proposing and designing new facilities, but in rural areas with very low ambient levels, the rule allows very large increases in sound level against which receptors have no recourse. Local governments have taken other approaches to control new sound sources by proposing acceptable increases over ambient noise level at receiving land uses or simply denying permits to new noise sources. Sources such as wind turbines are particularly difficult to address, since these are commonly placed in quiet rural areas. The paper discusses some problems and attempted solutions to allow new noise sources in rural areas without compromising quiet rural environments.

9:35

2aNS5. The sound of quietness. How to design sites that are perceived as quiet? Andre Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de) and Brigitte Schulte-Fortkamp (TU Berlin, Berlin 10587, Germany)

The EC Noise Directive 2002/49, related to the assessment and the management of environmental noise, demands "to preserve environmental noise quality where it is good." In addition, one major issue of the directive concerns the creation of quiet zones in urban areas. This ambitious goal of creating "acoustically green" and quiet areas in cities cannot be achieved without an interdisciplinary approach, which holistically reflects the human perception of acoustical sceneries. Sound sources constitute a soundscape and semantic attributions to sources become intrinsic properties of the respective acoustic signals. This means that humans cannot distinguish between the physical and the interpreted events. Psychological and social aspects have a sustainable impact on evaluations and must be considered to create quiet areas, which are perceived as quiet and recreative. Thus to understand residents' and visitors' reactions to environmental noise, it is imperative to investigate their expectations and experiences besides considering noise metrics and acoustical indicators. The paper will present outcomes of recent surveys, where quiet areas in Aachen, Berlin, and Leipzig and their impacts on residents and visitors were scrutinized. The analyses have been conducted with special focus on Soundscape design.

9:55

2aNS6. Individual expectations and noise policy in quiet residential settings. Nancy S. Timmerman P. E. (Consultant in Acoust. and Noise Control, 25 Upton St., Boston, MA 02118-1609, nancy.timmerman@alum.mit.edu)

Consultants in the noise field tend to encounter the more sensitive individuals. These people may feel entitled to not hear anyone else. This paper will characterize the quiet acoustical setting, and consider the expectations of the more normal individuals. It quiet settings, it is as easy to be heard as to hear. The author, whose family owned a cottage in the Manistee National Forest in Michigan, will recount some personal acoustical observations from being there in the summer. This paper will also examine noise policies for these settings and look at "dB above ambient" as a criterion, since it is in use in Massachusetts, where this consultant practices. Illustrations will also be taken from clients of the author in western Massachusetts and in Maryland.

10:15–10:30 Break

Contributed Papers

10:30

2aNS7. Proving a facility complies with noise regulations. Mark V. Giglio (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, mvg@cavtocci.com)

This paper examines the difficulties associated with proving compliance with environmental noise regulations. Topics include determining pre-existing background sound, post-construction measurements during facility operations, and balancing corporate rights with neighborhood expectations. When a new facility is built in a quiet neighborhood, noise complaints are commonplace. The new facility is required to prove compliance with the local noise regulation. Compliance measurements must be made when the facility cannot be shut down and ambient sound levels are unknown. A statistical analysis of the pre-existing background sound often suggests the probability of proving compliance is minimal. In addition, bias toward/against one party in the dispute can have significant ramifications.

10:45

2aNS8. A three-dimensional noise mapping study for heterogeneous traffic conditions. Ramachandraiah Alur and Kalaiselvi Ramasamy (Dept. of Civil Eng., Indian Inst. of Technol., Chennai, India)

Characteristics of noise in terms of frequency, noise level, vary considerably with respect to heterogeneous traffic. Urban traffic noise characteristics in the cities of a like India are slightly varied in the sense that the composition of the traffic is heterogeneous. A heterogeneous traffic stream composes of vehicles having different speeds, sizes, and operating characteristics. While studying the feasibility of noise reduction in some of the areas around a heterogeneous traffic stream, a three-dimensional (3-D) noise mapping study has been attempted with the help of Arc GIS, Arc Scene along with field measurements. The methodology involves to build 3-D noise models to analyze the 3-D occurrence of noise pollution. Specifically in this work a 3-D acoustical model has been developed for a local area. Subsequently, the observation points around the buildings have been determined and the noise levels have been calculated using the authors' regression model. The noise mapping parameters such as Ld, LN, and Lden have been calculated incorporating the geometrical features of the roads and varying heights of the buildings, heterogeneity, and prominent honking conditions. This work leads to the prediction of noise levels in front of building facades both in horizontal and vertical directions.

11:00

2aNS9. Energy-based method for near real-time modeling of sound field in complex urban environments. Stephanie M. Pasareanu, Marcel C. Remillieux, and Ricardo A. Burdisso (Mech. Eng. Dept., Virginia Tech, 142 Durham Hall, Blacksburg, VA 24061, psteph7@vt.edu)

Prediction of the sound field in large urban environments has been limited thus far by the computational heaviness of standard numerical methods such as boundary element (BE) or finite-difference time-domain methods. Recently, a considerable amount of work has been devoted to developing energy-based methods for this application, and results have shown the potential to compete with conventional methods. However, these developments have been limited to 2-D models, and no real description of the phenomena at issue has been exposed. Here, the mathematical theory of diffusion is used to predict the sound field in complete 3-D complex urban environments. A 3-D diffusion equation is implemented by means of a simple finite-difference scheme and applied to two different types of urban configurations. This modeling approach is validated against BE and geometrical acoustic solutions, showing a good overall agreement. The role played by diffraction near the source is discussed, and suggestions are made on the possibility to accurately predict the sound field in complex urban environments in near real time simulations.

11:15

2aNS10. Experimental and numerical study on the propagation of impulsive sound around buildings and induced structural loading. Marcel C. Remillieux, Joseph M. Corcoran, T. Ryan Haac, Ricardo A. Burdisso (Vib. and Acoust. Labs., Mech. Eng. Dept., Virginia Tech, Blacksburg, VA 24061, mremilli@vt.edu), and U. Peter Svensson (Acoust. Res. Ctr., Dept. of Electron. and Telecommunications, Norwegian Univ. of Sci. and Technol., NO-7491 Trondheim, Norway)

Propagation of impulsive sound around buildings and induced structural loading are investigated experimentally and numerically. Experiments were conducted on a rectangular building at Virginia Tech using sonic booms generated by an explosive technique. Assuming linear-acoustic propagation and acoustically rigid surfaces, these experiments were simulated with a three-dimensional numerical model, in the context of geometrical acoustics, by combining the image source method for the reflected field (specular reflections) with the Biot-Tolstoy-Medwin method for the diffracted field. This numerical model is validated against a boundary element solution and against experimental data, showing a good overall agreement. Some of the key advantages of this modeling approach for this application are pointed out such as the ability to model three-dimensional domains over a wide frequency range and also to decompose the sound field into direct, reflected, and diffracted components, thus allowing a better understanding of the sound-propagation mechanisms. Finally, this validated numerical model is used to investigate sound propagation around a cluster of six rectangular buildings, for a range of elevated source positions.

11:30

2aNS11. Sound transmission modeling for residential buildings using finite elements. Beom Soo Kim (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg. University Park, PA 16802, buk104@psu.edu) and Victor W. Sparrow (The Penn State University, 201 Appl. Sci. Bldg., University Park, PA 16802)

Noise transmission into residential building structures has been studied in order to understand the outdoor-to-indoor transmission characteristics. This work is motivated by the need to model the transmission of subsonic aircraft noise into homes. As a first step, finite element (FE) models of a simplified residential house developed using both MSC MD Nastran and FFT Actran were compared to check the validity of the modeling techniques as well as to analyze their applicability for low frequencies. Fluid structure interaction was considered to include the presence of the enclosed acoustic fluid. Model relevance was also checked with impulsive load measured data supplied by NASA in 2007. The FE model of a detailed individual room was developed to show interior pressure responses. [Work supported by FAA.]

11:45

2aNS12. Investigation of the sound distribution in street canyons with non-parallel building Façades. Kaj Erik Piippo and Shiu-keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong)

Sound propagation in street canyons have been a topic of interest for decades. In a congested city such as Hong Kong, street canyons with large height to width ratios are a common sight. Many prediction models have been developed over the years, but the problem is still of interest because of the complexity due to the multiple reflections and varying surface conditions in real life cases. In this paper, a 1:4 scale model experiment of a street canyon, 4 m long and 2 m high, is presented where one of the facades is inclined. The reference case is with two parallel facades, which is compared with cases when one facade is tilted 80°, 70°, and 60°. The measurements were conducted in an anechoic chamber and a line source array generating white noise was used for simulating traffic noise. The sound distribution patterns were studied on the reference wall and the inclined wall. It was observed that the sound strength and the reverberation decrease rapidly when inclining the opposite wall. For low and mid frequencies, the sound reduction was more significant closer to the top of the inclined facade, while the opposite could be seen for the vertical facade.

TUESDAY MORNING, 1 NOVEMBER 2011 — ROYAL PALM 3/4, 7:30 A.M. TO 12:00 NOON

Session 2aPA

Physical Acoustics: Physical Acoustics of Extreme Environments, Condensed Matter, and More

Veerle M. Keppens, Cochair
Materials Science and Engineering, Univ. of Tennessee, Dougherty Hall, Knoxville, TN 37996

Cristian Pantea, Cochair
Sensors and Electrochemical, Los Alamos National Lab., Los Alamos, NM 87545

Chair's Introduction—7:30

Invited Papers

7:35

2aPA1. Resonant ultrasound techniques for measurements from 500 mK to 700 K. Jonathan B. Betts, Albert Migliori (NHMFL-PFF Los Alamos Natl. Lab., Los Alamos, NM 87507), Arkady Shehter (NHMFL, 1800 E. Paul Dirac Dr., Tallahassee, FL 32310-3706), Victor R. Fanelli, and Fedor F. Balakirev (NHMFL-PFF Los Alamos Natl. Lab. Los Alamos, NM 87507)

Measuring elastic moduli using resonant ultrasound spectroscopy (RUS) over a very wide temperature range requires very specialized hardware. The authors have developed both this hardware and some interesting software to help make measurements in these very challenging conditions. Solutions to these problems will be presented, along with results from some recent RUS experiments carried out at the National High Magnetic Field Laboratory in Los Alamos. [Work was supported by the National Science Foundation, State of Florida, and the U.S. Dept. of Energy.]

8:00

2aPA2. Resonant ultrasound spectroscopy at high temperatures and pressures: Palladium hydride near the tri-critical point. Joseph R. Gladden III and Rasheed Adebisi (Dept. of Phys. & NCPA, 108 Lewis Hall, Univ. of Mississippi, University, MS 38677, jgladden@olemiss.edu)

Much fundamental quantum and critical phenomenon physics has been learned from precision elastic constant and attenuation measurements at low temperatures. Less attention has been paid to phenomenon in the high temperature regime outside of specialized areas such as geophysics. However, interesting fundamental and applied physics does not stop at room temperature. In this talk, I will review methods developed for applying resonant ultrasound spectroscopy at high temperature (<1300 K) and high hydrostatic pressure (<140 atm). One of the phenomenon we have studied using this apparatus is palladium hydride near the tri-critical point in the temperature, pressure phase diagram. The results from these measurements have validated a recent theory by Schwarz and Khachaturyan (2006) regarding the hysteresis in metal hydride systems which predicts a strong softening in the elastic moduli.

8:25

2aPA3. Phonon-driven electron localization in delta-plutonium. Albert Migliori (Natl. High Magnetic Field Lab., Los Alamos Natl. Lab., Los Alamos, NM 87545)

Three remarkable properties of delta-Pu, accurately measured, lead to exceptionally strong constraints on *ab-initio* electronic structure models and suggest a new route must be taken to understand the localization of electronic states. The three properties that lead to this are the thermal expansion coefficient, the temperature dependence of the elastic moduli, and the absolute values of the elastic moduli. Both neutron diffraction and dilatometer measurements of the thermal expansion reveal a broad temperature range where thermal expansion is zero, and where small changes in Ga concentration can make thermal expansion vary from small and positive to small and negative. Measurements of the bulk modulus of these alloys over the same temperature range reveal extreme softening on warming. How can the bulk modulus, which is the curvature of the energy with respect to volume, change when volume does not? This is the central property that is not encompassed by present electronic structure models. We discuss theory and RUS measurements to understand better what must be true.

8:50

2aPA4. Acoustic velocity measurements on solids and liquids under extreme conditions of pressure and temperature. Baosheng Li and Wei Liu (Mineral Phys. Inst., Stony Brook Univ., Stony Brook, NY 11794)

Elastic properties of materials under extreme conditions of pressure and temperature are of great interests to researchers in many disciplines. In the last decade, acoustic velocity measurements using an improved ultrasonic interferometry method have been developed in both MA-6 and MA-8 types of multi-anvil high pressure apparatus. By placing the piezoelectric transducer (lithium niobate, 10 deg Y-cut) in a stress-free location and using extended delayline, high S/N acoustic signal can be maintained, while sample is under high pressure and temperature. By combining with synchrotron X-radiation, measurements of sound velocities using ultrasonic interferometry, crystal structure and unit cell parameters using X-ray diffraction, and sample strain (length) using X-radiographic imaging, can be made simultaneously, all *in-situ* at a high pressure and temperature, enabling a pressure-standard-free characterization of solid and liquid materials to 25 GPa and 1800 K. Results on ceramic and metallic materials from recent experiments will be presented to show velocities as a function of pressure and temperature, absolute pressure determination, equation of state for glass, and the application to liquids. Other new developments, such as controlling sample stress state at high P and T, the study of composites, and materials undergoing phase transformations will also be reviewed [Work sponsored by NSF and DOE/NNSA.]

9:15

2aPA5. Acoustics of metals under extreme conditions by laser-ultrasonics in diamond anvils cell. Frédéric Decremps (UPMC, 4 place Jussieu Paris 75005, France), Laurent Belliard, Bernard Perrin, and Michel Gauthier (UPMC, 4 place Jussieu, Paris 75005, France)

Major progress on ultrafast acoustics instrumentation and diamond anvils design during the last 2 yr now allows detailed elastic and visco-elastic studies under extreme conditions and on a wide variety of systems. I will here mainly review the state of the art of the recent development of a method combining the time-resolved picosecond optical technique and a diamond anvil cell to measure sound velocity Decremps *et al*., [Phys. Rev. Lett. **100**, 3550 (2008)]; Decremps *et al*., [Rev. Sci. Instrum. **80**, 73902 (2009)]. Contrary to other groups which currently scope with these problems mostly using large facilities, we propose here entirely new and novel technique to measure the sound velocity of solid and liquid under high pressure and high temperatures. I will illustrate these possibilities by a number of recent studies on crystalline, polycrystalline, and liquid metals. Prospects will be discussed.

Contributed Papers

9:40

2aPA6. Acoustic microscopy investigation of superconducting materials. T. Tahraoui, S. Debboub, Y. Boumaïza, and A. Boudour (Faculty of Sci., Dept. of Phys., Lab. LEAM, Badji Mokhtar Univ. ANNABA B.P.12, 23000, ANNABA, Algeria)

The present work has as subject the study of the ultrasonic attenuation in some superconductor materials. This study is based on the simulation of the acoustic signal obtained by the reflection acoustic microscope upon the exploration of a coated or uncoated material. The examination of the simulated signal has permitted the determination of the variation of the reflection coefficient with respect to the incidence angle of the exciting acoustic wave. We have also determined the elastic constants, the velocities of different modes of propagation, the acoustic attenuation of the Rayleigh mode as a function of temperature from the reflection coefficient, and the acoustic signature.

9:55

2aPA7. Weakly nonlinear simple waves in Hertzian chains. B. Edward McDonald and David Calvo (Naval Res. Lab., Washington, DC 20375, ed.mcdonald@nrl.navy.mil)

The discrete system of equations for a granular chain consisting of a large number of spheres interacting via the hertz force is cast as an effective

medium. In the long wavelength limit, the second order equation of motion for the effective medium possess a subset of simple waves obeying a first order equation of reduced nonlinear index. Simple waves are those in which knowledge of one dependent variable determines all the rest. For a given initial strain, the simple wave solution prescribes initial mass velocity. Strain and velocity profiles from the first order equation are used as initial conditions in simulations for the second order discrete system. Results for viscous and inviscid shock formation compare very well between the second order system and the reduced first order equation. Second order simulation of colliding waves reveals the ability of waves to pass through each other, with a phase advance accruing during the collision process. Results may be related to explosions in granular media. [Work supported by the Office of Naval Research.]

10:10–10:30 Break

10:30

2aPA8. Cylindrical bubble dynamics. Yurii A. Ilinskii, Todd A. Hay, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Bubbles generated between closely spaced parallel surfaces are approximately cylindrical. Here, cylindrical bubble dynamics are compared with those of spherical bubbles. In the linear approximation, the product of bubble radius and natural frequency of the monopole mode in water is $\sim$1m/s for a cylindrical bubble versus $\sim$3m/s for a spherical bubble. Radiation damping of a cylindrical bubble is an order of magnitude greater than for a spherical bubble, and for cylindrical bubbles larger than $\sim$1 μm it is the dominant loss mechanism, whereas spherical bubbles must be larger than $\sim$ 1 mm before radiation losses dominate. Also in contrast with spherical bubbles, the inertial load on a cylindrical bubble in an unbounded incompressible liquid is infinite and prohibits the existence of a monopole mode. Derivation of a dynamical equation for cylindrical bubbles in Rayleigh-Plesset form therefore requires the incompressible liquid to be of finite extent. The radial extent needed to impose the appropriate inertial load is found by comparison with results for a compressible liquid to be approximately 200 bubble radii. The resulting dynamical equation was further augmented to account for bubble deformation due to coupling of surface modes through nonlinearity. [Work supported by NIH DK070618 and EB011603.]

10:45

2aPA9. Simulations of self-organization of bubbles in acoustic fields in three dimensions. Nail A. Gumerov (UMIACS, Univ. of Maryland, 115 A.V.Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu), Iskander S. Akhatov (North Dakota State Univ., Fargo, ND 58108), Yekaterina V. Volkova, and Uliana O. Agisheva (Bashkir State Univ., Ufa, 450074, Russia)

Self-organization of bubbles (structure formation) and related self-action of the acoustic field is a strongly non-linear phenomenon, which is observed, e.g., in acoustic cavitation and sonochemical reactors. The phenomenon is a manifestation of a two-way field-particle interaction, when the bubbles change their position and sizes due to the Bjerknes and other forces and rectified diffusion, and affect the acoustic properties of the medium, which leads to restructuring of the acoustic field. The models of this phenomenon available in literature are revised, and a model based on the spatio-temporal averaging, which includes the above mentioned effects, is derived. Based on this model, a 3-D pseudospectral code for simulation of bubble self-organization is developed and tested. Simulations show interesting spatio-temporal behavior of the bubbles and the acoustic field, which is initialized as a standing wave but may drastically change its structure due to bubble dynamics (especially near the resonance). Bubble pattern formation is sensitive to many parameters including parameters of the acoustic field, bubble initial size, number density, spatial distribution, and ambient conditions. The effects found in the result of simulations are discussed. [This research is supported by the Grant of the Ministry of Education and Science of the Russian Federation (G34.31.0040).]

11:00

2aPA10. Sonic boom structural response simulation of an historic structure. Michael M. James, J. Micah Downing (Blue Ridge Res. and Consulting, LLC, 15 W. Walnut St., Ste. C, Asheville, NC 28801, Michael.James@BlueRidgeRes.com), and Joel M. Garrelick (Appl. Physical Sci., Inc., Lexington, MA 02421)

The potential for structural damage from sonic booms can be assessed by using stationary sound sources. Such sources simulate sonic booms as incident, linear, and plane acoustic waves. In theory, the source simulates the pressure field on a structure arising from this incident wave, including influences from the surrounding acoustic environment. In practice, compromises must be made in the placement of the acoustical source and its orientation to the structure and the resulting change in the expected incident angle of a sonic boom. In this effort, we evaluated multiple structural elements of Fort Jefferson in the Dry Tortugas, Fl which ranged from structurally sound to fragile. The structural concern was damage to the more fragile elements from sonic booms generated in the surrounding airspace. The methodologies and results will be discussed. [Work sponsored by Naval Facilities Command, Southeast.]

11:15

2aPA11. Acoustic oscillations in the early universe. David Knobles (KSA, P.O. Box 27200, Austin, TX 78731)

Acoustic oscillations in the baryonic-photon plasma leave an imprint on both the cosmic microwave background radiation (CMBR) and the power spectrum of the mass distribution in the universe. This effect is well known and is currently applied to determine the value for quantities such as the percentage of dark matter and dark energy in the universe. This study examines the contributions of higher order terms in the equations for the radiation-baryonic acoustic oscillations prior to radiation decoupling from matter. The energy and momentum conservation equations are combined with the Einstein field equations for a Robertson Walker metric containing the perturbations. An attempt is made to consider the effect of the higher order perturbations on the observed power spectrum and on the anisotropy of the CMBR. The sensitivity of these higher order terms to adiabatic, instead of non-adiabatic, boundary conditions on the fluctuations outside the particle horizon is considered. [Work supported by KSA.]

11:30

2aPA12. Acoustic levitator for contactless transport and mixing of droplets in air. Nada Bjelobrk, Majid Nabavi, and Dimos Poulikakos (Dept. of Mech. and Process Eng., Inst. of Energy Technol., Lab. of Thermodynamics in Emerging Technologies, ETH Zurich, CH-8092, Zurich, Switzerland, nadab@ethz.ch)

Acoustic levitation is a simple and cost-effective method for investigation of small amounts of any kind of material. Standard levitators focus the standing acoustic wave only along its traveling direction. Therefore, levitated objects (entrapped in the potential minima) are separated by potential maxima which cannot be overcome. Thorough design of the emitter plate and consequently its vibration pattern allows a more complex acoustic field between the emitter and reflector, establishing multiple potential minima perpendicular to the wave propagation direction in the H1-mode. The number of minima is regulated by changing the emitter-reflector distance H. Thus, multiple solid or liquid samples can be simultaneously levitated and rapidly transported over several centimeters by controlling H. A piezoelectrically driven droplet generator was used to eject pL droplets into the acoustic field and coalesce them to a 1–2 μL sized droplet in the preferred potential minimum. Two suspended droplets were transported and mixed when two potential minima were merged into one by increasing H. During pattern transition, high pressure fluctuation might increase droplet Bond number beyond its critical value and cause droplet atomization. This was prevented by continuous adjustment of acoustic power.

11:45

2aPA13. Effects of heat conduction in wall on thermoacoustic wave propagation in a gas-filled, channel subject to temperature gradient. Nobumasa Sugimoto and Hiroaki Hyodo (Dept. of Mech. Sci. Graduate School of Eng. Sci., Univ. of Osaka, Toyonaka, Osaka 560-8531, Japan, sugimoto@me.es.osaka-u.ac.jp)

This paper examines effects of heat conduction on acoustic wave propagation in a gas-filled, channel subject to temperature gradient axially and extending infinitely. Within the narrow-tube approximation in the sense that a typical axial length is much longer than a channel width, the system of linearized equations for the gas supplemented by the equation for heat conduction in the solid wall is reduced to a thermoacoustic-wave equation for excess pressure uniform over the cross-section. This is the one-dimensional equation taking account of the wall friction and the heat flux at the wall surfaces given in the form of hereditary integrals. This equation is derived rigorously under the narrow-tube approximation, and is valid for any form of disturbances. The effects of heat conduction appear in the form proportional to the square root of the product of the ratio of the heat capacity per volume of the gas to that of the solid, and the ratio of the thermal conductivity of the gas to that of the solid. Although the product is very small usually, which endorses validity in neglect of the heat conduction, it is revealed that there are situations in which its effects are enhanced, depending on geometry and materials.

TUESDAY MORNING, 1 NOVEMBER 2011 ROYAL PALM 5/6, 8:00 TO 11:35 A.M.

Session 2aSA

Structural Acoustics and Vibration and Architectural Acoustics: Extraction of Information from Correlations of Random Vibrations

Earl G. Williams, Cochair
Naval Research Lab., Acoustics Div., 4555 Overlook Ave., SW, Washington, D.C. 20375

Karim G. Sabra, Cochair
Mechanical Engineering, Georgia Inst. of Technology, 771 Ferst Dr., NW, Atlanta, GA 30332

Invited Papers

8:00

2aSA1. Extracting the earth response from noise and complex earthquake data. Roel Snieder, Nori Nakata, Kees Wapenaar, and Evert Slob (Ctr. for Wave Phenomena, Colorado School of Mines, CO 80401 rsnieder@mines.edu)

Theory shows that when random noise is generated in proportion to the local dissipation rate, one can extract the response of a system from correlations of field measurements of such noise. Such a condition of equilibrium is rarely satisfied for elastic waves of the earth. Yet the earth response can be extracted from measured noise. We show examples of the retrieval of P-waves and of S-waves. The latter type of waves are retrieved from traffic noise in an urban environment. Since the traffic noise is excited by localized sources (road and railroads), it displays strong amplitude variations. In order to compensate for such amplitude variations, we use cross-coherence rather than cross-correlations for the data processing. We also analyze complicated waveforms excited by earthquakes to create maps of the shear-wave velocity in Japan and show that the shear wave velocity changes with time; this velocity drops throughout northeastern Japan with about 5% after the recent Tohoku-Oki earthquake. Our measurements show that the shallow subsurface in Japan weakens after the earthquake over a region about 1 200 km wide.

8:25

2aSA2. Green's function retrieval from noise by multidimensional deconvolution. Kees Wapenaar, Joost van der Neut, Evert Slob (Dept. of Geotechnol., Delft Univ. of Technol., Stevinweg 1, 2628CN Delft, The Netherlands, c.p.a.wapenaar@tudelft.nl), and Roel Snieder (Colorado School of Mines, Golden, CO 80401-1887)

The correlation of noise at two receivers is approximately proportional to the Green's function between these receivers. The approximation is accurate when the medium is lossless and the noise field is equipartitioned. These assumptions are in practice often violated: the medium of interest is often illuminated from one side only, the sources may be irregularly distributed and losses may be significant. For those situations, the correlation function is proportional to a Green's function with a blurred source. The source blurring is quantified by a so-called point-spread function which, like the correlation function, can be derived from the observed data (i.e., without the need to know the actual sources and the medium). The blurred source can be focused by multidimensionally deconvolving the correlation function for the point-spread function. We illustrate the correlation and deconvolution methods with several examples and discuss the advantages and limitations of both methods.

8:50

2aSA3. The potential for extracting the electromagnetic earth response from uncorrelated noise. Evert Slob, Kees Wapenaar (Dept. of Geotechnology, Delft Univ. of Technol., Stevinweg 1, 2628 CN, Delft, Netherlands, e.c.slob@tudelft.nl), and Roel Snieder (Ctr. for Wave Phenomena, Colorado School of Mines, Golden, CO 80401-1887)

Thermal electromagnetic radiation from an absorbing medium allows for extracting the response of this medium. In the earth, thermal noise is usually very weak and other forms of electromagnetic noise prevail. Noise below radio frequencies is generated in the atmosphere, ionosphere, and magnetosphere, while cosmic noise generates electromagnetic waves above radio frequencies that reach the earth's surface. The noise requirements are discussed for energy and Lagrangian forms of earth response extraction by correlation and multi-dimensional deconvolution. It is shown that if the uncorrelated noise sources are located outside the earth, only the earth response to incident electromagnetic waves can be extracted. Potential applications are found for ground-penetrating radar, and these are illustrated with numerical examples. For coupled seismic waves and electromagnetic fields in fluid-filled porous media, uncorrelated seismic noise is shown to be sufficient to approximately extract the electro-seismic earth response.

9:15

2aSA4. Imaging and monitoring with ambient vibrations: A review. Larose Eric (ISTERRE, CNRS, & UJF, BP 53, 38041 Grenoble cedex 9, France)

The principle of passive imaging and reconstructing the Green functions by means of correlating ambient vibrations or noise will be reviewed. Some basic processing procedures for optimizing the convergence of the correlations, along with the role of multiple scattering, will also be presented. Monitoring with ambient noise constitutes a different goal that relies on different assumptions on the background noise structure. Similarities and differences between the imaging and the monitoring approaches will be addressed.

9:40

2aSA5. Correlation processing of ocean noise. W.A. Kuperman (Marine Physical Lab. of the Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238)

Correlation processing of ocean ambient noise has been a topic of growing interest. While theory confirms the efficacy of this procedure, experimental confirmation has been limited to a few data sets. The basic issue of the potential utility of this procedure is the time needed to build up the relevant cross-correlation peaks that are diagnostic of the ocean environment. This time is a function of frequency/bandwidth, the ocean environment and noise structure/distribution, sensor separation and, if employed, the array configuration. After reviewing some basic background, a selection of experimental results for noise originating from ships, surface generated sources, and geophysical sources is presented.

10:05–10:15 Break

10:15

2aSA6. Using cross-correlations of ambient vibrations for passive structural health monitoring of a high-speed naval ship. Karim G. Sabra (School of Mech. Eng. Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332-0405)

Previous studies have used the cross-correlation of ambient vibrations (CAVs) technique to estimate the impulse response (or Green's function) between passive sensors for passive imaging purposes in various engineering applications. The technique (CAV) relies on extracting deterministic coherent time signatures from the noise cross-correlation function computed between passive sensors, without the use of controlled active sources. Provided that the ambient structure-borne noise field remains stable, these resulting coherent waveforms obtained from CAV can then be used for structural monitoring even if they differ from the actual impulse response between the passive sensors. This article presents experimental CAV results using low-frequency random vibration data ($>$50 Hz) collected on an all-aluminum naval vessel (the HSV-2 Swift) operating at high speed (up to 40 kn) during high sea states. The primary excitation sources were strong wave impact loadings and rotating machinery vibrations. The consistency of the CAV results is established by extracting similar coherent arrivals from ambient vibrations between the pairs of strain gages, symmetrically located across the ship's centerline. The influence of the ship's operating conditions on the stability of the peak coherent arrival time, during the 7 days trial, is also discussed. [Sponsored by ONR, N00014-09-1-0440.]

10:40

2aSA7. Extracting information from an array of sensors in a diffuse noise field using random matrix theory. Ravi Menon, Peter Gerstoft, and William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093)

Isotropic noise fields are often used to model several practical diffuse noise fields. For an array of equidistant sensors in such a noise field, the cross-spectral density matrix (CSDM) of the array is a Toeplitz sinc matrix. Here, the eigenvalues of the CSDM for ideal isotropic noise fields are first derived for infinite arrays. The eigenvalues have close connections with classical array processing concepts such as the invisible region in frequency-wavenumber space (region where there is no propagating energy, but a spectrum can be calculated). Random matrix theory deals with eigenvalue distributions of random matrices and its concepts are applied here by modeling the array snapshot vectors as zero-mean, unit variance Gaussian random variables, with a sinc covariance matrix. Using the Stieltjes transform, the eigenvalues of the ideal CSDM are related to those of the sample CSDM, and an analytical solution for the distribution of the eigenvalues of the sample CSDM is obtained. At frequencies where the array is spatially undersampled, increasing the number of observations results in the noise masquerading as a signal, which could lead to erroneous signal detections. We demonstrate how knowing and understanding the eigenvalue distribution helps improve the extraction of information from ocean ambient noise.

Contributed Papers

11:05

2aSA8. Analyzing structure-borne sound by using the forced vibro-acoustic components. Logesh Kumar Natarajan and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, logesh@wayne.edu)

At the 158th meeting of the ASA, a numerical study on analyzing structure-borne sound radiation using the forced vibro-acoustic components (F-VACs) established by the Helmholtz equation least squares (HELS) method was presented [Natarajan *et al*., JASA, **126**, 2244 (2009)]. The present paper presents an experimental case study of this approach on a baffled square plate subjected to a point force excitation using random signals. The radiated acoustic pressures are measured by a planar array of microphones at a very close distance to the plate surface. These input data are used to determine the F-VACs of the plate obtained by decomposing the transfer functions into orthogonal space via singular value decomposition. These F-VACs are correlated to sound radiation through sound power calculations. Meanwhile, the F-VACs are expanded in terms of the natural modes of the plate, so the dominant F-VACs that are directly responsible for sound radiation can be correlated to the natural modes of the plate. Using this F-VAC analysis, one can identify the critical natural modes of a structure, which by themselves have no direct relationships to sound radiation. Based on the F-VAC analysis results, the plate is modified and tested to examine its effectiveness in reducing structure-borne radiation.

11:20

2aSA9. An improved method of structural-borne random vibration mass attenuation. Arya Majed, Ed Henkel (Appl. Structural Dynam., Inc., P.O. Box 272428, Houston, TX 77277), and Ali Kolaini (NASA/Jet Propulsion Lab., California Inst. of Technol., Pasadena, CA 91125)

In 2009, the NASA Engineering and Safety Center (NESC) Vibroacoustics Working Group identified vibroacoustic environment predictions as one of the highest risk areas for new launch vehicle programs. Specifically, the working group identified the need for improved random vibration mass attenuation prediction methods as key to mitigating this risk. This paper derives a multi drive point, multi axis, interface sized equation between the unloaded and loaded drive point random vibration accelerations and cross correlations. The method's development and associated acoustic test validation program were a collaborative effort between ASD, NESC, and NASA/JPL. The derivation utilizes the methods of modal synthesis and random vibration averaging and precludes the use of any simplifying structural interaction assumptions besides the interfaces behaving linearly (no interface deadbands). Key to the method's improved accuracy is its built-in mechanism for combining the contributions of the different unloaded drive-point accelerations and associated cross-correlations. The method reduces to the time-tested method of Norton–Thevenin for the single drive-point case. Predictions are compared to measurements from NASA/JPL acoustic chamber test of a panel loaded by a component. It is shown that the method accurately captures the major spectral characteristics of attenuations and amplifications resulting in improved mass loaded environment predictions.

TUESDAY MORNING, 1 NOVEMBER 2011 — PACIFIC SALON 4/5, 7:55 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication and Psychological and Physiological Acoustics: Fifty Years of Invariant/Variant Speech Features: Where Do We Stand?

Jont B. Allen, Cochair
Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801

Peggy B. Nelson, Cochair
Dept. of Speech, Languages, Hearing Sciences, Univ. of Minneosta, 164 Pillsbury Dr., S.E., Minneapolis, MN 55455

Sandra Gordon-Salant, Cochair
Dept. of Hearing and Speech Science, Univ. of Maryland, College Park, MD 20742

Chair's Introduction—7:55

Invited Papers

8:00

2aSC1. What causes the confusion patterns in speech perception? Feipeng Li (Dept. of Biomedical Eng., Johns Hopkins Univ., 108 W 39 St., 33, Baltimore, MD 21210) and Jont Allen (Beckman Inst., Univ. of Illinois at Champaign-Urbana, Urbana, IL 61801)

It is widely believed that speech contains redundant cues which make it robust in noise. We have discovered that natural speech often contains conflicting cues that cause significant confusion patterns in speech perception. To demonstrate the impact of conflicting cues on the identification of consonant sounds, we selected four plosives /ta, ka, da, ga/ and four fricatives /si, Si, zi, Zi/ that show significant confusion with other sounds from a female high-performance talker in the LDC database and derived a set of test stimuli by removing the dominant cue and manipulating the conflicting cue. It was found that the consonants always morph to their competing sounds, once the dominant cue is unavailable. The confusion can be strengthened by enhancing the conflicting cue. These results demonstrate that the identification of consonant is dependent on a careful balancing between the dominant cue and the conflicting cues. Such prior knowledge about perceptual cues can be used to manipulate the perception of consonants, as demonstrated by the selected examples of nonsense syllables, meaningful words, and sentence.

8:20

2aSC2. Speak, memory—Wherefore art thou, invariance? Steven Greenberg (Silicon Speech, 4683 Hawaina Way, Kelseyville, CA 95451, steveng@silicon-speech.com)

Spoken language is highly variable, reflecting factors of environmental (e.g., acoustic-background noise, reverberation), linguistic (e.g., speaking-style), and idiosyncratic (e.g., voice-quality) origin. Despite such variability, listeners rarely experience difficulty understanding speech. What brain mechanisms underlie this perceptual resilience, and where does the invariance reside (if anywhere) that enables the signal to be reliably decoded and understood? A theoretical framework—DejaNets—is described for how the brain may go from "sound to meaning." Key is speech representations in memory, crucial for the parsing, analysis, and interpretation of sensory signals. The acoustic waveform is viewed as inherently ambiguous, its interpretation dependent on combining data streams, some sensory (e.g., visual-speech cues), others internal, derived from memory and knowledge schema. This interpretative process is mediated by a hierarchical network of neural oscillators spanning a broad range of time constants (ca. 15–2 000 ms), consistent with the time course and temporal structure of spoken language. They reflect data-fetching, parsing, and pattern-matching involved in decoding and interpreting the speech signal. DejaNets accounts for many (otherwise) paradoxical and mysterious properties of spoken language including categorical perception, the McGurk effect, phonemic restoration, semantic context, and robustness/sensitivity to variation in pronunciation, speaking rate and the ambient acoustic environment. [Work supported by AFOSR.]

8:40

2aSC3. Invariant acoustic cues of consonants in a vowel context. Riya Singh and Jont B. Allen (Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801)

The classic JASA papers by French and Steinberg (1947), Fletcher and Galt (1950), Miller and Nicely (1955), and Furui (1986) provided us with detailed CV+VC confusions due to masking noise and bandwidth and temporal truncations. FS47 and FG50 led to the succinctly summarizing articulation index (AI), while MN55 first introduced information-theory. Allen and his students have repeated these classic experiments and analyzed the error patterns for large numbers of individual utterances [http://hear.beckman.illinois.edu/wiki/Main/Publications], and showed that the averaging of scores removes critical details. Without such averaging, consonant scores are binary, suggesting invariant features used by the auditory system to decode consonants in isolated CV. Masking a binary feature causes the consonant error to jump from zero to chance (within some small subgroup of sounds), with an entropy determined by conflicting cues, typically present in naturally spoken sounds. These same invariant features are also used when decoding sentences having varying degrees of context. A precise knowledge of acoustic features has allowed us to reverse engineer Fletcher's error-product rule (FG50), providing deep insight into the workings of the AI. Applications of this knowledge is being applied to a better understanding of the huge individual differences in hearing impaired ears and machine recognition of consonants.

9:00

2aSC4. The perception of phonetic features and acoustic cues by impaired listeners. Matthew B. Winn, Monita Chatterjee (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD 20742, mwinn1@.umd.edu), and William J. Idsardi (Dept. of Linguist., Univ of Maryland, College Park, MD 20742)

The search for invariant acoustic cues in the perception of speech features has been an under-utilized blessing to those interested in individuals with hearing impairment. While multiple co-varying acoustic cues complicate the search for invariance, they provide plentiful opportunity for impaired listeners to perceive contrasts that would otherwise be lost when salient information is compromised by hearing loss or noise. Classic confusion matrices and information transfer analyses account for phonetic features perceived, but do not always reveal the acoustic cues that drive these perceptions; it is rarely acknowledged that difficult listening conditions that are likely to result in a re-prioritization of acoustic cues. In this presentation, we discuss some completed and ongoing work suggesting that impaired listeners (or simulated impaired listeners) don't merely show different amounts of success but exhibit different listening strategies (i.e., cue-weighting strategies) when perceiving phonetic features. This is similar to findings related to developing language learners and adult learners of a second language. Implications are discussed here for cochlear implant users, listeners with high-frequency hearing loss, and listeners in background noise in tasks that measure the use of cues in both the auditory and visual domains.

9:20

2aSC5. Individual variance of hearing-impaired consonant perception. Woojae Han (Dept. of Speech and Hearing Sci. Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, woojaehan@gmail.com) and Jont Allen (Univ. of Illinois at Urbana-Champaign)

Individuals with sensorineural hearing loss (SNHL) are prescribed hearing aids (HAs), based on the results of clinical measurements. Although the HAs do help these individuals communicate in quiet surroundings, many listeners have complained that their HAs do not provide enough benefit to facilitate understanding of normal speech. We argue that the current clinical measurements, which interpret the result as a mean score (e.g., pure-tone average, speech recognition threshold, AI-gram, etc.), do not deliver sufficient information about the characteristics of a SNHL listener's impairment when hearing speech, resulting in a poorly fitting HA. We confirm how reliably this consonant-vowel (CV) test could measure a SNHL listener's consonant loss using only zero-error utterances (in normal hearing listeners) and having a statistically suitable number of presentations in CVs, in order to characterize unique SNHL consonant loss. As noise increased, the percentage of error and confusions of target consonants increased. Although some consonants showed significantly higher errors and resulted in more confusion than others, SNHL ears have a very different consonant perception/error, which may not be either measured or analyzed by the use of average scores. Comparison between the two (separated) phases of the study supports a good internal consistency for all SNHL ears.

9:40

2aSC6. Dynamic spectral structure really does support vowel recognition. Joanna H. Lowenstein and Susan Nittrouer (Dept. of Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., 4th Fl, Columbus, OH 43212, lowenstein.6@osu.edu)

Traditional approaches to the study of speech perception manipulate cues that are spectrally and temporally discrete, assuming these cues define phonetic segments. Alternatively, dynamic structure across longer signal stretches may support recognition of linguistic units. Strange and colleagues [Strange *et al.* J. Acoust. Soc. Am. **74**, 695–705 (1983)] seemed to support this view by showing that adults can identify vowels in CVCs even when large portions of the syllable centers are missing, but dynamic structure at the margins is preserved. However, adults may just be "filling in the missing parts," having learned how structure at the margins covaries with structure in the middle. To test these alternatives, adults and 7-yr-olds labeled sine-wave, and vocoded versions of silent-center stimuli. The former preserves dynamic structure; the latter obliterates it. If listeners simply fill in missing parts, adults should perform equally well with sine-wave and vocoded stimuli, and children should perform equally poorly with both. Instead, all listeners performed better with sine-wave stimuli. These outcomes provide further support for the perspective that speech perception is facilitated by broader and longer acoustic structure than that represented by notions of the acoustic cue. [Work supported by NIDCD Grant DC-00633.]

9:55

2aSC7. Deriving invariance by integrating cues computed relative to expectations. Allard Jongman (Dept. of Linguist., Univ. of Kansas, Lawrence, KS 66045, jongman@ku.edu) and Bob McMurray (Univ. of Iowa, Iowa City, IA 52240)

Invariance and variability are notions that have stimulated much research on the acoustics and perception of speech. While some invariance must exist to explain how listeners generally derive the same message from spoken input, the question remains at which level invariance occurs. The theory of acoustic invariance explicitly states that the invariance is found in the acoustic signal. While some phonemic distinctions do seem to be represented by invariant acoustic properties, it is clear that many acoustic cues are context-dependent. Normalization or compensation processes for coping with specific sources of variability have been proposed as solutions to this context-dependency. We examined acoustic properties for a large corpus of American English fricatives. We argue that the lack of invariance is not a major obstacle and demonstrate that it can be overcome by the use of multiple acoustic cues and a relatively simple compensation scheme by which cues are recoded relative to expectations derived from context (vowel quality and speaker). A comparison of this approach (C-CuRE, Computing Cues Relative to Expectations) [McMurray and Jongman, Psych. Rev. **118**, 219–246 (2011)] to acoustic invariance and exemplar approaches suggests that only C-CuRE achieved a similar accuracy to listeners and showed the same effects of context.

10:10–10:25 Break

10:25

2aSC8. Invariants for phones or phonemes?: Only literacy leads us to expect them. Robert Port (Dept.s of Ling'cs and Cognit. Sci, Indiana Univ, Bloomington, IN 47405, port@indiana.edu)

This special session asks about "invariants," but *invariants in what domain under what transformations?* If it is acoustic invariants for phones or phonemes, i.e., letter-sized units for spelling words, which implies the same acoustic cues for all contexts where a phone or phoneme is used in transcription. This clear impossibility is why Liberman turned to "motor invariants." Yet acoustic invariants for phones are what is required for perceptual categorization. Phones (and phonemes) must be rejected as a linguistic memory code due to, at least, (a) evidence of indexical information in speech memory, (b) lack of evidence of discrete jumps for feature changes, and (c) evidence of real-time patterns in language (see Port and Leary, 2005, Lang; Port, 2011, Eco Psych). Our strong intuitions of abstract and letter-like linguistic memory result from our lifelong training with an alphabet, not from data. Memory for speech must incorporate rich detail and continuous (not serial) time and also supports some degree of abstraction for generalization to new contexts. Linguistic units like distinctive features, phones, and phonemes are merely categories, not symbol tokens. Categories are cultural conventions regarding events "considered the same" by some community (like *tree* and *game*) and do not necessarily have any invariant properties.

10:40

2aSC9. In defense of discrete features. Khalil Iskarous (Haskins Labs., 300 George St., New Haven, CT)

A major challenge for discrete feature theories of speech perception is that the acoustic signal is continuously changing within and between segments at a high rate. These theories must explain how continuously changing spectral patterns are perceived as discrete and independent features. This problem does not disappear after auditory transformations of the acoustic signal, since temporal and frequency resolution, despite being coarse, still preserve most of the signal dynamics. In this work, it is shown that if articulatory features are extracted from the continuously changing acoustic signal, assuming that the signal is the output of a lossless vocal tract terminated in a unit resistance, i.e., a Darlington configuration (Iskarous, Journal of Phonetics, 2010), it is possible to obtain a discretely changing place of articulation feature from the continuous signal change. This will be illustrated using articulatory inversion of VV and CV transitions from the X-ray microbeam database for five male and five female participants. This work supports theories of discrete and independent perceptual features, as assumed for instance by Miller and Nicely (1955), if the acoustic signal is interpreted in terms of the vocal tract actions that cause it (Goldstein and Fowler, 2003).

10:55

2aSC10. On the non-acoustic nature of acoustic invariants. D. H. Whalen (Haskins Labs, 300 George St., Ste. 900, New Haven, CT 06511, and City U New York, whalen@haskins.yale.edu)

Various proposals have attempted to put speech targets into invariant "acoustic" form, sometimes with an additional transformation into "auditory" space. These targets, however, are not strictly acoustic. Targets for vowels, for example, are transformed so that vocal tract length differences (between talkers, especially across men, women, and children) are taken into account. Such a transformation makes the resultant targets combinations of articulatory and acoustic information. The auditory transformation improves automatic speech recognition, but the theoretical underpinnings for this result have been unclear. Ghosh, Goldstein, and Narayanan [J. Acoust. Soc. Am. **129**, 4014–4022] shows that the articulatory information is maximized by the auditory transform, indicating that this transform is not solely in the acoustic domain. Moreover, a lowered F3 has been proposed as the production target for American English /r/ [e.g., Nieto-Castanon *et al*., J. Acoust. Soc. Am. 117, 3196–3212], but synthesis that retains an exemplary /r/ F3 while altering F1 and F2 results in other percepts, such as /w/ or a pharyngeal glide. F3 as an acoustic target is insufficient by itself and must incorporate articulatory dynamics implied by the other formants. Thus, "acoustic" invariants, to the extent they work at all, do so by incorporating articulation.

11:10

2aSC11. Perceptual recovery of phonetic features in blanked segments of disyllabic words. Pierre L. Divenyi (Speech and Hearing Res., VA Northern California Health Care System, Bldg. R4, 150 Muir Rd., Martinez, CA 94553, pdivenyi@ebire.org) and Adam C. Lammert (Signal Anal. and interpretation Lab., Dept. of Comput. Sci. Univ. of Southern California, Los Angeles, CA 90089)

Spondees, both true disyllabic words and concatenated monosyllabic word pairs, had their middle section replaced by silence that extended from the midpoint of the first to the midpoint of the second vowel; the silence

thus included the final consonant(s) of the first and the initial consonant(s) of the second syllable. These stimuli were presented to normal-hearing young listeners instructed to guess both monosyllabic half words. Input and response spondees were orthographically aligned and analyzed as confusion matrices. Articulatory gestures were estimated for each token via the Haskins Laboratories TADA articulatory synthesis. After time alignment, articulatory distances were calculated for each stimulus-response pair. In the blanked middle of the spondee, phoneme-based confusions of place-of-articulation were high (39% accuracy), while gestural distances underlying this feature (location and degree of tongue tip constriction, lip aperture, and velar constriction) were under 10%. These results suggest that acoustic traces of a significant portion of gesture trajectories underlying consonantal place-of-articulation that start before and/or terminate after the silence, are perceived by the listener. An articulatory phonology might thus be more robust to degradation. [Work supported by NSF, AFOSR, and the VA Medical Research.]

11:25–12:00 Panel Discussion

TUESDAY MORNING, 1 NOVEMBER 2011 ROYAL PALM 1/2, 8:00 A.M. TO 12:00 NOON

Session 2aSP

Signal Processing in Acoustics, Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Overcoming Environment/Medium Effects in Acoustic Tracking of Features or Parameters

Ananya Sen Gupta, Cochair
Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543-1053

R. Lee Culver, Cochair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Invited Paper

8:00

2aSP1. Modeling and tracking temporal fluctuations for underwater acoustic communications. James Preisig (Dept. of Appl. Ocean Phys. and Eng., WHOI, Woods Hole, MA 02543, jpreisig@whoi.edu)

High-rate, phase-coherent communications through the underwater acoustic channel requires implicitly or explicitly tracking the fluctuations of the channel impulse response. Current techniques estimate the time-varying channel impulse response using observations of signals that have propagated through the channel over some averaging interval and often rely on some method of representing the time variability in the channel. For example, the work by Stojanovic that enabled phase-coherent communications through the underwater channel represented the time-variation as a common phase rotation (Doppler shift) of all taps of the channel impulse response. The tracking of channel fluctuations involves a fundamental trade-off between the number of independent parameters used to represent both the channel impulse response coefficients (i.e., the dimensionality of the estimation problem) and their temporal variations, the rate of channel fluctuations that can be tracked, and the signal to noise ratio (SNR). This talk will present new and review established techniques for modeling time variability and reducing dimensionality in underwater acoustic channel. The performance of these approaches as a function of SNR and rate of channel fluctuation will be compared and their computational complexity will be discussed.

Contributed Papers

8:20

2aSP2. Effectiveness of sparse optimization techniques against rapid fluctuations in the shallow water acoustic channel. Ananya Sen Gupta and James Preisig (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543)

The shallow water acoustic channel exhibits rapid temporal fluctuations due to fluid and platform motion as well as reflection from moving surfaces. Tracking the time-varying channel delay spread effectively for subsequent equalization is an open signal processing challenge. The delay-Doppler spread function characterizes this time-variability over a selected range of Doppler frequencies but itself does not exhibit stationary behavior over longer time intervals. Typically the delay-Doppler spread function and sometimes the delay spread itself follow a sparse distribution where most of the energy is concentrated in a few significant components distributed sparsely over a larger support. A variety of sparse optimization techniques have recently been proposed in the compressive sensing literature to efficiently track sparsely distributed coefficients. However, the ill-conditioned nature of the delay-Doppler spread estimation problem, coupled with the necessity to track the complex-valued coefficients directly in real time, renders direct application of traditional sparse sensing techniques infeasible and intractable in the shallow water acoustic paradigm. The talk will provide a synopsis of well-known and recently proposed sparse optimization techniques, with focus on mixed norm algorithms, along with a comparative analysis of these techniques in the shallow water acoustic paradigm over simulated and experimental field data.

8:35

2aSP3. A wave-making towing tank for underwater communication in a multi-path environment using time reversal method. Gee-Pinn James Too (Natl. Cheng Kung Univ., Dept. of Systems and Naval Mechatronic Eng., No. 1 University Rd. Tainan 70101 China, z8008070@email.ncku.edu.tw)

A 4 m $\times$ 8 m $\times$ 176 m wave-making towing tank, which is original, used for resistance measurement for boats is a perfect test platform for underwater communication in a multi-path environment. The objective in the present study is to establish a process of underwater communication in a multi-path environment using the time reversal method. The source projector sends out a BPSK signal, while the signal is received and processed in order to restore the original signal at the source by time reversal method. The communication process is conducted via both numerical simulation and experiment for cases of underwater communication in a 4 m $\times$ 8 m $\times$ 176 m towing tank. It is found that TRM process reduces the error rate of the transmission significantly and improves the communication quality in multi-path environment. [Works are supported by National Science Council of Taiwan.]

Invited Papers

8:50

2aSP4. Geophysical parameter estimation. Max Deffenbaugh (ExxonMobil Res. and Eng. Co., 1545 Rte. 22 East, Annandale, NJ 08801, max.deffenbaugh@exxonmobil.com)

Seismic exploration for oil and gas is a parameter estimation problem. Geological properties and fluid content of subsurface reservoirs are sensed from the earth's surface. Seismic data are acquired by generating elastic waves at the surface and recording the reflections off subsurface targets using large receiver arrays. At present, it is not practical to determine earth parameters by iteratively refining an earth model until simulated seismic data match recorded field data. The computational demands of accurately simulating viscoelastic wave propagation in a 3-D heterogeneous earth are prohibitive. Instead, seismic data are processed to remove many complexities of wave propagation in the real environment, like multipath, attenuation, dispersion, shear waves, and interface waves. With these complexities removed, the processed data conform to simpler propagation models which can be solved in reasonable computer time. The cost of each simplification, however, is the loss of some information about the subsurface parameters of interest. This tradeoff between the tractability of the inversion and the accuracy of the result can be quantified by comparing computational time versus the Cramér-Rao bound on subsurface parameter estimates. Examples are discussed for some commonly used seismic signal processing algorithms.

9:10

2aSP5. Modeling dominant mode rejection beamformer notch depth using random matrix theory. John R. Buck (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, johnbuck@ieee.org) and Kathleen E. Wage (George Mason Univ., Fairfax, VA 22030)

Any practical tracking algorithm must mitigate the effect of both nonstationary environments and loud interfering sources. The dominant mode rejection (DMR) adaptive beamformer (ABF) is often used to address these challenges. The DMR beamformer periodically updates the sample covariance matrix (SCM) employed to compute the ABF weights. A tension arises between the large number of snapshots required for each SCM update to null loud interferers effectively and the small number of snapshots required for each SCM update to track nonstationarities. In practice, observed notch depths for DMR fall dramatically short of those predicted theoretically from ensemble statistics. This talk bridges the gap between theoretical and observed performance by presenting a simple linear asymptotic model for the DMR notch depth derived from random matrix theory results on the accuracy of the SCM eigenvectors. The model predicts the mean DMR notch depth as a function of the number of snapshots given the interferer-to-noise ratio (INR), the array size, and the interferer location relative to the look direction. Close agreement is demonstrated between the model and simulations over a wide range of INRs and array sizes. [Work supported by ONR 321US.]

9:30

2aSP6. Dominant mode rejection beamformer notch depth: Theory versus experiment. Kathleen E. Wage (ECE Dept., George Mason Univ., 4400 Univ. Dr. MSN 1G5, Fairfax, VA, kwage@gmu.edu), John R. Buck (Univ. of Massachusetts Dartmouth, N. Dartmouth, MA 02747), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

The dominant mode rejection (DMR) adaptive beamformer attenuates loud interferers by directing beampattern nulls toward signals contained in the dominant subspace [Abraham/Owsley, Proc. Oceans, 1990]. The dominant subspace is defined by the eigenvectors associated with the largest eigenvalues of the sample covariance matrix. DMR performance is primarily determined by how closely the eigenvectors of the sample covariance matrix match the true interferer directions. Random matrix theory (RMT) describes how the accuracy of the sample eigenvectors varies with the interferer-to-noise ratio (INR), array size, and the number of snapshots used to estimate the sample covariance. A simplified analytical model based on RMT predicts the mean DMR notch depth as a function of INR, array size, interferer location and the number of snapshots. This talk compares the RMT predictions with experimental results obtained by cancelling array strum in data from a vertical array deployed in the Philippine Sea. The array strum interference predominantly falls within the subspace spanned by the first two eigenvectors of the covariance matrix. Notch depth statistics obtained using a large set of receptions show good agreement between theory and experiment. [Work supported by ONR.]

9:50

2aSP7. A moment-based technique to palliate pervasive clutter while preserving objects of interest. Roger C. Gauss and Joseph M. Fialkowski (Naval Res. Lab., Code 7144, Washington, DC 20375-5350, roger.gauss@nrl.navy.mil)

High false alarm rates are a persistent problem for ASW active sonars. This talk describes a recently developed sonar clutter characterization and control method, the "Poisson-Rayleigh Inspired Method" (PRIM). The closed-form PRIM is based on NRL's two-parameter Poisson-Rayleigh (P-R) model that, like the popular K-type model, provides a physical context for relating the characteristics of data distributions to scatterer attributes (density and relative strength). However, with its extra degree of freedom, the P-R model offers the potential to exploit more information through higher-order (4th and 6th) data moments, and thus do a better job of characterizing clutter. The technique is demonstrated on normalized, broadband sonar clutter data collected in two range-dependent shallow-water environments, one bottom dominated and one fish dominated. The data results suggest that in contrast to the popular K-distribution, the power of the discrete-scatterer component in the P-R distribution can provide feature information that is largely independent of the peak signal-to-noise ratio of an echo, implying a potential for improved rejection of clutter relative to target-like objects. [Work supported by ONR.]

10:10–10:30 Break

10:30

2aSP8. Underwater target tracking using the waveguide invariance. Lisa M. Zurk (NEAR-Lab., Portland State Univ., 1900 SW 4th, Portland, OR 97201)

Target tracking in underwater environments is complicated by false alarms introduced from scattering from the sea bottom and sea-air interface, as well as returns from biologics in the water column. It is difficult to devise a method to discriminate the false alarms from targets of interest because their nature is highly dependent on the environmental conditions, which are often poorly known. A topic of recent interest in active sonar is the application of the waveguide invariant to improve target discrimination and tracking. The invariant relationship gives a robust (i.e., not environmentally dependent) method of relating the time-evolving frequency content to attributes of the target. In this talk, the concept of target discrimination and tracking using the waveguide invariant is discussed in the context of other environmentally robust techniques. Results are presented for fixed and towed array systems.

10:50

2aSP9. Integrated approaches to tracking in cluttered environments. John S. Allen, Grant Pusey (Dept. of Mech. Eng., 2540 Dole St., Holmes 302, Honolulu, HI 96822), John Gebbie, and Martin Siderius (Dept. of ECE, NEAR Lab., Portland State Univ., Portland, OR 97207)

A variety of acoustic applications encompass the detection and tracking of signals in highly cluttered environments. Despite the interdisciplinary nature of this problem, often novel approaches and advances are not well known and hence not typically applied outside their respective sub-fields. In this study, we highlight array processing, time frequency analysis and non-stationary signal processing techniques with respect to applications in both physical and underwater acoustics. In particular, we examine the acoustic tracking of small vessels, divers and AUVs in shallow water, harbor areas. An examination of the underlying physical acoustics of the ambient noise sources is a significant factor in the development of improved and novel tracking methods. Time scale filters are investigated for ambient noise reduction from snapping shrimp. The signal processing advantages of a combined two array systems in an L shaped configuration are discussed. Theoretical predictions and simulations are compared with experiments results from a synchronized system of two 24 element arrays deployed at the Kilo Nalu Nearshore Observatory (Honolulu, HI) near the Honolulu Harbor. Novel acoustic tracking techniques based on depth are explored with the two array systems. [Work sponsored by DHS.]

11:10

2aSP10. Acoustic tracking using compressive sensing. Geoffrey F. Edelmann and Charles F. Gaumond (U. S. Naval Res. Lab., 4555 Overlook Ave., SW, Code 7140, Washington, DC 20375, geoffrey.edelmann@nrl.navy.mil)

This paper presents the application of compressive sensing to several problems of acoustic detection and localization on numerical and at-sea data. Compressive sensing results are shown for the detection of ship tones, bearing estimation from horizontal towed array, and target detection using a vertical array. Compressive sensing is established to be robust to low signal-to-noise ratio, to require few snap shots, and to need few sensors to achieve high probability of detection and low probability of false alarm. As a technique, compressive sensing appears insensitive to noise anisotropy, noise spectral coloration, and mild signal deviations from the sparseness paradigm. This technique potentially applies to a broad spectrum of acoustic applications. [Work supported by the Office of Naval Research.]

Contributed Papers

11:30

2aSP11. Tracking a defect in the multiple scattering regime. Planes Thomas, Larose Eric (ISTERRE, CNRS, & UJF, BP 53, 38041 Grenoble cedex 9, France), Rossetto Vincent (LPMMC, CNRS, & UJF, BP 166, 38042 Grenoble cedex 9, France.), and Margerin Ludovic (Universit de Toulouse, CNRS, Institut de Recherche en Astrophysique et Planétologie, 31400 Toulouse, France)

We describe a time-resolved monitoring technique for heterogeneous media, especially multiply scattering media. Our approach is based on the spatial variations of the cross-coherence of diffuse waves acquired at fixed positions but at different dates. The technique applies to all kind of waves, but a particular attention will be paid to ultrasound propagating in concrete. To locate and characterize a defect occurring between successive acquisitions, we use a maximum likelihood approach combined with a diffusive propagation model. We quantify the performance of this technique called LOCADIFF with numerical simulations. In several illustrative examples, we show that the change can be located with a precision of a few wavelengths and that its effective scattering cross-section can be retrieved. We investigate how the accuracy and precision of the method depends on the number of source–receiver pairs, on the time window used to compute the cross-correlation and on the errors in the propagation model. Applications can be found in nondestructive testing (civil engineering), seismology, radar, and sonar location.

11:45

2aSP12. Passive acoustic tracking of marine mammals and anthropogenic sound sources with autonomous three-dimensional small-aperture arrays. Martin Gassmann, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0205)

Marine mammals produce a wide range of frequency-modulated sounds at low and high frequencies as well as directional broadband echolocation sounds in a refractive ocean environment. This creates several challenges for passive acoustic long-term tracking of the various marine mammal species. To overcome these, three-dimensional small-aperture hydrophone arrays coupled to seafloor multi-channel recording packages were deployed in a large aperture array in the Southern California Bight. Taking advantage of the experimental setup in the oceanic waveguide, time and frequency-domain tracking methods will be presented and tracks of marine mammals as well as anthropogenic sources will be shown. This provides a tool to study over long timescales behavioral responses of tracked marine mammals to tracked anthropogenic sources.

Session 2aUWa

Underwater Acoustics and Acoustical Oceanography: Theory and Practical Applications for Bottom Loss I

Nicholas P. Chotiros, Cochair
Applied Research Lab., Univ. of Texas, P.O. Box 8029, Austin, TX 78713

Martin Siderius, Cochair
Electrical and Computer Engineering, Portland State Univ., 1900 S.W. 4th Ave., Portland, OR 97201

Roger W. Meredith, Cochair
U.S. Oceanographic Office, Stennis Space Center, MS 39529

Chair's Introduction—7:55

Invited Papers

8:00

2aUWa1. Sediment acoustic models and Biot's theory. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, TX 78713-8029)

The development of geo-physical descriptions of the sediment and models to predict the acoustic properties are critical to the application of sediment acoustics. The former follow standard geo-physical methods of sediment classification, based on grain size, density, and other physical descriptors. The latter started as a fluid approximation, followed by a visco-elastic approximation with five frequency-independent parameters, consistent with sediment acoustic data up to the 1980s. Recent experimental data have revealed deficiencies in this approach, particularly in the case of sandy sediments, which cover a large fraction of the continental shelves. The measurements are more consistent with a poro-elastic model, consisting of Biot's theory with extensions to account for the particular physics of granular media. There are currently two approaches to the remedy: (a) a visco-elastic model with frequency dependent parameters that mimic the experimental data and (b) a poro-elastic model with the necessary attributes. It is shown that (a) would be a significant improvement over existing models, but (b) is the preferred solution. A recent discovery concerning the viscosity of nano-meter water films has resolved a problem with the dimensions of the grain contact gap. Future plans will center on further rationalization and reduction of input parameters to develop a practical poro-elastic model. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

8:20

2aUWa2. Bottom loss from geoacoustic inversions. N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada)

The concept of bottom loss has traditionally been used as a measure of acoustic reflectivity and transmission at the ocean bottom. The measure was generally derived experimentally from measurements of transmission loss versus range, using a simple model that assumed perfect reflection at the sea floor to process the transmission loss data of bottom reflected signal paths. More recently, the interaction of sound with the ocean bottom has been described in terms of a geoacoustic model that defines a physical structure of sound speed, density and attenuation in the material beneath the sea floor. The model parameters are inferred from acoustic field data or observables derived from the field using sophisticated inversion techniques. The estimated model can be compared with bottom loss measurements through calculations of the plane wave reflection coefficient. This paper illustrates comparisons of bottom loss measurements for low- and high-frequency bands (50 –20,000 Hz) and calculations from estimated geoacoustic profiles for deep and shallow water environments. Conditions are discussed that limit the performance of present day inversion techniques: rough interfaces on and below the sea floor, consolidated material that supports shear wave propagation and range variation of sub-bottom structure. [Work supported by ONR Ocean Acoustics Team.]

8:40

2aUWa3. A bottom-sediments province database and derived products, Naval Oceanographic Office. Peter Fleischer, William M. Becker, and Peter D. Baas (Naval Oceanogr. Office, Stennis Space Ctr., MS 39522, peter.fleischer@navy.mil)

The Naval Oceanographic Office maintains a bottom-sediment province database at three levels of resolution: (1) a low-resolution legacy data set derived from secondary sources for the low- and mid-latitude ocean basins; (2) medium-resolution, actively maintained ''Regional Sediments'' data sets covering most of the continental margins of Eurasia and North America; and (3) high-resolution, limited-extent data sets derived from acoustic imagery. Sediment provinces are categorized via an ''Enhanced'' sediment classification consisting of seven locational, ten compositional, and 97 textural components. The resultant large number of categories allows for retention of source-data nomenclature but can be unwieldy and redundant. To provide accessibility and consistency to a variety of users, the

Enhanced categories are reclassified into simplified or specialized sediment category sets. These include the High Frequency Environmental Acoustics categories for performance prediction, a mean grain-size set, a non-technical littoral set, as well as other bottom characteristics such as burial potential and bottom type. The Enhanced categories have also been assigned a limited number of geoacoustic and physical properties. Although the province approach and the reclassification schemes produce useful inputs for models and other predictions, users must recognize that they are subject to inherent limitations and ambiguities.

9:00

2aUWa4. Bottom loss measurements and their use in studying the physics of sound propagation in marine sediments. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA 16804)

Bottom reflection loss measurements have been conducted for four decades and have provided key insights into the physics of propagation in marine sediments. The advantage of bottom loss as an analysis quantity is that, in principle, it completely isolates the role of the seabed and permits separation and identification of key physical mechanisms. For example, it was early deep water bottom loss measurements that first made it clear that strong positive sediment sound speed gradients existed and the concomitant surprising realization that the dominant seabed interacting path was the refracted and not reflected path over a wide angular and frequency band. In this talk, various discoveries from bottom loss measurements are discussed including the role of sediment layering, attenuation gradients, and shear waves.

9:20

2aUWa5. Variability in normal-incidence acoustic response in shallow-water marine sediments. Roger W. Meredith (U. S. Naval Oceanogr. Office, Stennis Space Ctr., MS 39529), Richard W. Faas (Dept. of Marine Sci. Univ. of Southern MS, SSC, MS 39529), and Douglas N. Lambert (Naval Res. Lab., Stennis Space Ctr., MS)

Due to several processes, lateral deviations in geotechnical properties of sediments occur at many spatial scales and in all types of sediments. These deviations influence sediment bulk elasticity and density and determine the degree and type of acoustic response. Small changes in one or more properties can produce a wide variation in the acoustic response and empirical curve fitting most often serves as models for these relationships. Sedimentary data and acoustic variability at 30 and 50 kHz from three sites in the Mississippi Sound [Lambert *et al*., Mar. Geol. **182** (2002)] have been further analyzed and compared with the available Shepard's sediment classes. Initial observations revealed trends in sediment variability that affect the accuracy of sediment classifications based on acoustic response. This analysis deviates from traditional sediment classification schemes and utilized hierarchical clustering with user-selected combinations of geotechnical parameters to group individual sediment samples. The sediment properties are correlated with the acoustic coefficient of variation of the ensemble-averaged acoustic response and with the group-averaged acoustic response. The variability of sediment properties and acoustic response show varying degrees of correlation between parameters of major and minor importance.

9:40

2aUWa6. Milestones in naval through-the-sensor environmental acoustic measurements. Martin Barlett, Walter E. Brown, and Joe M. Clements (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713)

This paper describes recent developments in the Bottom Loss and Scattering Strength Measurement System (BLOSSMS). The objectives of the BLOSSMS project are to (1) create technology from which measurements of several types of active sonar scattering phenomena may be made using a single unifying measurement and analysis scheme, (2) integrate measurements from Fleet active sonar systems with seafloor bathymetry and modeled acoustic data into a single root database, (3) develop algorithms that are based on sound physics and oceanographic principles to identify salient information in the root database for analysis, and (4) produce acoustic scattering parameters that are useful in sonar system performance analysis and the optimization of sonar operations. In this paper, the methods used to measure bottom backscatter strength, bottom forward scatter (bottom loss), and volume scatter strength are discussed. Examples of the resulting scattering parameters are provided. Also discussed are the advantages, disadvantages, and challenges that are associated with this approach to measuring fundamental acoustic scattering phenomena.

10:00–10:15 Break

Contributed Papers

10:15

2aUWa7. Bottom loss modeling and sand grain size. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, misakson@arlut.utexas.edu)

Bottom loss is a critical parameter in sonar performance assessment and propagation modeling. There exist many empirical models that predict bottom loss based both on measured parameters and on sand grain size. In this study, the empirical model predictions are compared with in situ measurements of bottom loss. The predictions are based on both the measured parameters and the sand grain size. It is found that at mid frequencies, the predictions based on sand grain size are much more consistent with the measurements than the model based on the measured parameters. This is due to unrealistic empirical sediment parameters such as density which mimic other physical processes not included in the model. However, when the empirical sand grain size model is extrapolated to predict bottom loss at higher frequencies, the results are inconsistent with measurements. [Work supported by ONR, Ocean Acoustics.]

10:30

2aUWa8. Comparative analysis of mid-frequency bottom loss derived from two distinct measurement paradigms. Martin Barlett, Walter E. Brown (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713), Carolyn Harris (Appl. Op. Res., Inc., Solana Beach, CA 92075-2077), and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR 97201)

This paper describes a comparison of acoustic mid-frequency bottom loss data derived from two distinctly different measurement methods, systems, and platforms. One data set was derived from measurements made by a passive drifting line array. The second data set was derived from measurements made from a Navy destroyer equipped with an AN/SQS-53C active

sonar system that utilizes a hull-mounted volumetric cylindrical array. Data were collected off the west coast of the United States. Detailed comparisons and analysis of the derived co-located bottom loss data were made. The metric used for comparison was the difference of the estimated MGS curves at specific co-located sites. The set of MGS curves used as the metric for comparison between the two methods ranges from MGS-1, corresponding to a low-loss bottom, to MGS-9, corresponding to a high-loss bottom, in one tenth increments, i.e., MGS-1.1, MGS-1.2, to MGS-8.9, MGS-9.0. The results from the two methods were found to be in excellent agreement.

10:45

2aUWa9. Role of rattlers in bottom loss and sound wave propagation. James G. Berryman (Earth Sci. Div. Lawrence Berkeley Natl. Lab., One Cyclotron Rd., MS90-R1116, Berkeley, CA 94720)

It is known from both laboratory and numerical experiments that weakly or poorly consolidated granular media have some fraction of the grains that do not contribute to the overall mechanical strength of the granular system. Such loose grains are sometimes called rattlers, and their presence affects both acoustic wave speeds and attenuation. Sound wave speeds tend to be reduced while the wave attenuation tends to be increased. An analytical model of such systems will be presented and comparisons to available data will be shown.

11:00

2aUWa10. Analysis of through-the-sensor observed and modeled reverberation using sensor derived scattering parameters. Martin Barlett, Walter E. Brown (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713), and Ruth E. Keenan (SAIC, Austin, TX 78758)

Recent developments in the bottom loss and scattering strength measurement system (BLOSSMS) program support the use of on-scene measurement of acoustic scattering parameters important to the performance of mid-frequency sonar systems. These parameters could include bottom and volume scatter strength (backscatter), and bottom loss. In this paper, examples of BLOSSMS-derived bottom loss and backscatter estimates for data collected off the west coast of United States are used to compute reverberation levels for select locations and bearings. Included in the discussion are analysis of the oceanographic conditions and relevant bathymetric features.

11:15

2aUWa11. Implications of the presence of shell hash on the speed and attenuation of sound in water-saturated granular sediments. Theodore F. Argo IV and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292)

The propagation of sound through water-saturated granular sediments has been widely studied, yet there is no consensus on an expected wave speed and attenuation in these materials owing to variation in both physical properties and test methods used for their interrogation. One aspect confounding model predictions is the presence of inhomogeneities such as rocks, bubbles, and biological organisms in an otherwise homogeneous sediment. In a laboratory setting, it is possible to control both manufacture of artificial sediments with specific inclusions and the measurement method used to observe their properties. To study the effect of inclusions on the speed of sound and attenuation in an otherwise homogeneous sediment, shells were systematically added to a sand sediment. The volume fraction of shells relative to sand grains was varied and the speed of sound and attenuation was measured using a time of flight technique from 200 to 800 kHz. Results will be compared to both sediment acoustic models and scattering models.

11:30

2aUWa12. Direct measurements of sediment sound speed at mid- to high-frequency in a sand sediment. Jie Yang, Brian T. Hefner, and Dajun Tang (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Direct measurements of sediment sound speed were made near Panama City, Florida in April 2011. Considerable effort is being made to provide detailed environmental characterization for an upcoming mid-frequency sound propagation and reverberation experiment in 2013 and the measurements presented here are part of that effort. Two direct measurement systems are shown: one is called the Sediment Acoustic-speed Measurement System (SAMS) and the other is the attenuation array which was deployed in the Sediment Acoustics Experiment 2004. SAMS consists of ten fixed sources and one receiver. The receiver is driven into the seabed by a motor, which allows precise penetration depth up to 3 m with arbitrary step size. Measurements were made in the frequency range of 1 – 50 kHz. The attenuation array consists of two transmitters and two receivers mounted on a diver-deployable frame and can provide surficial sediment sound speed and attenuation to a depth of about 10 cm between 40 and 260 kHz. Sediment sound speeds obtained using the two systems can be compared in the overlapping frequency region for consistency. Initial analysis of sediment sound speed will be shown. [Work supported by ONR.]

11:45

2aUWa13. High frequency dispersion model for the water-saturated granular medium. Haesang Yang (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul 151-742, Korea, coupon3@snu.ac.kr), Keunhwa Lee (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, kel003@ucsd.edu), and Woojae Seong (Seoul Natl. Univ., Seoul 151-742, Korea)

Dispersion relation for the p-wave sound speed and attenuation has been described by several models based on continuum or scattering theory. As an alternative approach, this study proposes a model describing the dispersion relation for the p-wave in case of grain scatterers existing in background porous medium. Dispersion relations are shown as a function of different grain size distribution and Rayleigh parameter ka. For quantitative analysis of the proposed model, experiments are performed using water-saturated glass beads. Two sets of experiments employing unimodal and bimodal grain size distributions are performed and are used for comparison with the current proposed model.

Session 2aUWb

Underwater Acoustics: Acoustic Prediction and Real-Time Sonar Signal Processing for Vehicle Autonomy Applications

Kevin D. LePage, Cochair
NATO Undersea Research Centre, Viale San Bartolomeo 400, La Spezia, 19126 Italy

Henrik Schmidt, Cochair
Dept. of Mechanical Engineering, Massachusetts Inst. of Technology, 77 Massachusetts Ave., Cambridge, MA 02139

Chair's Introduction—7:45

Invited Papers

7:50

2aUWb1. Pragmatic model-based adaptation for optimal acoustic communication and sensing on autonomous marine vehicles. Toby E. Schneider and Henrik Schmidt (Ctr. for Ocean Eng., Dept. of Mech. Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, tes@mit.edu)

Autonomous underwater vehicles can be valuable acoustic sensing platforms due to their maneuverability, low cost, and sensor-driven adaptivity compared to ships. However, the logistics of integrating acoustics research into artificially intelligent systems can be daunting. In this work, the autonomy software provides an abstract representation of the acoustic environment (e.g., sea surface, water column, and sea floor parameters, source and receiver positions) which it updates continuously from local and remote sensor data. Upon request, this environment is translated into the native representation of an acoustic model which returns the requested calculation (e.g., transmission loss, travel times). The model is set up as a server, capable of handling requests from multiple autonomy subsystems at once (e.g., target tracking prediction, acoustic communications optimization). Thus, the acoustic model and the autonomy software are kept ignorant of each other's implementation specifics. Results will be presented from the shallow water GLINT10 experiment where a vehicle adaptively tracked in depth the minimum modeled transmission loss from a buoy source. Furthermore, a deep sea simulation study which combines target tracking and acoustic communications was conducted. For both studies, the autonomy software MOOS-IVP and the BELLHOP ray tracing codes were used.

8:15

2aUWb2. Deployment of wideband bio-inspired sonars for autonomous underwater operations. David M. Lane, Chris Capus, Keith Brown, and Yan Pailhas (Ocean Systems Lab., Heriot-Watt Univ. Edinburgh, Scotland, EH14 4AS United kingdom)

Recent developments in the understanding and application of wideband bio-inspired acoustic sensors are enabling new applications of autonomous underwater vehicles (AUVs) for security and oilfield use. Appropriate use of wideband signals has enabled improved discrimination between natural and man-made objects that have physically similar appearance, as well as the detection of buried and partially buried objects. Wideband sonar developments in the Ocean Systems Laboratory at Heriot-Watt University have focused on prototypes based on the bottlenose dolphin sonar, covering a frequency band from around 30 to 150 kHz and having a frequency dependent beam-width that is, considerably larger than conventional imaging sonars. In parallel, AUV technology has developed to allow much greater levels of autonomy in allowable vehicle behavior. New generations of vehicle have moved beyond switching of pre-programmed scripts (behaviors) and their parameters, to systems that interleave planning and execution at a fine grain, or can re-plan mission sections on the fly in response to unexpected events. Combining this improved sensor performance with increased autonomy has enabled a first generation of fielded system to act responsively in tasks such as tracking and inspection of proud or buried cables, and the detection and characterization of mines, beyond being mine like objects. Second generation systems that use their autonomy to optimally take multiple looks and control the spectral content of pings in an environmentally adaptable way are under consideration.

8:40

2aUWb3. Bearings-only target localization for unmanned underwater vehicles. Donald P. Eickstedt (iRobot Corp., 8 Crosby Dr., Bedford, MA 01730, eicksted@mit.edu)

This paper reports on target localization for autonomous underwater vehicles (AUVs) with acoustic bearing sensors where the bearing sensor provides detection and beamforming capabilities and reports target bearings with the associated measurement uncertainties. Estimating the state of a moving target using a bearing sensor on a moving platform is a difficult problem due to both the nonlinear nature of the measurement equations with respect to the unknown target parameters but also the observability conditions for the unknown parameters which require the observer to maneuver. This paper will report on a maximum likelihood (ML) method using Levenberg–Marquardt (LM) optimization for estimating the target state parameters. A known issue with ML solutions to this problem is that the algorithm may have difficulty reaching the global minimum unless the initial solution guess is close to the true solution. This paper will report on a genetic algorithm approach to making the initial solution guess. The performance of this approach, obtained from MATLAB

simulation, is presented. The implementation of the algorithms in the MOOS architecture is described using a very fast LAPACK implementation for the LM parameter estimation to support the goal of optimal AUV maneuvering to improve the target state estimate.

9:05

2aUWb4. Adaptive ocean sampling for acoustic ocean uncertainty reduction. Kevin D. Heaney and Richard L. Campbell (Ocean Acoust. and Instrumentation Systems, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

For many oceanic environments, acoustic propagation is critically dependent upon the local environment through boundary interaction and volume variability and fluctuations. In shallow water, near the continental shelves, ocean dynamics and variability can lead to significantly different propagation regimes on timescales of hours and spatial scales of kilometers. Optimal adaptive sampling, with assimilation of oceanographic variables into a dynamic ocean model, can be used to reduce the model forecast uncertainty and improve the confidence in acoustic predictions. Oceanographic uncertainty and acoustic sensitivity, however, are not always well correlated. In this paper, an acoustic uncertainty cost function will be used to drive the adaptive sampling of an oceanographic AUV in coastal water. Applications to an experiment off the coast of Taiwan (quantifying, predicting, and exploiting uncertainty) will be performed, with model-data comparisons of acoustic propagation for moving sources and drifting receivers. During the test, acoustic adaptive sampling computations were performed in real time, oceanographic measurements were made, assimilated into a dynamic ocean model and acoustic transmissions were recorded. [Work Supported by ONR.]

9:30

2aUWb5. Ambient noise modeling for high fidelity acoustic simulation. Andrew J. Poulsen (Appl. Physical Sci. Corp., 49 Waltham St., Ste. 4, Lexington, MA 02421, apoulsen@aphysci.com) and Henrik Schmidt (MIT, Cambridge, MA 02139)

A high fidelity acoustic simulator has been developed to enable realistic simulation of array time series in order to test sensing/processing algorithms. This simulator, capable of generating calibrated hydrophone or vector sensor array data, is fully integrated into a three-dimensional hydrodynamic array model, generating sensor time series for dynamically evolving array shape, location, and orientation. Furthermore, the simulator handles a changing number and configuration of acoustic sources, targets, and receivers while utilizing the legacy ray tracing code BELLHOP to account for ocean multipath effects. Properly modeling ambient noise is particularly important when determining the effect of noise on sensor/processing performance, e.g., array gain can vary significantly based on the directionality of ambient noise. Embedded in the simulator is the capability to efficiently generate broadband noise for an arbitrary noise intensity distribution as a function of depth and elevation angle (azimuthally symmetric ambient noise) using theoretical expressions for the array covariance structure. This approach is ideal for modeling surface noise in situations where the array covariance expressions need to be repeatedly recomputed to account for changing array shape. Results illustrate the advantages of the proposed approach for generating high fidelity time series to model real-world complexities in an efficient, elegant manner.

Contributed Papers

9:55

2aUWb6. Results of the development of a long-range acoustic homing system for autonomous underwater vehicles. G. J. Heard, N. Pelavas, C. E. Lucas, and R. Fleming (Defence RD Canada Atlantic, PO Box 1012, Dartmouth, NS, Canada B2Y 3Z7, garry.heard@drdc-rddc.gc.ca)

Modified international submarine engineering (ISE) explorer AUVs with an endurance of over 400 km are being used to aid in the mapping of the under-ice Arctic seafloor. The explorers are equipped with a DRDC-developed acoustic homing system built into the AUV nose cone. The acoustic receiver consists of seven digital hydrophones arranged in a tri-axis cross-dipole array. A controller/data processor located within the AUV pressure hull handles the real-time acoustic arrival azimuth and elevation estimation, as well as the control and calculations for an on-demand short-range three dimensional (3-D) localization system, and the control of vehicle telemetry data flow. The processor and array consume less than 2 W. A small, easily transportable, DRDC-designed acoustic transducer provides a powerful acoustic homing signal. Using the homing system, the vehicles were able to locate an acoustic beacon at a randomly drifting Ice Camp from a range in excess of 50 km (100-km ranges possible). The design, development, and use of the homing system are described in this paper. In addition, on-going software improvements providing enhanced capabilities and system miniaturization are described.

10:10–10:25 Break

10:25

2aUWb7. Interfaces between acoustic prediction, embedded signal processing, and behaviors at NATO Undersea Research Centre. Kevin D. LePage, Francesco Baralli, Robert Been, Ryan Goldhahn, Michael J. Hamilton, Stephanie Kemna, Michele Micheli, Juri Sildam, and Arjan Vermeij (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, SP Italy)

The use of acoustic sensing systems for ASW in heterogeneous sensor networks utilizing marine robots has been a subject of research at the NATO Undersea Research Centre for the past several years. In this talk, we discuss the unique challenges of implementing ASW on autonomous, collaborative networks of AUVs, including the challenges of embedding the active sonar signal processing, implementing effective underwater messaging, and designing adaptive behaviors to optimize system performance. Theoretical studies, simulations, and results from the recent GLINT series of sea trials are shown and the way forward for autonomous sensor system studies at NURC is discussed.

10:40

2aUWb8. Application of the range dependent waveguide invariant distribution processor to unintentionally radiated broadband noise from surface ships in a range- and azimuthally dependent environment. Alexander W. Sell and R. Lee Culver (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., Univ. Park, PA 16802, aws164@psu.edu)

Waveguide invariant analysis is a useful tool for understanding spectral interference patterns from broadband sources in shallow water waveguides. These interference patterns (in the form of intensity striations) were observed in time-frequency plots of surface ships passing a horizontal line array located along the continental shelf off southeast Florida during a 2007 acoustical experiment. Previous work has shown that results from the Range Dependent Waveguide Invariant Distribution (RaDWID) method match well with simulations from parabolic equation acoustical models and require significantly less computation time; however, work to understand the processor's ability to recreate real data was incomplete. We will discuss how RaDWID processing can be applied to ship-radiated broadband noise to model spectral interference patterns. The implementation of the processor on these data including handling of environmental parameter uncertainty, broadband source power spectrum, and environmental features will also be discussed. [Work supported by ONR Undersea Signal Processing.]

10:55

2aUWb9. Environmentally sensitive autonomous underwater vehicle (AUV) behavior design. Kevin D. LePage (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, SP Italy)

The design of environmentally adaptive behaviors for AUVs conducting active ASW is discussed within the framework of the MOOS-IvP middleware. The control of a single or a group of autonomous, collaborative AUVs in the underwater environment requires the installation of on-board autonomy to enable the vehicles to optimize their trajectories and strategies as they prosecute underwater targets. As the underwater environment is harsh and large differences in performance can be anticipated as a function of lateral position, speed, and depth, a suite of environmentally sensitive behaviors has been developed to provide the required autonomy. Results from simulation and at-sea experiments will be shown.

11:10

2aUWb10. Real-time sonar signal processing on-board an autonomous underwater vehicle. Michael Hamilton and Michele Micheli (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, Italy, hamilton@nurc.nato.int)

An active, bistatic signal processing system has been implemented for use on-board autonomous underwater vehicles (AUVs) using towed arrays. The NATO Undersea Research Centre's (NURC) AUVs are programmed to maneuver in order to best track a target. To perform this action autonomously, the vehicle must be able to fully process and track targets via its towed array data in real time or faster. The processor implemented includes processing from the array hydrophone data, through beamforming, matched filtering, Doppler processing, and tracking. This system adapts many previously developed algorithms to function in real time. Research and algorithm adaptations in the areas of normalization, CFAR detection, and array navigation, have also been developed to deal with practical issues which have arisen in sea trials. The implementation, practical issues, and steps taken to address them will be presented. This research is supported by the NURC Consolidated Programme of Work, Cooperative ASW program.

11:25

2aUWb11. Supervised machine learning for real time classification of underwater targets from sampled scattered acoustic fields. Erin M. Fischell and Henrik Schmidt (Ctr. for Ocean Eng., Dept. of Mech. Eng., MIT, 77 Mass. Ave., Bldg. 5-223, Cambridge, MA 02139)

Autonomous underwater vehicles (AUVs) are often used for bottom surveys in the ocean, but the collected data is usually processed on shore. The ability to identify and classify targets while in flight would allow more effective use of mission time and reduce the need for additional surveys. It is demonstrated that geometry can be classified from hydrophone sampling of the scattering field off of a target in the presence of an acoustic source in real time simulation. Scattering field data for multiple target shapes is simulated using the OASES and SCATT software packages. This data is then used to produce training data sets for a support vector machine (SVM), a type of supervised machine learning that generates a classifying hyperplane. The trained SVM classifies new data, such as that collected by an AUV in a real or simulated scattered field, with minimal computation by comparing it to the hyperplane. The SVM also identifies divergence between target classes, allowing identification of regions where targets will look the most different to sampling, enabling optimized path planning for classification. This would allow a vehicle looking for a particular geometry to shift its search pattern to best identify relevant features.

11:40

2aUWb12. Time-varying spatial spectrum estimation using a maneuverable sonar array. Jonathan L. Odom and Jeffrey L. Krolik (Dept. of Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, jonathan.odom@duke.edu)

This paper addresses the problem of spatial spectrum estimation in dynamic environments with a maneuverable sensor array. Estimation of the time-varying acoustic field directionality is of fundamental importance in passive sonar. In this paper, mobility of the array is treated as a feature allowing for left-right disambiguation as well as improved resolution toward endfire. Two new methods for on-line spatial spectrum estimation are presented: (1) recursive maximum likelihood estimation using the EM algorithm and (2) time-varying spatial spectrum estimation via derivative-based updating. A multi-source simulation is used to compare the proposed algorithms in terms of their ability to suppress ambiguous array backlobes. A broadband method is presented utilizing knowledge of the source temporal spectrum. Detection performance of weak high-bearing rate sources in interference-dominated environments is evaluated for a flat spectrum. [This work was supported by ONR under grant N000140810947.]

Meeting of the Standards Committee Plenary Group
to be held jointly with the meetings of the
ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machines,
ISO/TC 108/SC 6, Vibration and shock generating systems,
and
IEC/TC 29, Electroacoustics

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

D. J. Evans, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108 Mechanical vibration, shock and condition monitoring, and ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

W. C. Foiles, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
BP America, 501 Westlake Park Blvd., Houston, TX 77079

R. Taddeo, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
NAVSEA, 1333 Isaac Hull Ave., SE, Washington Navy Yard, Washington, DC 20376

D. D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Ct., Las Vegas, NV 89120

D. J. Vendittis, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
701 Northeast Harbour Ter., Boca Raton, FL 33431

R. Taddeo, Vice Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

C. Peterson, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 6 Vibration and shock generating systems
200 Dixie Ave., Kalamazoo, MI 49001

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Dr., Stop 8221, Gaithersburg, MD 20899-8221

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

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3, S3/SC 1 and S12, which are scheduled to take place in the following sequence:

Tuesday, November 1, 2011	10:30 a.m.–11:30 a.m.	ASC S2, Mechanical Vibration & Shock
Tuesday, November 1, 2011	1:00 p.m.–2:00 p.m.	ASC S1, Acoustics
Tuesday, November 1, 2011	2:15 p.m.–3:30 p.m.	ASC S12, Noise
Tuesday, November 1, 2011	3:45 p.m.–5:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Wednesday, November 2, 2011	8:30 a.m.–9:45 a.m.	ASC S3, Bioacoustics

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.
The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SC1** Noise	ASC S12
D.J. Evans, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
W.C. Foiles, Co-Chair R. Taddeo, Co-Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair R. Taddeo, Vice Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machines	ASC S2
C. Peterson, Chair	**ISO/TC 108/SC6** Vibration and shock generating systems	ASC S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

TUESDAY MORNING, 1 NOVEMBER 2011 — CRESCENT, 10:30 TO 11:30 A.M.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

A. T Herfat, Chair ASC S2
Emerson Climate Technologies, Inc., 1675 West Campbell Rd., PO 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 vibration, shock and condition monitoring, and its five subcommittees, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, November 1, 2011.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

Session 2pAA

Architectural Acoustics: Update on Acoustic Products, Treatments, and Solutions

Matthew V. Golden, Cochair
Kinetics Noise Control, 6300 Irelan Pl., Dublin, OH 43017-0655

Kenneth W. Good, Jr., Cochair
Armstrong World Industries, Inc., 2500 Columbia Ave., Lancaster, PA 17603

Chair's Introduction—1:15

Invited Papers

1:20

2pAA1. Products focused on improving acoustic performance. Ron Freiheit (Wenger Corp., 555 Park Dr. Owatonna, MN 55060, ron.freiheit@wengercorp.com)

New product information with a focus on performance, practice, and rehearsal room acoustics will be presented. Performance-area acoustics are enhanced by full-stage acoustical shells, ceiling panels, and clouds for the audience area. These range from standard products to unique customizations. Portable shells that roll through standard doorways will also be shown. New modular sound-isolating practice rooms feature enhanced acoustic performance, improved aesthetic appearance, and upgraded optional floating floor. Active acoustics technology can be retrofit in an existing room, such as a teaching studio. Interesting applications of acoustical doors will also be highlighted. Other products include instrument storage cabinets with tunable absorption integrated into the design. An acoustic shield positioned behind an individual performer helps reduce potentially damaging sound energy from behind, as indicated by binaural measurements and recordings. Finally, an acoustician-specific, restricted-access website features third-party test data and other commonly requested technical information.

1:40

2pAA2. A re-introduction of pre-stressed molded fiberglass isolation pads for floating floors. Matthew Golden (Kinetics Noise Control, 6300 Irelan Pl., Dublin, OH 43017)

The pre-stressed molded fiberglass isolation pad has been available for over 50 years. In that time, no other material has been developed that exhibits the unique performance features of fiberglass. Fiberglass pads are a very non-linear spring. The dynamic stiffness of fiberglass pads also changes under load. Together these features allow them to provide consistent performance over a very wide load range. Recently, the next generation of pre-stressed molded fiberglass isolation pad has been introduced to the market. This talk will cover what has been improved with this new generation of pad along key performance dynamics of floating floors for acoustical and vibration isolation applications.

2:00

2pAA3. Sound rating of a new air handling product. Stephen J. Lind (Ingersoll Rand, Trane Bldg 12-1, 3600 Pammel Creek Rd., La Crosse, WI 54601)

It is important in building design to have an accurate acoustical description of equipment that will operate in the building. Noise from air conditioning systems is often a significant contributor to the building acoustic environment. In order to design the building correctly, the sound levels produced by the equipment must be known. Current theoretical or computer modeling methods are not able to accurately predict the resulting unit sound levels. The sound levels produced depend on many factors including equipment design, operating conditions, options chosen, and sound component. Empirical models are needed. The test method to develop the models follows Air Conditioning, Heating, and Refrigeration Institute Standard 260. Sound power levels for a range of fan sizes, types, and operating conditions are measured for the discharge, inlet, and casing sound components. Equipment options that affect the sound as it propagates through the equipment are also measured. The test results are used to create mathematical models to describe the unit sound power. The resulting mathematical models are then incorporated into a computer program to allow the user to accurately determine the sound in the product as configured for each job.

2:20

2pAA4. Solutions to retain acoustical functionality against the tide of fad, fashion, and finance. Kenneth Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Too often in recent years, building acoustics have been the casualty of other design considerations. Changes in design styles, flexibility, and building densities have left many building occupants stressed and distracted. The Center for the Built Environment and other

data suggest that the recent Green focus has made the situation even worse as the (acoustical) function of buildings is being compromised to fad, fashion, finance, and other considerations. This paper will look at the acoustical properties of products to meet these changes, solutions to maintain the acoustical functionality of interior spaces.

2:40

2pAA5. Active acoustic treatment. Roger W. Schwenke (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702)

We call building materials "acoustic treatment" when they change the response of a room to sounds that occur in it. Absorption reduces the reverberation time of a room as well as strength. Reverberant chambers increase the cubic volume of the room and increase the reverberation time without significantly changing the strength. Diffusion changes the distribution of reflections in time and level. Electro-acoustic systems using microphones, signal processors, and speakers can also be used to change the response of a room to the sounds that occur in it. This paper will illustrate many ways that Electro-acoustics can be thought of as an active acoustic treatment—one of the many acoustic treatments acousticians have in their palette to achieve their design goals.

3:00

2pAA6. Acoustical performance of lightweight construction materials. Erin L. Dugan (USG Corp., Corporate Innovation Ctr., 700 N. US Hwy. 45, Libertyville, IL 60048, edugan@usg.com)

Traditionally, acoustical privacy between spaces has been achieved by adding denser, heavier materials to the partition. Recently, several lightweight construction materials have been released into the construction market. Advances in material science have provided a new generation of lighter weight gypsum panels with multiple benefits to builders. Although lighter and significantly easier for contractors to install, the reduced mass of the gypsum panels raises concerns about its effects on the sound attenuation performance of these products. However, analyses have shown that these lightweight gypsum panels are able to provide nearly comparable levels of sound transmission performance as their heavier counterparts. An investigation of the intrinsic mechanical properties contributing to the overall acoustical performance and full-scale sound transmission results is presented comparing wall systems constructed with lightweight panels and standard-weight panels. Alternative assemblies used to increase the sound-attenuating performance are also discussed.

3:20–3:35 Break

3:35

2pAA7. Use of damped drywall in architectural acoustics. Benjamin M. Shafer and Brandon Tinianov (Serious Energy, Inc., 1250 Elko Dr., Sunnyvale, CA 94089)

Damped drywall, specifically QuietRock, has been tested both by various NVLAP-accredited laboratories and *in situ* in a variety of assemblies as an acoustic treatment for the transmission of airborne sound through building partitions. The application of damped drywall and its use for noise control through building partitions continue to expand as the body of test data grows and analytic models are developed to support this class of materials. However, the manufacturer of QuietRock has conducted several research studies that illustrate how damped drywall can be used to improve the transmission loss over a broad range of frequencies and in various assemblies. This presentation is a summary of these research studies and serves as a guide for the application of damped drywall in building construction.

3:55

2pAA8. RPG Diffusor Systems: Overview of nearly 30 years of research and development. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Each company is defined by its mission. RPG Diffusor System mission is to provide continuing innovation and expands the acoustical palette through a commitment to pioneering fundamental acoustics research. Rather than focus on any particular products, this presentation will provide an overview of nearly 30 years of research and development of a unique range of absorptive, reflective, and diffusive acoustic tools. With respect to absorption, the presentation will describe the progression from traditional fabric covered porous absorbers, to binary amplitude diffsorbers (attenuating the unneeded high frequency absorption), to absorptive wood Helmholtz absorbers, to transparent microperforated and microslit absorbers, to absorptive/diffusive CMU, membrane and plate resonator absorbers. With respect to reflection, we will describe the advancement from flat reflectors, to combined reflectors, to one and two-dimensional curved shapes in wood and glass reinforced gypsum. Finally, the evolution of diffusive surfaces from the venerable QRD in the early 1980s to modulated optimized diffusors, which provide unlimited bandwidth and freedom from grating lobes and flat plate limitations, to shape-optimized wood and glass reinforced gypsum spline, bicubic and other curvilinear architectural shapes. As part of the historical development, theoretical explanations, proof of performance metrics, and installation photos of completed projects will be provided.

Contributed Papers

4:15

2pAA9. Acoustic characteristics of floor treatments for elementary school classrooms. Ari M. Lesser, Adam P. Wells, Michelle C. Vigeant, and Robert D. Celmer (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117)

Phase 1 of this study [J. Acoust. Soc. Am. **129**(4), 2523 (2011)], determined the effects of hard versus soft flooring on overall speech and activity noise levels in elementary classrooms. A significant decrease in overall levels was found in carpeted rooms. This phase sought to investigate a range of floor materials and their pertinent properties. Nine different floor materials were mounted to 3 in. concrete slabs and evaluated using a battery of acoustic, impact, and chair scrape tests. Tested materials included vinyl composition tile, resilient rubber athletic flooring (virgin, blended/synthetic, and recycled), polyurethane, vinyl cushion tufted textile carpet, and rubber-backed commercial nylon carpet. Impedance tube measurements of sound absorption were made using International Organization for Standardization (ISO) 10534-2, while sound power measurements according to ISO 3741 were made while either (a) tapping on each mounted sample with a standard tapping machine or (b) while reciprocating an elementary classroom chair back and forth to produce repeatable scraping sounds. In general, the two carpet samples resulted in the lowest sound levels and the highest absorption. The relative performance of each material will be presented along with a discussion of additional usability factors, such as maintenance, cost, and durability. [Work supported by Paul S. Veneklasen Research Foundation.]

4:30

2pAA10. The dB focus tube uses to improve the transmission loss efficacy of walls, ceilings, and pipes. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, E. Hampton, NY 11937, bonnie@soundsense.com)

Studies indicate that locating and treating even very small acoustic leakage points in wall, floor, and/or ceiling partitions will significantly improve the FSTC performance of these partitions. The dB focus tube, patented on March 22 of this year, is an inexpensive small mechanism that can easily be transported to and from job sites. It is audible even during high levels of construction activity. The resulting patented small tube can be utilized by an acoustic engineer to locate even the smallest of acoustic leakage points in a partition which, when sealed, could increase the FSTC performance of the partition by typically 10 points. Additionally, when an engineer is trying to determine the main paths of flanking, this tube proves to be an ideal mechanism for facilitating this problem solving method. An acoustic installer can use the focus tube to confirm the optimum efficacy of the acoustic treatment of PVC piping or duct work. In addition, another interesting application is using the dB focus tube to locate air leaks in window assemblies which, when sealed, improve the window performance in regards to not only the STC and air infiltration, but also its thermal capacity.

4:45

2pAA11. Modeling treatments to reduce sound transmission through an open window into a room: Effect of window thickness. Caleb F. Sieck and Siu-Kit Lau (Architectural Eng., Univ. of Nebraska-Lincoln, 203C PKI, 1110 S 67th St., Omaha, NE 68182-0681, cfsieck@gmail.com)

Sound transmission loss through an exterior wall is limited by its weakest structure, generally a window, especially if it is open. Considering the acoustic modes within a window and a room are important because much of the acoustic energy from noise sources such as traffic and large wind turbines is in the low to middle frequency range. Previous models of open windows have either neglected the thickness of the window or the influence of room modes on the sound transmission. The present investigation considered a baffled rectangular aperture of finite thickness backed by a rigid walled cavity. An impedance/mobility approach was used to study the effect of the thickness of an open window on the insertion loss and sound pressure levels inside the cavity. The insertion loss study was confirmed using FEM modeling, and the difference in sound pressure levels was compared to experimental results. Increasing window thickness decreases the amount of sound transmitted at frequencies below the second order modes of the cavity for both window sizes under investigation. Using the impedance/mobility approach was effective in this study and allows the model to be easily extended.

5:00

2pAA12. Increasing diffuser efficiency through asymmetrical third dimension modification. Richard L. Lenz, Jr. (RealAcoustix LLC, 887 N. McCormick Way, Layton, UT 84041, RL@RealAcoustix.com)

In 1975, Dr. Manfred Schroeder presented his seminal work on quadratic-residue diffusion, giving us a frequency-based method for creating acoustic diffusers. The QRD equation defines the method for creating the depths (Z plane) of the wells in a diffuser. Subsequent work has given us understanding of the effects of the width of the diffuser wells (X plane) on high frequency performance. One artifact not addressed in the Schroeder equation is the inherent absorption realized in the execution of the design. This absorption becomes more pronounced as the design gets deeper, thereby limiting the practical low-frequency cutoff of QRDs. Experiments by the author, utilizing an asymmetrical cellular acoustic diffuser design, have yielded surprising results in the efficiency of acoustic diffusion. By reducing both the width of the cells and separating the diffuser into asymmetrical zones in the X and Y (vertical) planes, it has been shown that the absorption below the design cutoff frequency of the diffuser design can be dramatically reduced. The presentation will show the evidences of these experiments and the increased efficiency of QRD design through asymmetrical third dimension modification. [Work support was provided by Ron Sauro of NWAA Labs.]

5:15

2pAA13. Conference rooms: Criteria, guidelines, acoustical problems solutions (case studies). Jack B. Evans and Joshua D. Leasure (JEAcoust., 1705 W Koening Ln, Austin, TX 78756, Evans@JEAcoust.com)

Conference rooms are subject to privacy and sound containment issues, intrusive and distracting noise, excessive continuous background sound, and reflection/reverberation problems, particularly those with microphones and loudspeakers. Good acoustical environments are necessary for intelligible speech communications and remote signal transmissions, while sound isolation is needed for privacy and prevention of distraction. Acoustical criteria are presented with practical guideline parameters. Case studies cover acoustical problem issues that needed correction in existing conference rooms. On-site observations, acoustical measurements and analyses of facility plans were used to diagnose problems and determine correction approaches. Photographs are shown to illustrate difficult conditions. While little is original, the case studies point out classic problems of flutter echoes, ceiling-mounted or suspended microphones, back radiated loudspeaker noise in ceiling plenums, glass walls, multiple sound flanking paths, and similar problems found. Solutions developed or implemented are presented. In some cases, before and after data are provided to show results.

Session 2pAO

Acoustical Oceanography: Tomographic, Geoacoustic, and Ambient Noise Inversions

Shane C. Walker, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

1:00

2pAO1. The build up rate of broadband spatial coherence between multi-sensor passive arrays in the ocean waveguide. Shane Walker (Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093)

Broadband noise correlation methods for the passive extraction of information about the point-to-point propagation of waves between distant sensor locations have received considerable attention in the literature. This talk addresses the rate at which the wave coherence accumulates to overcome stochastic random fluctuations in a spatially correlated random wave field. It is shown that the expected magnitude of the random uncertainty associated with a single realization of the sample cross-correlation function depends on the total power incident on the sensors. This notion is applied to quantify the emergence rate of the coherence between correlated beams in the ocean waveguide. The point-to-point build-up rate is compared to the gain achieved through the application of directional filters over multi-sensor line arrays. Various array geometry scenarios of experimental interest are considered. These results are straight forwardly extendable to other environments such as seismics, helioseismics, and nondestrutive testing.

1:15

2pAO2. Low-frequency broadband noise correlation processing in deep-water. Stephanie Fried (Marine Physical Lab. of the Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238), Karim G. Sabra (Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332-0405), W. A. Kuperman (Univ. of California, San Diego, La Jolla, CA 92093-0238), and Mark K Prior (Preparatory Commission for the Comprehensive Nuclear-Test-Ban Treaty Organisation, 1400 Vienna, Austria)

The Comprehensive Nuclear-Test-Ban Treaty Organization operates an International Monitoring System (IMS). The IMS includes hydroacoustic stations composed of hydrophones deployed in the ocean deep-sound-channel in a two-kilometers-side triangular configuration(referred to as triad) in the horizontal plane. Data are continuously recorded on hydrophone triads (with a sampling frequency of 250 Hz) and have been archived during the last decade. Previous experimental studies have demonstrated that coherent waveform can be extracted from broadband coherent processing of ocean ambient noise, typically above $f > 100$ Hz [e.g., see Roux *et al*., J. Acoust. Soc. Am. **116**(4), 1995–2003 (2004)] . We investigated here the emergence of coherent arrivals from the correlation processing of the low-frequency broadband ambient noise recorded during the years 2006–2007 on IMS hydrophones located in the Southern Hemisphere. This low-frequency acoustic ambient noise includes various components from anthropogenic and biological sources as well as from seismic origin (e.g., earthquakes and microseims) and also significant ice-breaking noise originating from Antarctica especially during the Austral summer period. The feasibility of passive basin scale tomography using long-term monitoring of ocean noise will be discussed.

1:30

2pAO3. Bayesian inversion of seabed interface-wave dispersion from ambient noise. Cuilin Li, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, B.C. V8W 3P6 Canada), Hefeng Dong (Norwegian Univ. of Sci. and Technol., NO-7491 Trondheim, Norway), Lanbo Liu (Univ. of Connecticut, Storrs, CT 06269), and Dingyong Yu (Ocean Univ. of China, Qingdao, China 266100)

This paper applies Bayesian inversion to estimate seabed shear-wave speed profiles and their uncertainties using interface-wave dispersion curves extracted from ocean ambient noise, and compares the resolution of seabed structure for fundamental-mode and multi-mode data. Ambient-noise recordings were collected for 2.15 h at hydrophones of an entrenched ocean-bottom cable in the North Sea. Scholte-wave dispersion curves for the fundamental mode and several higher-order modes within the frequency range 0.26–3.8 Hz are extracted from cross-correlations of noise recordings at sensor pairs via slowness-frequency transform. The Bayesian information criterion is used to determine the preferred model parameterizations in terms of the number of sediment layers supported by the data for inversions based on the fundamental mode alone and on the first three modes. Adaptive-hybrid optimization and Metropolis-Hastings sampling are applied to estimate the most-probable shear-wave speed models and to compute marginal posterior probability profiles. The results show quantitatively that multi-mode inversion provides higher-resolution of shear-wave speed structure at shallow depths and smaller uncertainties at all depths than inversion of the fundamental mode alone. [We thank StatOil for providing data.]

1:45

2pAO4. A computationally light method for simulating the physics and statistics of weak signals in spatially correlated random wave [and vibration] fields. Shane Walker (Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093)

Ocean noise (natural plus man made) incident on an array has coherent and incoherent components. Simulations must accurately include both components so that the synthetic data are faithful to realistic scenarios wherein spatial correlations associated with random wave fields emerge over time. Indeed, noise simulations involving a large number of sensors over long time intervals can be quite computationally expensive. This talk introduces a computationally light method for simulating stochastic (time-domain) realizations of arbitrary duration of spatially correlated noise characteristic of the ocean soundscape. The method is well suited for studying the physics of weak signals in noise and provides an opportunity for studying the emergence of spatial correlations associated with random wave fields. While this talk focuses on passive SONAR simulations, the method can be generally applied to the study of random wave fields and vibrations in other environments such as seismics, helioseismics, and nondestrutive testing.

2:00

2pAO5. Coherent averaging of the passive fathometer response using short correlation time. James Traer and Peter Gerstoft (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr. La Jolla, CA 92093)

The passive fathometer algorithm was applied to data from two drifting array experiments in the Mediterranean, Boundary 2003 and 2004. The passive fathometer response was computed with correlation times from 0.34 to 90 s and, for correlation times less than a few seconds, the observed signal-to-noise ratio (SNR) agrees with a 1 D model of SNR of the passive fathometer response in an ideal waveguide. In the 2004 experiment, the fathometer response showed the array depth varied periodically with an amplitude of 1 m and a period of 7 s consistent with wave driven motion of the array. This introduced a destructive interference, which prevents the SNR growing with increasing correlation time. A peak-tracking algorithm applied to the fathometer response of experimental data was used to remove this motion allowing the coherent passive fathometer response to be averaged over several minutes without destructive interference. Multirate adaptive beamforming, using 90 s correlation time to form adaptive steer vectors which were applied to 0.34 s data snapshots, increases the SNR of the passive fathometer response.

2:15

2pAO6. Performance bounds on geoacoustic inversions. Arthur B. Baggeroer (Depts of Mech. and Elec. Eng., MIT, Cambridge, MA 02139)

Lower bounds on the mean square error for parameter estimates using geoinversion methods are developed in terms of the tomographic version of the Cramer Rao bound. Two approaches are considered: (i) an active, coherent source with known waveform and time synchronization as in ocean acoustic tomography (OAT) and (ii) a passive, broadband source with known spectrum but with no time synchronization as in matched field tomography (MFT). Both approaches assume the receiver to be an array, e.g., vertical line array or towed horizontal line array (HLA). The formulation requires specifying the Green's functions connecting source and the sensors of the receiving array. The bounds demonstrate (i) OAT performance depends upon the mean square group speed spread of modes and (ii) MFT performance depends upon the phase speed spread where both are integrated across the band of the respective sources. The bound also indicates the coupling among parameter estimates. Uncorrelated estimates are usually desirable for efficient parameterization formulations. An example using a Pekeris waveguide with three unknown parameters: the ocean and bottom sound speeds and the depth. The extension to Bayesian problems where a joint a priori probability is given. This permits the evaluation enabled by the observed data.

2:30

2pAO7. Passive geoacoustic inversion in a dispersive waveguide. Julien Bonnel, Cedric Gervaise (ENSTA-Bretagne, Pole STIC, 2 rue F. Verny, 29200 Brest, France, julien.bonnel@ensta-bretagne.fr), Barbara Nicolas, and Jerome Mars (GIPSA-Lab, Image-Signal Dept, Grenoble INP, France)

This study introduces a single-receiver geoacoustic inversion method adapted to shallow water and low frequency sources. Because of the single receiver context, most existing methods are based on the time-frequency (TF) analysis of the received signal, and their practical applications are restricted to impulsive sources. The proposed method is different and allows for geoacoustic inversion using unknown frequency-modulated sources. To perform the inversion, the modes are first filtered from the received signal using advanced TF analysis. Then, the filtered modes are processed in the frequency domain using a new transformation called modal reversal. This transformation, parametrized using environmental information, undoes dispersion for a given mode. When environment is well known, dispersion is perfectly compensated and all the reversed modes are in phase. This is not the case when modal reversal is ill-parametrized. Consequently, modal reversal can be used as the core of an original inversion scheme. Inversion results are obtain through a specific cost function adding up the reversed modes. Source/receiver range and source frequency-law are obtained as a by-product of the geoacoustic inversion. The method can thus be adapted to the study of low-frequency calls of marine mammals in shallow water.

2:45

2pAO8. Seismo-acoustic propagation effects at the seafloor with applications to geoacoustic inversion using hybrid parabolic equation solutions. Adam M. Metzler (Rensselar Polytechnic Inst., 110 8th St., Troy, NY 12180) and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Accurate and efficient prediction of propagation over realistic models of elastic ocean sediments has been achieved recently using parabolic equations. Historic treatments that ignore effects due to elasticity may not be accurate for certain scenarios, specifically for example shallow-water environments. Neglecting elastic phenomena does not account for second order effects such as energy loss due to frequency dependent attenuation. In this presentation, a three-layer ocean environment is examined consisting of a fluid overlying a transitional solid layer overlying an elastic basement. The transitional solid layer is investigated for various types of media, including fluid, poro-elastic, and elastic. Solutions obtained from hybrid parabolic equations are used to quantify the nature of the transitional solid layer and establish regimes where each approximation would be appropriate. An application in geoacoustic inversion to identify bottom type is shown through comparisons of interface wave structure for different transitional solid layers. [Work supported by the ONR.]

3:00–3:15 Break

3:15

2pAO9. Stabilizing reverberation inversion by regression relations involving a grain size parameter. L. Abrahamsson, B. L. Andersson, S. Ivansson, J. Pihl (Swedish Defence Res. Agency, Underwater Research, 16490 Stockholm, Sweden, leif.abrahamsson@foi.se), M. Ainslie, F. Benders, M. Colin, R. van Vossen (TNO Defence, Security and Safety, The Hague, Netherlands), B. Chalindar, X. Cristol, B. Juhel (Thales Underwater Systems, 06560 Sophia-Antipolis, France), J. Dybedal, G. K. Olsen (Kongsberg Defence Systems, 7501 Stjordal, Norway), and E. Eidem (FFI, Forsvarets forskningsinstitutt, N-3191 Horten, Norway)

Within the European Defence Agency (EDA) project Rumble-2, an operational low-frequency active sonar system has been used to collect reverberation data at several sea trials in the North Sea. A global optimization method is used to determine the bottom parameters that provide the best match between measured and modeled time traces. A fast ray model is used for the forward computations. The bottom parameters are the Lambert back-scattering parameter and the sound speed c, density ρ, absorption α, and thickness of the sediment. The reverberation data do not constrain all these parameters to unique values, however, and different approaches have been tried in the project to reduce the ambiguity problems. The approach reported here is to use the mean grain size M_z as a common descriptive parameter. From regression relations by Hamilton and Bachman, c, ρ, and α can be set as functions of M_z. More ambitiously, the regression relations could be applied as *a priori* constraints, with uncertainties, in a Bayesian framework. The obtained inversion results are consistent with ground truth for the grain size, as measured from bottom samples. Moreover, similar results are obtained for trials in the same area with quite different environmental conditions.

3:30

2pAO10. Geoacoustic inversion in shallow water using broadband synthetic aperture and single hydrophone acoustic data. Bien Aik Tan, Peter Gerstoft, Caglar Yardim, and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238)

A majority of geoacoustic inversion experiments sample the acoustic field on long arrays. This paper uses a moving single hydrophone to create a large synthetic aperture array for geoacoustic inversion. Practically, this is operationally attractive compared to using long arrays. For example, one possible application is the use of autonomous underwater vehicles (AUVs) to perform acoustic field sampling and pre-processing for geoacoustic inversion. The approach comprises synthetic aperture formation and a directed Monte Carlo Bayesian broadband frequency coherent geoacoustic inversion. This is demonstrated with simulated and real data from the MAPEX2000 experiment conducted by the NATO Undersea Research Center, using only one hydrophone of a towed array and a moored source in the Mediterranean Sea. The method yielded similar results compared to an equivalent physical array.

3:45

2pAO11. Estimation of shear speed using interface wave dispersion. Jennifer Giard, Gopu R. Potty, James H. Miller, Jeannette M. Greene (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882), Andrew R. McNeese, and Preston S. Wilson (The Univ. of Texas at Austin, 1 Univ. Station C2200, Austin, TX 78712)

Our recent work has highlighted the effect of shear on the dispersion of acoustic normal modes. Specifically, sediment shear speed can significantly impact compressional modal arrival times near the airy phase. In addition to underwater acoustic propagation effects, shear properties of sediments are directly related to the strength of the sediments in geotechnical applications. All of these factors emphasize the importance of estimating shear speeds in semi-consolidated shallow water sediments. One of the most promising approaches to estimate shear speed is to invert the shear speed profile using the dispersion of interface waves (Scholte waves). 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. Results from the tests conducted at Narragansett Bay in water depths ranging from 10 to 25 m using the shear measurement system, developed at the University of Rhode Island based on this concept, will be presented. Combustive sound source (CSS) will be used to generate interface waves. Data collected during these tests will be shown and preliminary estimates of the shear speed will be presented and compared with ground truth data. [Work supported by Office of Naval Research.]

4:00

2pAO12. Perspective of tomography inversion using direction-of-arrival and direction-of-departure. Florian Aulanier (Gipsa Lab., DIS, Grenoble INP, 961 rue de la Houille blanche, BP 46, F-38402 Grenoble Cedex, France, Florian.Aulanier@gipsa-lab.grenoble-inp.fr), Barbara Nicolas (Gipsa Lab, DIS, Grenoble INP, 38402 Saint Martin d'Hères, France), Philippe Roux (ISTerre, Maison des Gosciences, 38400 Saint Martin d'Hères, France), and Jèrôme Mars (Gipsa Lab, DIS, Grenoble INP, 38402 Saint Martin d'Hères, France)

In the ocean, local sound speed variations induce acoustic path changes. Travel-time (TT) variations of acoustic paths are classically used to perform ocean tomography inversion. Initially introduced to cope with multi-arrival interferences and to separate eigenray paths, source–receiver arrays combined with array processing techniques now give access to new observables that could be used for tomography such as direction-of-arrivals (DOAs) and direction of departure (DOD). The cumulative use of TT, DOA, and DOD in the inversion process first requires to study the forward problem which links sound speed variations to these observables measured through array processing from two source–receiver arrays. The so-called sensitivity kernels are established using (1) the first order Born approximation that relates the sound speed variation to the amplitude and phase change of the perturbed received signal and (2) a first order Taylor development which links the received signal perturbations to the relevant observables. In the present work, theoretical TT, DOA, and DOD sensitivity kernels are compared with parabolic equation simulations and tank experiment estimations.

4:15

2pAO13. Interferometry for three-dimensional acoustics in shallow water. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716), Boris G. Katsnelson (Voronezh State Univ., Voronezh, Russia 394006), and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Interferometry in optics has been known for decades. It requires identification of specific refracted rays via some reflecting front and existence of a mechanism for interference between rays to occur (i.e., constructive and destructive amplitude and phase information). In underwater acoustics, the principal phenomenon is the same. However, identification of the refracted (or reflected) rays requires a well defined geometry between the acoustic source and receiver path and the reflecting (or refracting) front. The interference between the direct and reflected wave fronts from sea surface has been shown for some time. Recent experimental observations have reported the identification of the interference in the horizontal plane [J. Acoust. Soc. Am. **129**(4), EL141 (2011)]. Theoretical description of the horizontal interference phenomenon as well as the follow up work for potential use of interferometry techniques for three dimensional acoustic wave propagation in shallow water is presented. Theoretical and experimental results are shown. [Work supported by ONR.]

4:30

2pAO14. Asymptotic behavior of vertical travel-time sensitivity kernels with increasing propagation range. E. K. Skarsoulis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. Hellas, 100 N. Plastira St., 70013 Heraklion, Crete, Greece), B. D. Cornuelle, and M. A. Dzieciuch (Univ. of California, San Diego, CA 92093-0230)

Vertical travel-time sensitivity kernels (VTSKs) describe the effect of horizontally uniform sound-speed changes on travel-times in range-independent environments. Wave-theoretic VTSKs can be obtained either analytically, through perturbation of the normal-mode representation, or numerically, as horizontal marginals of the corresponding 2D and 3D travel-time sensitivity kernels. In previous works it has been observed that, as the propagation range increases, wave-theoretic VTSKs approach the corresponding ray-theoretic sensitivity kernels even for low frequencies. In the present work an asymptotic expression of the wave-theoretic finite-frequency VTSKs is obtained, using a stationary-phase approach. Numerical results show that wave-theoretic VTSKs converge with increasing range toward the asymptotic form, which in turn lies very close to the ray-theoretic VTSK. [Work supported by ONR.]

4:45

2pAO15. Automatic detection of the number of raypaths in colored noise using short-length samples. Longyu Jiang and Jérôme Mars (GIPSA-Lab/DIS, Grenoble INP, 961 rue de la Houille Blanche BP 46F, 38402, Grenoble Cedex, Long-Yu.Jiang@gipsa-lab.grenoble-inp.fr)

In ocean acoustic tomography (OAT) (especially in shallow water where raypaths are mixed), knowledge of the number of raypath is crucial for inversion algorithm. In this paper, a noise-whitening exponential fitting test (NWEFT) is presented in this context for detecting the number of raypaths. Classically, two suggested approaches are the Akaike information criterion (AIC) and the minimum description length (MDL). Based on ideal assumption of ergodic Gaussian random processes and white Gaussian noise, MDL is shown to be asymptotically consistent, whereas the AIC tends to overestimate the order of model. However, these assumptions could not be fulfilled in practical case of OAT. In order to be adapted for real case of OAT, noise-whitening processing is applied as first step. Then, NWEFT bases on the fact that the profile of the ordered eigenvalues fits an exponential law for short-length samples of white Gaussian noise. The number of raypaths could be detected when a mismatch occurs between observed profile and exponential model. The fact that NWFET works on short-length samples is very important as a long duration of the received signal in OAT is unavailable. Its performance is studied with synthetic and real data set and compared with classical algorithms.

5:00

2pAO16. Krylov methods in inverse scattering and imaging. Paul E. Barbone, Gonzalo R. Feijóo (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), and Assad A. Oberai (Dept. of Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., 5048 JEC, 110 8th St., Troy, NY 12180)

Inverse scattering requires the measurement and inversion of a compact operator. The compactness of the operator implies that its range is low dimensional (i.e., sparse.) This implies the theoretical possibility of measuring the full operator with relatively few measurements and inverting it on a sparse basis. One issue, however, is that the basis on which the operator is sparse is unknown *a priori*. We show that Krylov methods can be used to simultaneously identify an efficient basis for the measurements and facilitate the inversion for imaging purposes. In particular, we show how imaging via Multiple SIgnal Classification (MUSIC) and by Krisch's factorization

method can be efficiently implemented in a Krylov space context. This method allows us to make most efficient use of *all* available acoustic sensors with few measurements and with minimal mutual interference.

5:15

2pAO17. Passive matched-field inversion using a horizontal planar array. Donald R. DelBalzo and James H. Leclere (QinetiQ North America, 40201 Hwy 190 East, Slidell, LA 70461)

Shallow-water acoustic predictions are severely limited by uncertainty in sediment property characteristics. Inverse methods with controlled active sources and vertical arrays have been used to estimate seabed properties; however, some applications require a covert approach and horizontal bottomed arrays. This study addresses the accuracy of low-frequency (100–200 Hz) matched-field correlations using broadband signals from surface ships with unknown source levels at unknown ranges. Matched-field techniques are applied in a realistic shallow-water environment with a horizontal planar array and high signal-to-noise ratios. The simulations indicate significant potential for accurate estimates of thick-sediment characterizations of grain size out to ranges of tens of water depths in shallow water, despite moderate mismatch conditions in the environmental model. The results show that: (1) the horizontal aperture should contain at least three hydrophones per wavelength to ensure high quality inversions; (2) the horizontal aperture should be several times longer than a vertical aperture; (3) coherent (phase-only) matched-field processing outperforms standard intensity processing by about 2 dB in good input SNR conditions; (4) incorrect assumptions about the assumed sound-speed profile (e.g., incorrect mixed-layer-depth) do not significantly affect the inversion results. [Work sponsored by QinetiQ North America.]

TUESDAY AFTERNOON, 1 NOVEMBER 2011 — PACIFIC SALON 6/7, 1:00 TO 5:00 P.M.

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics

David A. Brown, Cochair
Electroacoustic Research Lab., Univ. of Massachusetts, Dartmouth, 151 Martine St., Falls River, MA 02723

Sairajan Sarangapani, Cochair
Ocean Engineering, Univ. of Rhode Island, 217 Sheets Bldg., Narragansett, RI 02882

Contributed Papers

1:00

2pEA1. Scattering reduction of an acoustically hard cylinder covered with layered pentamode metamaterials. Jeffrey E. Boisvert (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841), Clyde L. Scandrett (Naval Postgrad. School, Monterey, CA 93943), and Thomas R. Howarth (NAVSEA Div. Newport, Newport, RI 02841)

Unlike traditional target strength reduction coatings that rely on energy dissipation or other mechanisms to mitigate reflection, a coating comprised of metamaterials would behave as an acoustic waveguide that diverts sound energy around the object, thus reducing scattered energy. The majority of the literature has featured theoretical ideal metamaterial designs that have unrealizable properties, i.e., infinite mass, vanishing bulk modulus. However, our analysis has suggested that it may be possible to obtain effective scattering reductions with realizable material properties in a layered configuration using metafluids. In this context, realizable implies material properties that are constrained to lie within reasonable ranges relative to the density and bulk modulus of water. The multistatic scattering reduction of an acoustically hard cylinder covered with layered metafluids for plane wave incidence is analyzed. A range of coatings are considered, from those comprised of fluid layers that are isotropic in bulk moduli with anisotropic density (inertia) to those having anisotropic bulk moduli and isotropic density (pentamode). [Work supported by NAVSEA Division Newport ILIR.]

1:15

2pEA2. An acoustic directional antenna with isotropic materials. Christopher N. Layman, Jr., Theodore P. Martin, and Gregory J. Orris (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375-5320)

New acoustic metamaterial devices offer promising applications, ranging from tunable sound blocking with superior efficiency to acoustical diodes. Transformation acoustics (TA), relying on the invariance of field equations under coordinate transformations, in conjunction with metamaterial features, has further expanded the range of functionalized acoustic materials. However, for a large variety of devices based on TA, such as ones relying on Pendry's concept, physical realization remains limited owing to the requirement of anisotropic effective properties. Here we examine the behavior of a directional four-wave acoustic antenna, designed from finite embedded coordinate transformations (FECT), which eliminate the constraint of anisotropy by way of a suitable conformal mapping. The two dimensional antenna consist of a square rod with an inhomogeneous and isotropic distributed mass density and bulk modulus designed from the FECT perspective. The rod is imbedded in an acoustically matched matrix and subsequently subjected to an internal coaxially line source to evaluate its transmitting performance across a large bandwidth. Passive characteristics are also studied by probing the antenna with a point source-receiver setup. Experimental data are compared to both established analytical FECT models and full-wave simulations. [Work is supported by the Office of Naval Research.]

1:30

2pEA3. Compact directional acoustic sensing using multi-fiber optical probes. Joseph Bucaro (Excet, Inc., 4555 Overlook Ave., Code 7130, Washington, DC 20375-5350, joseph.bucaro.ctr@nrl.navy.mil), Nicholas Lagakos (Sotera Defense Solutions, Crofton, MD 21114), Brian Houston, Saikat Dey, and Maxim Zalalutdinov (Naval Res. Lab., Washington, DC 20375-5350)

A compact directional acoustic sensor concept is described, which uses an multi-optical fiber probe, a light emitting diode source, a photo-diode detector, and a short, slender cylindrical cantilever to the end of which is attached an optical reflector. A portion of the light exiting one fiber is collected by a second fiber after reflection from the mirror. Acoustically induced transverse displacement of the cantilever tip modulates the light collected by the second fiber, which then conveys the light to a photo-detector. Directional sensitivity is achieved by virtue of the dependence of the

collected light on the cosine of the angle between the line connecting the probe fiber centers and the direction of displacement of the cantilever tip (the acoustic wave direction). An analytic model of the acoustic response of the cantilever tip is constructed, which is partially verified using a finite element-based model and experimentally validated using measurements of the acoustic response in air. The model is used to predict its acoustic response versus frequency, how that response depends upon damping near the cantilever resonance frequency, and to what extent and over what frequency band that response depends upon the acoustically generated flow force. [Work supported by ONR.]

1:45

2pEA4. Synchronized vibrations measurements at multiple locations using a single continuously scanning laser doppler vibrometer. Applications to non-contact sensing of human body vibrations. Muhammad Salman and Karim G. Sabra (School of Mech. Eng. Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332)

Laser Doppler vibrometry (LDV) is a non-contact method to measure surface velocity. Typical LDV configurations uses one fixed laser (He-Ne) beam at a specific point and orientation on a surface under test. A continuously scanning laser Doppler vibrometry (CSLDV) technique using a single laser beam performing quickly a long scan was developed to measure surface velocity at each laser position during its scan. This fast CSLDV can then replace several fixed LDVs. This technique is especially advantageous for sensing human body natural vibrations (typically below 100 Hz) at multiple locations (e.g., along small muscles) as it does not require traditional skin-mounted sensors (e.g., accelerometers array), which eliminates mass artifacts and the set-up time to attach those sensors. Experimental measurements were conducted using a CSLDV with a 200 Hz linear scan rate, over scan lengths up to 5 cm, to measure low-frequency vibrations ($f<100$ Hz) on gel samples which mimic human soft tissues. Validations of the CSLDV measurements were done using an array of several fixed LDVs distributed along the same scan line. The effects of speckle noise on CSLDV measurements will be quantified. Applications of this CSLDV technique for active and passive elastography measurements will be presented.

2:00

2pEA5. Response surface optimization of a directional endfire microphone array for hearing aids. Thomas Burns (Starkey Labs, Inc., 6600 Washington Ave. S., Eden Prairie, MN 55344, tburns@starkey.com)

The optimal operating parameters for a directional microphone array worn *in situ* are not necessarily equivalent to the optimal parameters while operating in the absence of head and torso related scattering. These parameters include the relative magnitude and phase of the microphones and their positional placement on the head, characterized as factors, operating over a range of levels, characterized by their production spread and susceptibility to drift. The goal is to understand how these factors operating over their levels contribute to the *in-situ* directional responses on a measurement manikin, characterized by the directivity index and the unidirectional index. Using 614 impulse responses acquired in ten deg resolution on the manikin, a simple central composite design of experiments was conducted to fit a quadratic polynomial and generate a response surface to the aforementioned directional indices, thereby leading to the critical first-order and two factor interactions of the system. The interactions, statistical validity of the predictive polynomials, and the sweet spot of operation will be described for some common hearing aid microphone arrays.

2:15

2pEA6. Optimization of tuning and matching of broadband transducers with power switching amplifiers. Corey Bachand, Boris Aronov, and David A. Brown (BTech Acoust. LLC, ATMC, UMass Dartmouth, 151 Martine St., Fall River, MA 02723, corey.bachand@cox.net)

Underwater transducers for broadband communication rely on effective tuning and matching to a power amplifier for maximum signal bandwidth and efficiency. This analysis follows a systematic approach to design an efficient and effective broadband acoustic transmit system. Power switching (class D) amplifiers use a variety of modulation schemes to reduce the losses incurred at the high power amplification stage. Lowpass filtering at the output stage of the switching amplifier is often employed to attenuate the high frequency carrier signal from the modulation stage. A matching transformer steps up the voltage delivered to the transducer. The tuning network can be designed to provide optimum cancellation of reactance over a wide band, thus improving the power factor bandwidth.

2:30

2pEA7. Development of an optical transducer for an electro-acoustic guitar. S. K. Cho and Y. W. Park (Dept. of Mechatronics Eng., Chungnam Natl. Univ., 99 Daehangno, Yuseong-gu, Daejeon 305-764, Korea, chosk79@nate.com)

This paper presents the development of an optical transducer (OT) for an electro-acoustic guitar. Two conceptual designs are proposed: one with one infrared light emitting diode (IR LED) and one photodetector, and the other with two IR LEDs and one photodetector. Both concepts are based on the top-to-bottom structure: IR LED is on the top, and the photodetector is at the bottom. After the preliminary tests, the latter design is selected as the proposed OT. The OT is fabricated on the PCB with proper electronic circuit, and mounted on the guitar. The developed OT is subjected to the performance evaluation with a dedicated measuring device. The performance of the OT is compared with commonly used piezoelectric transducer. Findings are summarized: (1) The output signals from the OT are much higher that those from the piezoelectric transducer in both average and peak-to-peak voltages. (2) The noise level from the OT is similar or less than that from the piezoelectric transducer. (3) SNR with the OT is increased by 45% in average, compared with the piezoelectric transducer.

2:45

2pEA8. Calculating piezoelectric parameters of stripe-electroded cylinders and bars with continuous no-uniform electric fields. Sairajan Sarangapani, David A. Brown (Acoust. Res. Lab., Adv. Technol. and Manufacturing Ctr. and Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth,151 Martine St., Fall River, MA 02723, ssarangapani@umassd.edu), and Boris Aronov (BTech Acoust., LLC, Fall River, MA 02723)

Tangentially poled thin-walled stripe-electroded piezoelectric bars and hollow cylinders are used in several electromechanical and electroacoustic applications. The 33 mode properties of the stripe-electroded bars and hollow cylinders are not fully realized due to the non-uniform polarization and non-uniform operational electric field due to fringing from the surface electrode geometry. A numerical finite difference method (FDM) was used to analyze the nonuniform electric field in the bar but under the assumption that the piezoelectric element is fully polarized. The effective electromechanical coupling coefficient $k_{3'3}$, the effective piezoelectric modulus $d_{3'3}$, the effective compliance $s^{E}_{3'3}$, and the effective relative dielectric constant $\epsilon_{3'3}$ (where the prime indicates field nonuniformity) are calculated using the energy method by accounting for the effects of nonuniform operational electric field and non uniform strain distributions. Analytical and experimental results are in good agreement and design optimizations are presented.

3:00–3:15 Break

3:15

2pEA9. Modeling piezoelectric parameters of bending mode transducers. Sairajan Sarangapani, David A. Brown (Acoust. Res. Lab., Adv. Technol. and Manufacturing Ctr. and Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723 ssarangapani@umassd.edu), and Boris Aronov (BTech Acoust., LLC, Fall River, MA 02723)

The piezoelectric bender bar vibrator is commonly used for generating low frequency flexural plate mode vibrations. This study considers the excitation of “benders” using stripe-electroded piezoelectric elements of various electrode patterns and poling configurations using the $k_{3'3}$ and $k_{3'1}$, where the prime indicates field nonuniformity. A numerical analysis is used to calculate the nonuniform electric field lines and the corresponding coupling coefficients as well as other related electromechanical parameters by developing a Lagrangian description of the electromechanical body and using the energy method. The effective electromechanical coupling coefficients are calculated by taking into account the internal energies due to transverse,

longitudinal, and shear vibrations in the bar. The contribution of the passive and the active capacitance is also explained and taken into account while calculating the electromechanical parameters.

3:30

2pEA10. Limits of dissipative coefficients in piezoelectric transverse isotropic materials. Gordon E. Martin (3675 Syracuse Ave., San Diego, CA 92122, gemartin@ieee.org)

This paper relates to limiting forms of complex coefficients in passive piezoelectric systems due to hysteretic dissipation The specific application is polarized ferroelectric ceramic materials. The research required three features. (1) Mathematical models require the theory of physically realizable networks with distributed components like electrical transmission lines. Such theory is well established for discrete components and special forms of distributed components. Theory for the general case is reported here. (2) Passive systems cannot create energy so complex coefficients of mathematical models must have limiting values for both real and imaginary coefficients of the constitutive matrices. (3) The mathematical model can be expressed in complex symmetric form. It is proved all imaginary parts of matrix coefficients have changes of first order due to small dissipative perturbations. For piezoelectric spectroscopy purposes, measured immittances have two kinds of limits due to the addition of small dissipative effects. They cause small changes in complex values and corresponding frequencies of zeros and poles. It is proved (a) all critical immittance values have small changes to first order and (b) all corresponding frequencies have small shifts to second order.

3:45

2pEA11. Evaluating piezocrystal and piezoceramic transducer bandwidth and effectiveness. Corey Bachand, David A. Brown, and Boris Aronov (BTech Acoust. LLC, ATMC, UMass Dartmouth, 151 Martine St., Fall River, MA 02723, corey.bachand@cox.net)

Conventional piezoceramic transducers offer moderate bandwidth and performance to serve the majority of underwater acoustic applications. The manufacturability of piezoceramic elements in a variety of shapes (bars, cylinders, and hemispheres) makes them a cost-effective solution in may transducer designs. However, the emergence of piezocrystals in transducer designs has significantly increased the usable bandwidth while reducing the device footprint. This is enabling in terms of size and weight for use on mobile platforms (UUVs), especially when considering that one piezocrystal transducer may replace several piezoceramic transducers and reduce the number of hardware (power amplifier) channels. There are still fabrication and operational challenges with piezocrystal transducers that need to be overcome before they are widely adopted in the underwater community.

4:00

2pEA12. Design for a modular and scalable sonar source using displacement amplifying lamina. Richard H. Lyon (60 Prentiss Ln., Belmont, MA 02478-2021, rhlyon@lyoncorp.com)

A sonar sound source is described that is capable of radiating increased sound power at low frequencies in a non-resonant mode of operation. Non-resonant operation is used so that the amplitude and phase of the generated signals are smooth over a range of frequencies. The improvement in output is achieved in part by the use of Galfenol, a fairly new magnetostrictive (MS) material with a high MS strain coefficient. The enhanced output is also due to the use of non-resonant amplifying volume displacing lamina. The system is modeled using a simplified equivalent circuit linear model that allows prediction of several quantities of interest, such as the radiated output, the strain in the MS material, the excitation power, and the sensitivity to ambient pressure fluctuations. [Work supported in part by the US Air Force and the US Navy.]

4:15

2pEA13. Comparison of three experimental methods for assessing the blocked electrical impedance of a moving-coil loudspeaker driver. Daniel R. Marquez, Timothy W. Leishman, and Rex P. Price (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N-283 Eyring Sci. Ctr., Provo, UT 84602, danmarquez7@yahoo.com)

In this paper, three methods are discussed for experimentally isolating the blocked electrical impedance of a loudspeaker driver. The first involves measurement of the voice-coil impedance in a vacuum, before and after an added mass is applied to its cone. The second involves the use of a scanning laser Doppler vibrometer in conjunction with frequency-dependent electrical measurements at the driver terminals. The third involves the traditional destructive method of potting the driver in a hard-drying compound to allow direct measurement of the blocked impedance. The advantages and disadvantages of each method are discussed. The impedances determined by the three methods are used to predict the frequency-dependent cone velocities of several drivers while under operation. Actual measured velocities are compared with the predictions to substantiate the accuracy of each method.

4:30

2pEA14. Comparison and verification of analogous circuit models for dynamic moving-coil transducers. Rex P. Price, Daniel R. Marquez, and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., ESC N-283, Provo, UT 84602, rexprice@yahoo.com)

For decades, analogous circuits have been used to model the electro-mechano-acoustical properties of moving-coil transducers. In recent years, many enhanced models have been proposed to improve the accuracy of their estimated voice-coil impedances. In this work, an iterative complex curve-fitting routine has been used to best fit and compare the various models to the measured complex input impedance data of several loudspeaker drivers in free air. The estimated blocked electrical impedance of each driver was extracted and used to predict its frequency-dependent cone velocity while under operation. Actual driver cone velocities were measured experimentally using a scanning laser Doppler vibrometer, and the predictions were compared to further substantiate the accuracy of each model. The results highlight those models with the greatest predictive capabilities and reliability.

4:45

2pEA15. Modeling and validation of magnetostrictive sound transducer including flat panel. H. J. Park and Y.W. Park (Dept. of Mechatronics Eng., Chungnam Natl. Univ., 99 Daehangno, Yuseong-gu, Daejeon 305-764, Korea, imal_hjpark@cnu.ac.kr)

This paper contains models of the magnetostrictive actuator and flat panel for the investigation of interaction between actuator and flat panel. (1) A transfer function of the magnetostrictive actuator between a displacement U_a and input current I: $G_a(s) = U_a(s)/I(s) = nd/(1 + c/c_o + sr/c_o + s^2 m/c_o)$, where n is number of coil turns, d is magnetostrictive constant, c is stiffness of prestress spring, c_o is open circuit stiffness, r is damping coefficient, and m is effective mass. (2) Transfer functions of a flat panel: $G_f(s) = U_f(s)/U_a(s) = -A/(-B + C + D)$, $A = m_1 k_1 s^2/(m_1 s^2 + c_1 s + k)$, $B = m_2 s^2 + (c_1 + c_2)s + (k_1 + k_2)$, $C = (c_1 s + k_1)^2/(m_1 s^2 + c_1 s + k_1)$, $D = (c_2 s + k_2)^2/(m_3 s^2 + (c_2 + c_3)s + k_2 + k_3)$, where m, c, and k are parameters of the actuator and flat panel models, and parameters are determined experimentally. The final transfer function of actuator and flat panel is expressed by multiplying $G_a(s)$ and $G_f(s)$. Simulations are performed through commercial program under the conditions applying a white noise to the final transfer function. The simulated and experimental frequency responses are compared.

2p TUE. PM

Session 2pEDa

Education in Acoustics and Physical Acoustics: Tools for Teaching Advanced Acoustics II

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

Scott D. Sommerfeldt, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N181 Eyring Science Center, Provo, UT 84602

Invited Papers

1:00

2pEDa1. Systematic circuit model construction for complex interconnected acoustic systems. Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292)

Complex, highly interconnected acoustic systems can be difficult to model for students and inexperienced practitioners. The systematic lumped-element circuit model construction method presented here is easy to learn and teach and allows for quick and easy, error-free circuit model construction. The present author discovered the method in a book by Mario Rossi [*Acoustics and Electroacoustics*, Artech House Publishers (1988)] and has included it in the electroacoustic transducers course taught at The University of Texas at Austin since 2003.

1:20

2pEDa2. Fourier: Making Waves—An interactive simulation for visualizing fourier analysis. Wendy K. Adams (Dept. of Phys., Univ. of Northern Colorado, CB 127, Greeley, CO 80639)

In this presentation, the PhET simulation "Fourier: Making Waves" http://PhET.colorado.edu/en/simulation/fourier, will be presented including the research behind the simulation, how students react to it and ideas for use in class. Students typically learn the math needed to do Fourier transforms and learn how to express a function in time or space and in terms of wavelength, wave number, or mode. However, many of these relationships are only memorized for the short term (exam) and are not retained. This simulation is designed to help students visualize how a combination of simple sines and cosines can create a more complicated function and listen to the sounds produced by each harmonic. This simulation features 11 adjustable harmonics which can be used to demonstrate various auditory perceptions. There is also a game tab with ten different levels that challenges students to choose the correct harmonics to match more and more complicated functions. For more mathematical explorations, students can investigate each of the symbols lambda, T, k, omega, and n to learn what each represents on the graph and their relationships with one another. Finally there is a tab to help students visualize moving from a discrete to a continuous series.

1:40

2pEDa3. Understanding sound wave propagation using computer animations. Jorge P. Arenas (Inst. of Acoust., Univ. Austral of Chile, P.O. Box 567, Valdivia, Chile)

It is well known that sound waves are often difficult, if not impossible, to visualize which makes their nature and effects much more difficult to explain than those of other kinds of waves. In this article, a visualization tool for enhancing the students' learning process for a fundamental of acoustics course is reported. The visualization is done through particle displacement computer animations of different sound propagation cases using a simple MATLAB script. Several examples that can enhance the material discussed during class time are presented, in particular those topics involving diffraction of sound waves. It is observed that visual displays used during lectures improve the students' retention of new material. Students seem to make better association between wave motion and particles in a medium. A visual advantage of the particle displacement animations is that they use tangible items to represent the invisible process, by visualizing the invisible particles of air as dots on the computer screen. In addition, this instructional tool used to visualize sound fields enhances the understanding of many conceptual aspects underlying sound wave motion and can be used to motivate a discussion of the wave equation later.

2:00

2pEDa4. A student-friendly algorithm for planar mode propagation in arbitrary transmission lines. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338, jerry.ginsberg@me.gatech.edu)

Many textbooks treat the propagation of a harmonic plane wave in an acoustic transmission line. Their scope generally is limited to a small number of interconnected uniform cross section branches, and they emphasize fitting the propagation properties to continuity conditions at junctions. In contrast, the network formalisms advocated for intricate engineered systems, such as that offered fairly

recently by Panigrahi and Munjal [J. Acoust. Soc. Am. **118**, pp. 2860–2868], make them unsuitable for a first graduate-level course in acoustics. The present work offers an algorithmic approach that is simple to formulate, yet more efficient than available alternatives, and capable of treating arbitrary networks. The steps required to implement the algorithm are sequential numbering of the branch nodes and of the junctions, definition of a connectivity matrix that indicates which nodes are common to a junction, and statement of the continuity and termination conditions in terms of the port pressures and particle velocities. Beyond that the algorithm operates automatically. The result is a set of simultaneous equations for the complex pressure amplitudes at the junctions. In contrast, the approach offered by Panigrahi and Munjal derives simultaneous equations for the nodal pressures and port velocities, whose number is far greater than the number of junctions.

Contributed Papers

2:20

2pEDa5. Acousto-mechanical modeling of an Edison tinfoil phonograph. Jason D. Sagers, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758), and Preston S. Wilson (The Univ. of Texas at Austin, Austin, TX 78712-0292)

A homemade reproduction of an Edison tinfoil phonograph was demonstrated at the 161st ASA meeting in Seattle, WA [JASA **129**, 2581 (2011)]. While past work focused on the history and development of the device, the present work is focused on analyzing the acousto mechanical behavior of the homemade device. A dynamical model is presented and is used to predict the frequency dependent vibration of the phonograph diaphragm due to an acoustic input. The model predictions are compared with experimental measurements of the diaphragm vibration and potential solutions for optimizing the device are discussed.

2:35

2pEDa6. Demonstration of coupled membrane modes on a musical drum. Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Musical drums such as tom toms and snare drums typically consist of two circular membranes attached to a cylindrical shell. Due to the enclosed air and the shell itself, a drum with two heads exhibits coupling of the lower frequency membrane modes, while the higher frequency modes of the two heads remain essentially independent. A simple demonstration has been developed that illustrates several aspects of drumhead vibrations, including the distinction between strongly and weakly coupled modes. Methods for determining the relative phases and amplitudes of the coupled oscillations have also been developed. The demonstration is useful at a variety of pedagogical levels and can be supplemented with more advanced experiments, including the use of electronic speckle pattern interferometry to identify the mode shapes on both membranes.

TUESDAY AFTERNOON, 1 NOVEMBER 2011 TOWNE/ESQUIRE, 2:55 TO 4:00 P.M.

Session 2pEDb

Education in Acoustics: Take 5's

Andrew Morrrison, Chair
Physics Dept., DePaul Univ., 2219 Kenmore Dr., Byrne Hall, Chicago, IL 60614

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign up for non-consecutive slots.

Session 2pMUa

Musical Acoustics: Music Perception and Cognition

Diana Deutsch, Chair
Dept. of Psychology, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

2:00

2pMUa1. Why does musical training benefit the neural encoding of speech? A new hypothesis. Aniruddh D. Patel (The Neurosci. Inst., 10640 John Jay Hopkins Dr., San Diego, CA 92121)

Mounting evidence suggests that musical training benefits the neural encoding of speech. This paper offers a hypothesis specifying why such benefits occur. The OPERA hypothesis proposes that such benefits are driven by adaptive plasticity in speech-processing networks, and that this plasticity occurs when five conditions are met. These are (1) **O**verlap: there is anatomical overlap in brain networks that process an acoustic feature used in both music and speech (e.g., waveform periodicity, amplitude envelope), (2) **P**recision: music places higher demands on these shared networks than does speech, in terms of the precision of processing, (3) **E**motion, (4) **R**epetition, and (5) **A**ttention: the musical activities that engage this network elicit strong positive emotion, are frequently repeated, and are associated with focused attention. According to the "opera" hypothesis, when these conditions are met neural plasticity drives the networks in question to function with higher precision than needed for ordinary speech communication. Yet since speech shares these networks with music, speech processing benefits. The OPERA hypothesis is used to account for the observed superior subcortical encoding of speech in musically trained individuals and to suggest mechanisms by which musical training might improve linguistic reading abilities.

2:15

2pMUa2. The octave illusion revisited: Performance measurements for handedness categorization. Michael Oehler (Musicological Inst., Univ. of Cologne, Cologne, 50674 Germany), Christoph Reuter, Harald Schandara, and Michael Kecht (Univ. of Vienna, Vienna, Austria)

An extended replication study of the octave illusion (Deutsch 1974, 1983) is presented. Since the first description of the octave illusion in 1974, several studies showed that the perception of the two-tone patterns depends on subjects' handedness. Partially almost 90% of the right-handed subjects reported to hear the high tone of the octave at the right ear. In all related studies the handedness categorization was done by means of a questionnaire, e.g., the handedness inventory of Varney and Benton (1975). Several current studies (e.g., Kopiez, Galley, Lehmann, 2010), however, showed that objective non-right-handed persons cannot be identified by handedness inventories. In concordance with Annett's "right shift theory" (2002), performance measurements as speed tapping seem to be a much more reliable handedness predictor. Therefore in the replication study (N=158) Varney and Benton's inventory as well as a speed tapping task were used to categorize left- and right-handed subjects. The results of Deutsch's study could be replicated when using the same handedness inventory. The performance measurement task, however, led to a significantly clearer distinction between the left- and right-handed subjects (w=0.39 in contrast to w=0.26 in the replication) and more structured perception patterns could be observed within the left-handed group.

2:30

2pMUa3. Large-scale direct-test study reveals unexpected characteristics of absolute pitch. Diana Deutsch (Dept. of Psychol., Univ. of California, San Diego, La Jolla, CA 92093), Jinghong Le (East China Normal Univ., Shanghai 200062, China), Jing Shen (Univ. of California, San Diego, La Jolla, CA 92093), and Xiaonuo Li (Shanghai Conservatory of Music, 20 Feng Yang Rd., Shanghai 200031, China)

Absolute pitch, the ability to name a musical note in the absence of a reference note, is very rare in North America and Europe, so that attempts to characterize its features in the western world have involved small numbers of subjects, informal self-report, questionnaires, or web-based exploration. The study reported here capitalized on the high prevalence of absolute pitch in China to explore its features in detail using direct, on-site testing of 160 subjects in a Chinese music conservatory. As expected, performance levels were extremely high, and there was a large effect of age of onset of musical training, with those who began training by age 5 scoring on average 83% correct not allowing for semitone errors and 90% correct allowing for semitone errors. It was found that errors tended to be on the sharp side. An advantage to white keys over black keys was also found; however this was not due to early experience with the piano, as had been hypothesized by others, since performers on different instruments showed an effect that was as large or larger. Furthermore, the special status for note A that had been hypothesized by others was not found, even for orchestral performers.

2:45

2pMUa4. Songs, cell phones, absolute pitch: Long-term pitch memory for familiar stimuli. Kevin Dooley (Dept. of Psychology, Univ. of California, San Diego, La Jolla, CA 92093)

Absolute pitch (AP) is a rare phenomenon as formally defined, but long-term pitch memory appears much more common when tests involve familiar musical material and do not require the use of formally learned pitch labels. It is unclear whether AP possession confers additional advantages to long-term pitch memory in such tasks or merely combines a rare ability to form pitch-label associations with a more general capacity for pitch memory. To test this, 36 trained musicians—18 AP possessors and 18 non-possessors with equivalent age of onset and duration of musical training—were asked to recall and vocalize a familiar song, and their responses were compared with the pitches of the actual recordings; this was repeated with their cell phone ringtones. Both groups were significantly more accurate than chance on the song task, but only the AP possessors performed above chance on the ringtone task. The findings confirm the existence of widespread long-term pitch memory but also point to an AP advantage under some circumstances.

Session 2pMUb

Musical Acoustics: Music and Auditory Space

Diana Deutsch, Chair
Dept. of Psychology, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

3:30

2pMUb1. The Instrument & the Room: A study of the grand piano focused on the needs of audio education. Brett Leonard, Grzegorz Sikora, and Martha de Francisco (Graduate Program in Sound Recording, Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke St., West Montreal, QC H3A 1E3, brett.leonard@mcgill.ca)

Through their training and education, aspiring recording engineers often encounter literature on the radiation patterns of typical musical instruments. The study of this information can greatly inform the placement of microphones and facilitate one's learning about the way acoustic instruments work. These musical acoustics studies, however, typically employ an anechoic or near-anechoic environment to minimize reflections from interfering with the instrument under test. Since the recording engineer works almost exclusively in environments with reflective surfaces, this causes a disconnect and can inhibit a full understanding of the relationship between the instrument and its environment. A case study of the acoustic grand piano is presented in which the instrument and the non-anechoic room are presented as a single, coupled acoustic system. Over 1300 measurement points are used to characterize the instrument/room combination. The study is conducted in both a small recording space and a large scoring stage, yielding non-room specific results that show areas of high frequency energy that are not present in typical anechoic measurements. Exploration of these differences and potential causes are presented.

3:45

2pMUb2. Methods for automating multichannel directivity measurements of musical instruments in an anechoic chamber. Nicholas J. Eyring II, Timothy W. Leishman, Kristina M. Sorensen, and Nathan G. W. Eyring (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, eyringj@gmail.com)

Comprehensive directivity measurements of musical instruments in anechoic environments involve several experimental challenges. However, by adapting methods currently used to characterize loudspeaker directivity (e.g., through high-resolution balloon plots), one can obtain highly detailed and instructive directivity data. This may be accomplished by rotating a musical instrument with sequential azimuthal angle increments under a fixed semicircular array of microphones while recording repeated notes or sequences of notes. The result is a computer-controlled acquisition of hundreds or even thousands of sound pressure measurements over a measurement sphere. The directivity data and corresponding balloon plots may be shown to vary as functions of time or frequency. This paper explores the approach applied to a grand piano with velocity controlled keys played through MIDI communication. Instruments played by live musicians may also be evaluated, although the process requires carefully developed techniques of control, feedback, and compensation to achieve acceptable results. These and other considerations of performing automated, multichannel directivity measurements of musical instruments are detailed in this presentation.

4:00

2pMUb3. Feature space minimization and its affect on head related transfer functions clustering. Areti Andreopoulou and Agnieszka Roginska (Dept. of Music and Perf. Arts Professions, New York Univ., 35 W. 4th St., Ste. 777, New York, NY 10012, aa1510@nyu.edu)

Several approaches have been taken toward data/feature-space reduction in HRIRs, operating either in the time (original, minimum phase, normalized HRIRs) or in the frequency domain (magnitude, log-magnitude, standardized log magnitude HRTFs). Shin and Park (2008) extracted only the response of the pinna (0.23 ms of the original HRIR), Hwang and Park (2007) included also the response of the head and torso (1.5 ms), while Bondu (2006) operated on the first 100 samples of the impulse responses. Other research focuses on employing PCA for minimizing the feature space to 5–12 orthogonal components and their corresponding weights (Langendijk, 2002; Huang, 2009; Hugeng, 2010), while others have managed to isolate directional and non-directional spectral cues of non-individualized HRTFs (Hu, 2008; Diepold, 2010). In previous work, the clustering tendencies of the standardized log-magnitude HRTFs of 110 subjects on the horizontal plane have been demonstrated, by applying k-means on 256-feature filters. In this study, those results are compared to the clustering behavior of data reduced filters by applying the previously mentioned techniques. The extent to which the original clustering tendencies are maintained is used as an evaluation criterion of the appropriateness of each data-reduction technique.

Session 2pNS

Noise: General Topics–Outdoor Noise

Dickson Hingson, Chair
275 S. River Run, #3, Flagstaff, AZ 86001

Chair's Introduction—1:00

Contributed Papers

1:05

2pNS1. Effects of the soil on the noise attenuation of environmental berm barriers. Jorge P. Arenas (Inst. of Acoust., Univ. Austral of Chile, P.O. Box 567, Valdivia, Chile), Jesus Alba, and Romina del Rey (Universidad Politécnica de Valencia, Campus de Gandía, 46730 Valencia, Spain)

Berm mounds are a commonly used technique to reduce the environmental noise levels produced by highways. A berm is a natural noise barrier constructed of soil, stone, rock, or rubble, often landscaped, running along a highway to protect adjacent communities from noise pollution. An earth mound may be constructed using surplus materials at project site, provided there is sufficient land area available for its construction. Therefore, berms are natural environmental barriers, having relative low costs and they are subjectively well perceived by residents. However, exact noise attenuation provided by berms has not been enough explored in the technical literature, as opposed to common barriers made of vertical rigid walls. Although, some highway noise prediction models assign a noise reduction bonus of 3 dB(A) to sound barriers made of earth mounds, experimental assessments have yielded mixed results. Few theoretical reports have studied this particular problem. In this work, numerical analysis using the classical theory of diffraction is performed on a berm made of different types of soil. The model assumes a line source and includes flow resistant data as boundary conditions. By integrating the results, noise attenuation is given in third-octave bands. It is concluded that soil's properties significantly influence the measured results and that this may be one of the causes of varied in-site empirical evidence.

1:20

2pNS2. Grand Canyon National Park Overflights environmental impact statement (EIS): Backcountry impairment under the National Park Services' (NPS) noise standards. Dickson J Hingson (Natl. Parks and Monuments Comm., 275 S River Run, Flagstaff, AZ 86001, dhingson@infowest.com)

The longstanding Grand Canyon overflights noise pollution saga approaches a decision—public comment on the DEIS having concluded in June. Current, longstanding park impairment of soundscape and wilderness character, throughout the park's popular East end backcountry, is readily apparent from detailed quantitative "Location Point" analysis. Per cent time audible and sound level for these, by alternative, and season was displayed with an elegant technique, presented 2009 to INCE/ASA, by Nick Miller. Although the "NPS preferred" draft alternative did not represent significant improvement, the quieter "seasonal use," Alternative "E" fared much better. This derives from seasonal closures, each year, alternating between the two currently used air tour loops. "E's" data indicate that more stringent daily limits on tour flight allocations will be required to avoid major adverse noise impacts continuance in park backcountry, even so. The core business of the park service being to prevent impairment in its wilderness backcountry, the stark findings afford the NPS a clear opportunity to adapt, with a more appropriate implementation alternative. This will be due by spring, 2012, to timely render a record of decision. An recent agreement between NPS and FAA as to relative roles was helpful in clarifying the applicable noise standards.

1:35

2pNS3. Propagation in a realistic outdoor environment. Whitney L. Coyle, Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., University Park, PA 16802, wlc5061@psu.edu), Bruce Ikelheimer, Micah Downing, Michael James, Kevin Bradley, and Josh Mellon (Blue Ridge Res. and Consulting, 13 1/2 W. Walnut St., Asheville, NC 28801)

A complementary experimental and computational study was conducted to assess variability in realistic outdoor sound propagation environments. Field measurements were conducted in a valley of the Smokey Mountains located in western North Carolina. This location exhibited complex terrain, vegetation, and weather conditions. Continuous and impulsive sources were positioned in multiple, reciprocal locations for the experiment. Receiver locations were spread throughout the area with many significant terrain features. Simultaneous atmospheric measurements were made to measure wind speed and temperature profiles and a level of turbulence. A Green's function parabolic equation model was written and used with matching conditions for comparison. This presentation will give a brief overview of the project and provide preliminary results. [Work supported by Spawar Systems Center Pacific.]

1:50

2pNS4. Model-data comparison for acoustic propagation over water. Sean P. Pecknold, Cristina Tollefsen, and Emma Murowinski (Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, Nora Scotia, B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca)

Modeling the propagation of sound over water is an important tool for determining the possible effects of noise sources on the environment, such as naval gunfire exercises on bird nesting grounds, or offshore wind turbine disturbance. Temperature, humidity, and wind speed and direction all play an important role in determining acoustic propagation over water. Here, a sound source is mounted on a boat moving up to 2.5 km toward and away from a receiver on another vessel, over the span of several days. The propagation loss measured as a function of range is compared to modeled results based on measured temperature, humidity, wind velocity, and surface roughness, using atmospheric turbulence models to improve prediction capability.

2:05

2pNS5. Application of the equivalent source method to outdoor directional sound sources. Sergey N. Vecherin, D. Keith Wilson (U. S. Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Sergey.N.Vecherin@usace.army.mil), and Vladimir E. Ostashev (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309)

Many outdoor sound sources, such as aircraft and moving ground vehicles, exhibit directional sound radiation patterns that can be

measured in the far field. However, this information is insufficient for specifying a source function that can be used in propagation algorithms. The equivalent source method (ESM) allows one to reconstruct an equivalent distribution of point sources having a given far-field radiation pattern. In this research, the application of the ESM to spatially complex radiation patterns, similar to those of actual helicopters, is studied in detail. Two algorithms for the source reconstruction are developed for arbitrarily complex radiation patterns. The first algorithm reconstructs three-dimensional source distributions that may not, however, be convenient for initializing calculations with parabolic equations. The second algorithm is designed for the two-dimensional parabolic equations and reconstructs strictly vertical source distributions having a given radiation pattern over a limited range of elevation angles. Some practical aspects of the measured data, such as outliers, data incompleteness, and phase loss in sound level measurements are also studied and recommendations are provided for mitigating their adverse effects on source reconstruction.

2:20

2pNS6. Spatio-temporal characteristics of countryside soundscapes in Hong Kong. Kin-che Lam (Dept. of Geography & Resource Management, Chinese Univ. of Hong Kong, Shatin, NT., Hong Kong, kinchelam@cuhk.edu.hk)

Over six hundred 15-min and one hundred 24-h digital sound recordings were undertaken in different parts of the Hong Kong countryside covering different landscape units, different time of day and seasons. Spectrograms were prepared from these recordings and 24-h spectrograms were subdivided into segments with relatively uniform acoustic characteristics. These segments were then analyzed for the sound sources, acoustic and psychoacoustic characteristics. These data were analyzed to determine for their spatial and temporal characteristics. Cluster and discriminant analyses were further undertaken to ascertain how acoustic and sound sources of the countryside vary over space and time. The results indicate that soundscape characteristics are determined primarily by the type of landscape, and then by the diurnal and seasonal characteristics. The implications of these findings on human experience, sampling strategy, and countryside management and planning will be discussed.

TUESDAY AFTERNOON, 1 NOVEMBER 2011 ROYAL PALM 3/4, 1:00 TO 5:45 P.M.

Session 2pPA

Physical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration: Hypersonic Spectroscopy of Microstructured Materials

Kenneth G. Foote, Cochair
Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543

Paul A. Snow, Cochair
Dept. of Physics, Univ. of Bath, Claverton Down, The Avenue, Bath, BA2 7AY, UK

Chair's Introduction—1:00

Invited Papers

1:05

2pPA1. Hypersonic spectroscopy of porous silicon for acoustic devices. Paul A. Snow, Leigh-Anne Thomas, Bernhard Goller, and Gazi N. Aliev (Dept. of Phys., Univ. of Bath, The Ave., Bath, BA2 7AY, UK)

I will review our work on porous silicon (pSi) presenting achievements while highlighting underlying physical questions that remain to be answered. pSi is produced by the electrochemical etching of crystalline silicon. It is typically mesoporous, having pores of 10–30 nm diameter. The etching current density determines the final porosity, the volume fraction of air, with a wide range of porosities, 25%–95%, achievable. For wavelengths much greater than the pore size, pSi gives a tunable effective medium for light and sound waves. We have characterized pSi acoustic properties using transmission spectroscopy with matched transducer pairs working at 0.5–2.5 GHz. The results for velocity, v, have fitted to a general law of $v=v_0(1-\phi)^k$, where v_0 is the velocity in bulk silicon, ϕ is porosity, and k is the fitting parameter. We have investigated the variation of k with the direction of propagation and the etching conditions used to extract the dependence of the elastic constants on porosity. The measurement of velocity has enabled us to produce and characterize pSi Bragg mirrors and rugate filters that have a smoothly varying acoustic impedance. This has demonstrated the potential use of pSi in acousto-optic phoxonic crystal devices that have both phononic and photonic bandgaps.

1:30

2pPA2. Picosecond ultrasonic microscopy of semiconductor nanostructures. Thomas J. Grimsley, Shan Che (Dept. of Phys., Brown Univ., Providence, RI 02912), G. Andrew Antonelli (Novellus Systems, Albany, NY 12203), Arto V. Nurmikko, and Humphrey J. Maris (Brown Univ., Providence, RI 02912)

We describe a new picosecond ultrasonics method for the study of nanostructures. A sound pulse is generated when an ultra-short laser pulse is absorbed in a transducer structure. The sound then propagates across a thin layer of water and is reflected from the surface of the sample being examined. A resonant optical cavity is used to improve the efficiency of optoacoustic detection and generation of the sound. We report on experiments in which sound is reflected from patterned nanostructures. In these experiments, we are able to study the propagation of sound down channels of width as small as 35 nm.

1:55

2pPA3. Probing acoustical, optical, and acousto-optical properties of nanostructured materials by picosecond laser ultrasonics. V. Gusev and P. Ruello (LPEC, UMR-CNRS 6087, Universit du Maine, av. O. Messiaen, 72085 Le Mans, France)

Research results on the characterization of the thin (submicrometers thick) films of the nanostructured materials by the experimental methods of picosecond laser ultrasonics are reviewed. These methods make use of femtosecond lasers to generate and to detect GHz–THz acoustic waves. In this communication, theoretical backgrounds of the fs-laser-based opto-acousto-optic techniques that are used for the evaluation of the material properties are first introduced. Then, the results of the experiments on nanoporous low-k films (for the microelectronics), on nanogranular sol–gel optical coatings (for laser optics), on anodized alumina (for the nanomaterial/nanostructure templates), on synthetic opals, on nanoparticles supra-crystals, and other nanostructured materials are discussed. The emerging opto-acousto-optic technology for the depth-profiling of acoustical, optical, and acousto-optical properties of inhomogeneous transparent films with the nanometers scale spatial resolution is also presented.

2:20

2pPA4. Using light to probe hypersound in porous materials systems. Lance C. Parsons, Jordan Peckham, Anna M. Polomska, and G. Todd Andrews (Dept. of Phys. and Physical Oceanogr., Memorial Univ. of New Foundland, St. John's, NL, A1B 3X7, Canada, tandrews@mun.ca)

An overview of the technique of Brillouin light scattering spectroscopy and its application to the study of hypersound in micro- and mesoporous materials systems will be presented. Particular emphasis will be placed on results obtained from light scattering experiments on porous silicon-based structures. For porous silicon films, it was found that the acoustic phonon velocities and elastic properties depend strongly on the film porosity and morphology. Brillouin studies of porous silicon superlattices with periodicity on the order of the hypersound wavelength reveal that these structures behave as hypersonic phononic crystals, while those with smaller modulation wavelengths act as effective elastic media. New results on localized acoustic modes in porous silicon multilayers will also be discussed. Collectively, these studies provide a detailed picture of hypersound propagation in porous silicon systems and demonstrate the utility of Brillouin spectroscopy for probing acoustic phonon behavior in this challenging class of materials. [This work was supported by the Canada Foundation for Innovation, Memorial University of Newfoundland, and the Natural Sciences and Engineering Research Council of Canada.]

2:45

2pPA5. High frequency soft phononics. George Fytas (Max Planck Inst. for Polymer Res., Ackermannweg 10, 55128 Mainz, Germany, fytas@iesl.forth.gr)

Phononic crystals, the acoustic equivalents of the photonic crystals, are controlled by a larger number of material parameters. The study of hypersonic crystals imposes substantial demand on fabrication and characterization techniques. Colloid and polymer science offer methods to create novel materials that possess periodic variations of density and elastic properties at mesoscopic length scales commensurate with the wave length of hypersonic phonons and hence photons of the visible light. Polymer- and colloid-based phononics is an emerging new field at the interface of soft materials science and condensed matter physics with rich perspectives ahead. The key quantity is the dispersion of high frequency (GHz) acoustic excitations which is nowadays at best measured by high resolution spontaneous Brillouin light scattering. Depending on the components of the nanostructured composite materials, the resolved vibration eigenmodes of the individual particles sensitively depend on the particle architecture and their thermo-mechanical properties [T. Still *et al.*, Nano Lett. **10**, 3194 (2008)]. In periodic structures of polymer based colloids, the dispersion relation $\omega(k)$ between the frequency and the phonon wave vector k has revealed hypersonic phononic band gaps of different nature [T. Still *et al.*, Phys. Rev. Lett. **106**, 175505 (2011)].

3:10–3:25 Break

3:25

2pPA6. Engineering the band structure of one-dimensional hypersonic phononic crystals. Dirk Schneider (Max Planck Inst. for Polymer Res., Ackermannweg 10, 55128 Mainz, Germany, schneider@mpip-mainz.mpg.de), El Houssaine El Boudouti (Universitè Mohamed I, 60000 Oujda, Morocco), Faroha Liaqat, Wolfgang Tremel (Johannes Gutenberg Univ., 55128 Mainz, Germany), Hans-Jürgen Butt (Max Planck Inst. for Polymer Res., 55128 Mainz, Germany), Bahram Djafari-Rouhani (Universitè de Lille 1, 59655 Villeneuve d'Ascq, France), and George Fytas (Univ. of Crete and FORTH, 71110 Heraklion, Greece)

Phononic crystals—the mechanical analogues of photonic crystals—have attracted increasing interest and have been widely studied within the past decade. The phononic dispersion relation at hypersonic frequencies can be directly measured by the powerful non-destructive technique of high resolution spontaneous Brillouin-light-scattering (BLS) [W. Cheng *et al.*, Nature Mater. **2006**, 5, 830]. Due to the vector nature of the elastic wave propagation, theoretical phononic band structures can be uniquely verified at low dimensionality, and hence 1D phononic crystals constitute model systems for fundamental studies. Such hybrid Bragg stacks, composed of alternating layers of silica and poly(methyl methacrylate) (PMMA), respectively, exhibit clear hypersonic phononic band gaps [Gomopoulos *et al.*, Nano Lett. **2010**, 10, 980]. In this paper, we report on the fabrication, characterization, and both experimental and theoretical dispersion diagrams along and normal to the periodicity direction of silica/PMMA multilayers. The width of the gap, the phonon frequencies, and their intensities near the first Brillouin zone are sensitive probes of the longitudinal moduli and elasto-optic constants of the individual layers and structural parameters. Mixing with layer modes under oblique incidence conditions allows access to the shear moduli of the two layers.

Contributed Papers

3:50

2pPA7. A holey structured acoustic metamaterial. J. Zhu (Nanoscale Sci. and Eng. Ctr. (SINAM), 3112 Etcheverry Hall, Univ. of California, Berkeley, CA 94720), J. Christensen, J. Jung (Universidad Autonoma de Madrid, E-28049 Madrid, Spain), L. Martin-Moreno (CSIC-Universidad de Zaragoza, E-50009 Zaragoza, Spain), X. Yin, L. Fok, X. Zhang (Univ. of California, CA 94720), and F. J. Garcia-Vidal (Universidad Autonoma de Madrid, E-28049 Madrid, Spain)

The resolution of acoustic imaging system is restricted by diffraction limit. To beat this limit, early research shows acoustic metamaterials that can manipulate acoustic waves artificially and may act as lens to achieve subwavelength resolution. However, these solutions suffer significant loss therefore lack convincing experimental demonstration. Recent study suggested that arrays of metallic nanorods or nanowire can be used as lens for optical imaging at subwavelength resolution. Similar acoustic hyperlens designs have also been explored, and latest experimental result provided resolution of wavelength/7. Here presented is a holey structured endoscope which supports the transmission of the otherwise-evanescent waves over distances, therefore beating diffraction limit and achieving deep subwavelength imaging. Experimental demonstration shows clear image with feature size of wavelength/50. Such a metamaterial endoscope brings new perspectives to the applications of medical ultrasonography, sonar and ultrasonic non-destructive evaluation.

4:05

2pPA8. Mechanisms of nonlinear saturation in focused acoustic beams of periodic waves and single pulses. Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Maria Karzova, Mikhail Averiyanov (Moscow State Univ., Moscow 119991, Russia), and Oleg Sapozhnikov (Univ. of Washington, WA 98105)

Physical mechanisms leading to saturation of various acoustic field parameters in nonlinear focused beams of periodic waves and single pulses were investigated numerically. A numerical algorithm based on the KZK equation was used in the simulations. Propagation of an initially harmonic wave and a single pulse (one period of a sine wave) emitted by a focused transducer with Gaussian apodization was modeled. It was shown that in periodic fields, saturation of the peak positive pressure is mainly due to the effect of nonlinear absorption at the shock front. In acoustic fields of single pulses the main mechanism of saturation is the nonlinear refraction. Maximum pressure in the periodic field, achieved at the focus, was found to be higher than that of the single pulse. The total energy of the beam of the periodic wave, however, decreases much faster with the distance from the source as compared to the single pulse focusing. These nonlinear propagation effects propose a possibility to use pulsed beams for more effective delivery of the wave energy to the focal region, and periodic waves—to achieve higher pressure values of at the focus. [Work supported by EB007643, NIH DK43881, DK075090, and RFBR 09-02-01530.]

4:20

2pPA9. A study of the nonlinear effects of air bubbles on the ultrasonic field in water. Christian Vanhille (Universidad Rey Juan Carlos, Tulipan s/n, 28933 Mostoles, Madrid, Spain, christian.vanhille@urjc.es) and Cleofé Campos-Pozuelo (Consejo Superior de Investigaciones Cientificas, 28006 Madrid, Spain)

We consider the propagation of ultrasonic waves in water with air bubbles. On the one hand, a numerical model has been developed to analyze the nonlinear effects of the bubbles at high amplitude, in several configurations (open-field, standing waves, 1–D, 2–D, 3–D, homogeneous bubble density, bubble layers, bubble clouds, bubble generation). On the other hand an experimental setup has been constructed and allows us to study the nonlinear behavior of the inertial cavitation field at high frequency and at high power. In particular, a bubble cloud is formed at a large distance from the sonotrode. [Work is part of the research project DPI2008-01429 funded by the Spanish Ministry of Science and Innovation.]

4:35

2pPA10. Dynamics of bubble clusters in acoustic field. I. S. Akhatov (Dept. of Mech. Eng., North Dakota State Univ., Fargo, ND 58108), E. S Nasibullaeva, Y. V. Volkova (Ctr. for Micro and Nanoscale Dynam. of Dispersed Systems, Bashkir State Univ., Ufa 450074, Russia), and N. A. Gumerov (UMIACS, Univ. of Maryland, College Park, MD 20742)

A bubble cluster composed of gas bubbles of various radii oscillating in an unbounded, slightly compressible viscous liquid under the action of an external acoustic field is considered. The mathematical model describing the dynamics of this bubble cluster is presented. The proposed model is used for an analytical study of small (linear) bubble oscillations in monodisperse and polydisperse clusters, for a numerical investigation of large (nonlinear) bubble oscillations, and for a diffusion stability analysis of gas bubbles in the cluster. The following phenomena have been revealed: (1) synchronization of the collapse phases of bubbles with different radii and (2) collapse intensification for bubbles of one size in the presence of bubbles of other size. These effects are explained by the interaction between bubbles of different radii in the cluster. For a monodisperse (one-fraction) cluster, numerical values were obtained for the initial gas concentrations in the liquid at which bubbles tend to one of two equilibrium states due to the rectified diffusion. It is also found that a polydisperse (two fraction) cluster tends to become a one fraction cluster due to the rectified diffusion. [This research is supported by the Grant of the Ministry of Education and Science of the Russian Federation (G34.31.0040).]

4:50

2pPA11. Soil plate oscillator: Modeling nonlinear mesoscopic elastic behavior and hysteresis in acoustic landmine detection, Part II. Dang V. Duong (Weapons and Systems Eng. Dept., U.S. Naval Acad., Annapolis, MD 21402) and Murray S. Korman (U.S. Naval Acad., Annapolis, MD 21402)

An apparatus (SPO), designed to study flexural vibrations of a soil loaded plate, consists of a thin circular elastic clamped plate (and cylindrical wall) supporting a vertical soil column. A small magnet attached to the center of the plate is driven by a rigid AC coil (located coaxially below the plate) to complete the electrodynamic soil plate oscillator SPO design. The mechanical impedance Z_{mech} (force/particle velocity, at the plate's center) versus frequency is inversely proportional to the electrical motional impedance Z_{mot}. Measurements of Z_{mot} are made using the complex output to input response of a Wheatstone bridge that has an identical coil element in one of its legs. Resonant oscillation measurements (with no soil) before and after a slight point mass loading at the center help determine effective mass, spring, damping and coupling constant parameters of the system. "Tuning curve" behavior of real Z_{mot} and imaginary Z_{mot} at successively higher vibration amplitudes exhibit a decrease "softening" in the resonance and an increase in the quality Q factor. A bilinear hysteresis model [T. K. Caughey, ASME, J. Applied Mech. Trans. 640 (1960)] predicts the tuning curve shape for this nonlinear mesoscopic elastic SPO behavior.

5:05–5:45 Panel Discussion

Session 2pSA

Structural Acoustics and Vibration: Session in Honor of Gideon Maidanik

Richard H. Lyon, Cochair
RH Lyon Corp, 60 Prentiss Ln., Belmont, MA 02478

Joseph W. Dickey, Cochair
3960 Birdsville Rd., Davidsonville, MD 21035

Chair's Introduction—1:15

Invited Papers

1:20 Open microphone—Reminiscences

1:40

2pSA1. Radiation efficiency, impedance, and the acoustics of rattle. Philip Shorter, Vincent Cotoni (ESI Group, 12555 High Bluff Dr., Ste. 250, San Diego, CA 92130), and Robin Langley (Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom)

Rattle issues consistently rank as one of the top consumer complaints in initial quality surveys for many new products. Predicting the acoustics of rattle is complicated by the need to model the vibro-acoustic response of large complex structures across a broad frequency range. The complexity of the analysis can be reduced by making use of standard methods derived almost 50 years ago. In particular, this paper discusses a computationally efficient method for assessing the propensity for rattle in large complex structures. A finite element model is used to predict the probability that impacts will occur when a product is exposed to a particular low frequency random vibro-acoustic environment. The expected contact forces arising from each impact are then estimated by making use of expressions involving the drive point impedances of infinite structures. Finally, the vibration and acoustic radiation associated with the various impacts are predicted and ranked using a SEA model. A number of examples are presented including one which makes use of a radiation efficiency formula due to G. Maidanik. The results are in very good agreement with analytical reference results.

2:00

2pSA2. Controlling the response of an oscillator using a coupled set of satellite oscillators. Ronald G. Hughes (NSWCCD 9500 MacArthur Blvd., West Bethesda, MD 20877) and Gideon Maidanik (Deceased)

The response of a system comprised of multiple dynamic systems is analyzed. The results shown here are for the main or master oscillator in that system. The balance of the dynamic systems is designated as satellite oscillators. Controlling the response of the master oscillator is described in terms of the couplings to the master oscillator, the frequency distribution of the satellite oscillators, the loss factors, and the masses of those oscillators. The frequency distributions and masses of the satellite oscillators are specified via normalizations with respect to the resonance frequency and mass of the master oscillator in order to generalize the approach. It is shown that contrary to reported results by others, there is no requirement to optimize the frequency distribution of the satellite oscillators to maximize the control of the response of the master oscillator. Further it is shown that increasing the loss factor of the satellite oscillators beyond certain values does not bring further benefit in controlling the response of the master oscillator beyond a certain level, in fact, a saturation is reached. We describe the on-set of saturation in terms of the modal overlap parameter.

2:20

2pSA3. Shaping the response of multi-degree-of-freedom mechanical systems. Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

The paper pays homage to the outstanding contribution by Gideon Maidanik to studying synergic action of multiple add-on oscillators in reducing resonance response of a main structure. Methods and results of constructing multi-degree-of-freedom mechanical systems with required responses in frequency and time domains are presented with application to optimal vibration isolation, impedance matching and shock absorption. It is shown, in particular, that the limiting quality of instantaneous shock isolation, which was usually attributed to highly non-linear, heavily damped or active systems, can be achieved by linear low-damped multi-degree-of-freedom mechanical systems. Such systems can be synthesized as multiple vibration absorbers, stacked oscillators (chains), or more general structures including motion transformation elements.

2:40

2pSA4. Fuzzy structures applied to a vibrating beam. David Feit (Acoust. Society of America, Ste. 1NO1, 2 Huntington Quadrangle, Melville, NY 11757-4502)

Gideon and I were both introduced to the ideas of "Fuzzy Structure Theory" during a visit with Christian Soize while involved in an exchange meeting between the U.S. Navy and the French Navy in 1986. In the years just prior to his passing, he together with Ron Hughes pursued this subject, and that work is also presented in this session. My presentation, a continuation of work that I had originally done with Murray Strasberg, discusses the transient response of a multiple set of fuzzy structures attached to a master structure, either a longitudinally or flexurally vibrating beam, that itself has multiple resonances. The transient response of the subordinate oscillators in certain parameter ranges gives rise to distinct packets of energy traveling with different wave speeds. This phenomenon has not been previously observed, and a tentative explanation is offered.

3:00–3:15 Break

3:15

2pSA5. Thoughts on submarine structural acoustics. Ira Dyer (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA 02139)

The simplest submarine model is a uniform circularly cylindrical shell with flat end caps. This is too simple to be useful, but the recent phenomenal advances in computational structural acoustics, and the availability of test and laboratory facilities, make it feasible to tackle greater complexity in the submarine structure. But the newer numerical computational analyses, and standard experimental approaches, have as their initial outputs large data sets. So, in effect, structural acousticians have traded simpler and less realistic structural models, for more dense data sets and more useful models, a trade bound to be highly positive. In this context, the author has observed, subjectively, that those who have the ability to aggregate their large data sets with physics-centric descriptions do present their results crisply and with deep understanding. Also, they are better prepared to adjudicate alternate interpretations and to suggest further steps to reach more unique conclusions. Accordingly, the author suggests that specialized data filters be researched and developed for use in data interpretation, these to be formulated from questions such as: Where in the submarine structure, and in what wavenumber band, does wavenumber matching needs to be controlled? What is the modal character of an internal stiffening ring, and where should absorptive sinks of resonance peaks be placed? Additional questions are posed and discussed in the paper.

3:35

2pSA6. State transitions in the Duffing resonator excited by narrow band random noise. Richard H. Lyon (60 Prentiss Ln., Belmont MA 02478-2021, rhlyon@lyoncorp.com)

The Dufffing hardening spring resonator has the status of a canonical model for nonlinear vibrations. Over a limited range of excitation frequencies and depending on the degree of nonlinearity, it has three states of response to sinusoidal excitation, one of which is unstable. The system will remain stably in either of the other two states depending on the history of excitation: in a higher energy state for the frequency ascending and in the lower energy state for the frequency descending. When the sinusoidal excitation is replaced by narrow band random excitation, the author showed in a 1961 paper experimentally that the system could make transitions between these two states and argued that the fluctuating phase of the excitation would allow the source to inject or draw energy from the resonator allowing a transition from one state to the other. This presentation develops a dynamic model for the system that allows energy transmission between the source and resonator and an indicator of the state of response based on instantaneous impedance.

3:55

2pSA7. Acoustic radiation pressure, torques, and scatterings: Insight for today from Gideon Maidanik's thesis research. Philip L. Marston, Likun Zhang, and David B. Thiessen (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Renewed interest in the identification of structural contributions to the scattering by complicated objects and the radiation forces and torques on simple objects in complicated acoustic beams makes it timely to review aspects of Maidanik's thesis research [J. Acoust. Soc. Am. 29, 738-742 (1957); 29, 936-940 (1957); 30, 620-623 (1958)]. Maidanik and Westervelt recognized the importance of King's earlier study of the low-frequency radiation forces on rigid movable spheres and extended that work to ka of 10. Hickling and Wang (1966) examined the properties of the scattering by such spheres and more recently Marston [J. Acoust. Soc. Am. 125, 3539-3547 (2009)] found that such a sphere approximates the low ka backscattering by an aluminum sphere in water. It was also found that negative radiation forces were possible for movable rigid spheres in a helicoidal Bessel beam. Recently, subtraction of complex amplitudes for movable semi-rigid targets has been used to isolate elastic contributions in finite-element based computations for finite cylinders [D. B. Thiessen *et al.*, J. Acoust. Soc. Am. 129, 2686 (2011)]. Maidanik's analysis of radiation torques has recently been reconciled with other approaches [L. K. Zhang and P. L. Marston, J. Acoust. Soc. Am. 129, 1679-1680 (2011)]. [Work supported by ONR and NASA.]

4:15

2pSA8. Estimating uncertainty in inverse elasticity with application to quantitative elastography. Paul E. Barbone, Bryan Chue (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), and Assad A. Oberai (Rensselaer Polytechnic Inst., Troy, NY 12180)

We consider an inverse elasticity problem motivated by medical ultrasound imaging: Given a displacement field measured in a 2D domain, determine the modulus distribution in that domain. An iterative approach to solve the inverse problem can be formulated by repeated solutions of the forward problem. That is, the shear modulus distribution sought is that which predicts a displacement field most consistent with the measured displacement field and any assumed *a priori* knowledge of the modulus distribution. All such inverse problem solutions are subject to uncertainties in the data, however, which results in uncertainties in the predictions. For diagnostic purposes, it is desirable to know the confidence intervals within which the stiffness at a point might reside. The focus of this presentation is

the computation of said confidence intervals. We discuss the formulation of the problem within a Bayesian context. We derive a formal solution for the *a posteriori* probability distribution of the modulus. We prove bounds on uncertainty in terms of the data at the continuous level and discuss the computational solution of the problem at the discrete level.

4:35

2pSA9. Wave approach for the resonances of irregular polygonal membranes. Joseph Dickey (3960 Birdsville, Rd. Davidsonville, MD 21035, Joe@JoeDickey.com)

This study develops a wave or ray technique for determining the resonance frequencies of irregular polygonal membranes. The technique is demonstrated for homogeneous, isotropic, rectangular, and triangular membranes with fixed, free, and mixed boundaries. Where possible, the results are compared with exact calculations. The membrane resonances are calculated using an equivalent string whose length is proportional to the reciprocal of the length of closed paths starting from an arbitrary point within the membrane. Closed paths are ray paths which arrive back at the starting point going in the same direction. The extension of the technique to other irregular polygons and the relationship of the resonance determination in determining the response of the membranes to point excitation are discussed.

4:55

2pSA10. Sound-structure interactions in a Japanese drum. Yun-Fan Hwang (Fanacoustics, Inc., 3024 Rancho La Presa, Carlsbad, CA 92009, yfhwang1@gmail.com) and Hideo Suzuki (A and D Co., Ltd., Kitamoto 364-8585, Japan)

Previous studies of the sound-structure interaction of a Japanese drum conducted by the authors were focused on the vibration of and the coupling between the two membranes attached at both ends of an air-filled hollow wood body which was treated as a rigid cylindrical shell. This is satisfactory for the lower modes where sound is produced primarily by the vibration of membranes. At higher frequencies, the vibration of the wood barrel cannot be ignored. In the current study, the wood barrel is modeled by using conical shell elements. Orthotropic conical shell finite-elements, which include the rotary inertia and transverse shear deformation, have been developed and coded in MATLAB. Experimental verification of the computed results and the effect of wood barrel vibrations on the acoustical characteristics of a drum are discussed. [This paper is dedicated to honor Dr. Gideon Maidanik for his monumental contributions in structural acoustics. The authors would like to thank Miyamoto Unosuke Shouten Co., Ltd., for providing the Japanese drum in this study.]

TUESDAY AFTERNOON, 1 NOVEMBER 2011 PACIFIC SALON 4/5, 1:00 TO 5:00 P.M.

Session 2pSCa

Speech Communication: Error Analysis in Automatic and Human Speech Recognition

Amit Juneja, Chair
Think A Move, Ltd., 23715 Mercantile Rd., Ste. 100, Beachwood, OH 44122

Chair's Introduction—1:00

Invited Papers

1:05

2pSCa1. Finding schwa: Comparing the results of an automatic aligner with human judgments when identifying schwa in a corpus of spoken French. Peter Milne (Dept. of Linguist., Univ. of Ottawa, 70 Laurier Ave. East, Ottawa, ON K1N 6N5, Canada, pmiln099@gmail.com)

This paper compares the results of an automatic aligner with human judgments when identifying schwa in a natural language corpus of spoken French. All word-final, postconsonantal possible schwa insertion sites in the standardized text portion of investigations conducted in both Qu'ebec and France were both hand coded for schwa and time aligned at the word and phone level by the Penn Phonetics Lab Forced Aligner, modified for use with French. The results of the two methods of coding were statistically compared to determine their level of agreement. Preliminary results show a strong correlation between the two methods. Possible effects due to dialect or phonetic context were investigated using a two-way, between subjects analysis of variance. A mixed design analysis of variance was also conducted. Initial results have found no significant effect due to dialect, but a possible effect due to context. This suggests that the results of automatic alignment are comparable with human judgments in both dialects of French, but results may differ for individual speakers in specific contexts. The advantages of using an automatic aligner that can aid in the collection of larger volumes of natural language data than is possible when working manually is discussed.

1:25

2pSCa2. Improving automatic speech recognition by learning from human errors. Bernd T. Meyer (Int. Comput. Sci. Inst., 1947 Center St., Ste. 600, Berkeley, CA, bmeyer@icsi.berkeley.edu)

This work presents a series of experiments that compare the performance of human speech recognition (HSR) and automatic speech recognition (ASR). The goal of this line of research is to learn from the differences between HSR and ASR and to use this knowledge to incorporate new signal processing strategies from the human auditory system in automatic classifiers. A database with noisy nonsense utterances is used both for HSR and ASR experiments with focus on the influence of intrinsic variation (arising from changes in speaking rate, effort, and style). A standard ASR system is found to reach human performance level only when the signal-to-noise ratio is increased by 15 dB, which can be seen as the human–machine gap for speech recognition on a sub-lexical level. The sources of intrinsic variation are found to severely degrade phoneme recognition scores both in HSR and in ASR. A comparison of utterances produced at different speaking rates indicates that temporal cues are not optimally exploited in ASR, which results in a strong increase of vowel confusions. Alternative feature extraction methods that take into account temporal and spectro-temporal modulations of speech signals are discussed.

1:45

2pSCa3. Automatic and human speech recognition in null grammar. Amit Juneja (Think A Move, Ltd., 23715 Mercantile Rd, Ste. 100, Beachwood, OH 44122)

The accuracy of automatic speech recognition (ASR) systems is generally evaluated using corpora of grammatically sound read speech or natural spontaneous speech. This prohibits an accurate estimation of the performance of the acoustic modeling part of ASR, since the language modeling performance is inherently integrated in the overall performance metric. Even though acoustic modeling accuracy for ASR can be evaluated on these corpora using a null grammar language model, the accuracy cannot be compared with human speech recognition (HSR) since human listeners cannot be asked to ignore grammar. In this work a null grammar speech corpus was collected for comparing HSR and ASR. The corpus was collected in a hemi-anechoic chamber using three different vocabulary sizes—1000, 5000, and 10000—in a quiet environment. Noisy speech files at different signal-to-noise ratios were generated by adding noise at different levels to the quiet speech recordings. Human listeners were employed to transcribe the recordings and their accuracy was compared with an ASR system under different vocabularies and noise levels.

2:05

2pSCa4. Being wrong—Insights from the speech-recognition battleground. Thomas U. Christiansen (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, DK-2800, Kgs. Lyngby, Denmark, tuc@elektro.dtu.dk) and Steven Greenberg (Silicon Speech, Kelseyville, CA 95451, steveng@silicon-speech.com)

Speech-recognition studies rarely use more than a single metric to evaluate recognition performance, usually percent correct (or percent wrong). Such a uni-dimensional evaluation may conceal more than it reveals. An alternative, based on information theory, offers greater insight into brain (and computational) processes associated with human and machine speech recognition. In this presentation, we examine errors associated with phonetic-segment recognition in human listeners and compare them with those committed by automatic speech-recognition (ASR) systems. Consonant errors are analyzed into the phonetic features of VOICING, place—(PLACE) and manner—(MANNER) of—articulation. For both humans and machines, PLACE information is far more vulnerable to distortion/interference than MANNER and VOICING, but is more important for consonant and lexical recognition than the other features. Moreover, PLACE is decoded only after VOICING and MANNER and is more challenging for machines to accurately recognize. The origins of these differences can be traced, in part, to the redundancy with which this information is distributed in the acoustic signal, as well as how the phonetic information is combined across the frequency spectrum. For such reasons, ASR performance could benefit by including phonetic-feature-based information in lexical representations. [Work supported by AFOSR and Technical University of Denmark.]

2:25

2pSCa5. Towards the recovery of targets from coarticulated speech for automatic speech recognition. John-Paul Hosom, Alexander Kain, and Brian Bush (Ctr. for Spoken Lang. Understanding, Oregon Health & Sci. Univ., 20000 NW Walker Rd., Beaverton, OR 97006)

An HMM-based ASR system tested on phoneme recognition of TIMIT (accuracy 74.2%) shows substitution errors covering all distinctive-feature dimensions of vowels: front/back, tense/lax, and high/low. These vowel-to-vowel errors account for about 30% of all substitution errors. These types of errors may be addressed by recovering vowel targets (and, as a by-product, coarticulation functions) during ASR. The current work models observed trajectories using a linear combination of target vectors, one vector per phoneme. A sigmoid function (with parameters for slope and position) models the evolution of the trajectory. In accordance with the Locus theory, if duration is sufficiently short and the rate of change is sufficiently slow, the targets may not be reached. Current data indicate that in clearly articulated speech, the vowel target is often reached, while in conversational speech, the vowel target is often not reached. This difference between speaking styles may explain the difficulty that current ASR systems have in recognizing conversational speech: by not always reaching the vowel target, the observed values for a phoneme have higher variance and increased overlap with other phonemes. By recovering the target values, variance of phonemes within the feature space may be reduced, thereby improving classification accuracy. [Work supported by NSF Grant IIS-0915754.]

2:45

2pSCa6. How knowledge of speech acoustics can improve the robustness of automatic speech recognition. Carol Espy-Wilson (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD 20742), Suzanne Boyce (Univ. of Cincinnati, Cincinnati, OH 45267), and Abeer Alwan (UXLA, Los Angeles, CA)

In this talk, we discuss the ways in which the parallel study of speech production and speech perception can help us develop better automatic speech recognition systems. The ultimate goal of speech recognition (recognition of spontaneous speech from any talker in any language) is still elusive due to a high degree of inter- and intra-speaker variability for production of a given sequence of sounds. While the acoustic information required for recognition may be present in the signal, its distribution, strength, and location are consistent and predictable only as a function of lawful changes in speech movements and/or listener perceptions. Understanding speech acoustics from this perspective is vitally important if we are going to achieve our ultimate goal. We will give several examples of lessons learned from studies of speech production and speech perception and how the knowledge gained can inform the engineering of robust ASR systems.

3:05–3:20 Break

3:20

2pSCa7. Recent progress in articulatory modeling for speech recognition. Karen Livescu (Toyota Technol. Inst. at Chicago, 6045 S. Kenwood Ave., Chicago, IL 60637)

The automatic speech recognition research community has experimented with models of speech articulation for several decades, but such models have not yet made it into mainstream recognition systems. The difficulties of adopting articulatory models include their relative complexity and dearth of data, compared to traditional phone-based models and data. This talk will review the current state of articulatory models and will describe one particular approach to incorporating such models in modern speech recognition. In this approach, the articulatory variables are based on the vocal tract variables of articulatory phonology, and the models are represented using dynamic graphical models, a generalization of the more commonly used hidden Markov models. This approach allows the probabilistic modeling of asynchrony between articulators and reduction in articulatory gestures. Results will be presented showing improvements in lexical access using this type of articulatory model with automatically learned context-dependent articulatory feature distributions. Recent efforts to mitigate the data sparseness problem, including manual and automatic transcription, will also be presented.

3:40

2pSCa8. Robust speech recognition with articulatory features using dynamic Bayesian networks. Vikramjit Mitra (Speech Technol. and Res. Lab., SRI Int., 333 Ravenswood Ave., Menlo Park, CA 94025), Hosung Nam (Haskins Labs., New Haven, CT 06511), Carol Espy-Wilson (Univ. of Maryland, College Park, MD 20742), Elliot Saltzman (Boston Univ., Boston, MA 02115), and Louis Goldstein (Univ. of Southern California, Los Angeles, CA 90089)

Previous studies have proposed ways to estimate articulatory information from the acoustic speech signal and have shown that when used with standard cepstral features, they help to improve word recognition performance in noise for a connected digit recognition task. In this paper, I present results from a word recognition and a phone recognition experiments in noise that uses two sets of articulatory representation: continuous (tract variable trajectories) and discrete (articulatory gestures) along with standard mel cepstral features for acoustic modeling. The acoustic model is a dynamic Bayesian network (DBN) that treats the continuous articulatory information as observed and the discrete articulatory presentation as hidden random variables. Our results indicate that the use of articulatory information improved noise robustness for both the word recognition and phone recognition tasks substantially.

4:00

2pSCa9. Semi-supervised learning for speech and audio processing. Mark A. Hasegawa-Johnson, Jui-Ting Huang, and Xiaodan Zhuang (ECE Dept., Univ. of Illinois. Urbana, IL 61801)

Semi-supervised learning requires one to make assumptions about the data. This talk will discuss two different assumptions, and algorithms that instantiate those assumptions, for the tasks of acoustic modeling and pronunciation modeling in automatic speech recognition. First, the acoustic spectra corresponding to different phonemes overlap, but there is a tendency for the instantiations of each phoneme to cluster within a well-defined region of the feature space—-a sort of "soft compactness" assumption. Softly compact distributions can be learned by an algorithm that encourages compactness without strictly requiring it, e.g., by maximizing likelihood of the unlabeled data, or even better, by minimizing its conditional class entropy. Second, the observed phone strings corresponding to coarticulated pronunciations of different words are also, often, indistinguishable, but can be transformed into a representation in which the degree of overlap is substantially reduced. The canonical phonetic pronunciations are transformed into an articulatory domain, possible mispronunciations are predicted based on a compactness criterion in the articulatory domain, and the result is transformed back into the phonetic domain, forming a finite state transducer that is able to effectively use hundreds of alternate pronunciations.

4:20

2pSCa10. Dealing with unknown unknowns in speech. Hynek Hermansky (Ctr. for Lang. and Speech Processing, The Johns Hopkins Univ., Baltimore, MD 21218)

Common belief in speech recognition community is that most significant improvements in performance on a machine come from more training data. Implicit is a tacit assumption that speech to be recognized comes from the same distribution as the speech on which the machine was trained. Problems occur when this assumption is violated. Words that are not in a lexicon of a machine, unexpected distortions of a signal and noises, unknown accents, and other speech peculiarities all create problems for the current ASR. The problem is inherent to machine learning and will not go away unless alternatives to extensive reliance on false beliefs of unchanging world are found. In an automatic recognition of speech, words that are not in the expected lexicon of the machine are typically substituted by some acoustically similar but nevertheless wrong words. Similarly, unexpected noise is typically ignored in human speech communication but causes significant problems to a machine. We discuss a biologically inspired multistream architecture of a speech recognition machine that could alleviate some of the problems with the unexpected acoustic inputs. Some published experimental results are given.

4:40–5:00 Panel Discussion

Session 2pSCb

Speech Communication: Deep Brain Stimulation in Parkinson's Disease: Speech and Nonspeech Outcomes

Emily Q. Wang, Cochair
Communication Disorders and Science, Rush Univ., 1653 W. Congress Pkwy., Chicago, IL 60612

Anders L. Lofqvist, Cochair
Dept. of Logopedics and Phoniatrics, Lund Univ., Lund, S-221 85, Sweden

Charles R. Larson, Cochair
Communication Science Disorders, Northwestern Univ., 2240 N. Campus Dr., Evanston, IL 60208

Chair's Introduction—1:30

Invited Papers

1:35

2pSCb1. Deep brain stimulation in Parkinson's disease: The basics. Leo Verhagen (Dept. of Neurological Sci., Rush Univ. Medical Ctr., 1725 W. Harrison, Chicago, IL 60612, lverhage@rush.edu)

Parkinson's disease (PD) affects over 1×10^6 people in the United States with 50 000 Americans being diagnosed each year. As a neurodegenerative movement disorder, it affects patients' lives through its slow but relentless progression of both motor and non-motor symptoms. Initially, most PD patients receive good benefit from dopaminergic treatment, but over time the symptomatology changes. The goal of this presentation is to discuss the motor features of advanced PD. Treatment of advanced PD typically consists of a combination of pharmaceuticals, but in recent years deep brain stimulation has been increasingly used to complement medical therapy. DBS for PD is indicated in some patients while it may not be the best treatment option for others. Selection criteria, indications, and relative contraindications will be discussed. The procedure will be reviewed and an overview of recent outcomes of DBS studies in PD will be provided. Attention will also be given to potential side effects of this state of the art treatment. Throughout the presentation, video clips will highlight the phenomenology under discussion.

2:05

2pSCb2. Effect of deep brain stimulation on speech in Parkinson's disease: From research lab to clinic. Emily Wang (Dept. of Commun. Disord. & Sci., 1611 W. Harrison, Ste. 530, Chicago, IL 60612, emily_wang@rush.edu)

Bilateral deep brain stimulation (DBS) of the subthalamic nucleus (STN), an evidence-based, effective surgical treatment with increasing popularity, can help patients with advanced Parkinson's disease eliminate or lessen many of the motor symptoms they experience. However, one of its unfortunate adverse effects is that it may worsen existing speech impairment as well as causing new impairment. As a result, the patients may be unable to speak loudly or clearly, or simply unable to initiate speech. The problems often associate with active as well as chronic STN stimulation. Further, there does not seem to be a uniform effect on different speech subsystems of respiration, phonation, and articulation. In this presentation, we will first review studies which have shown differential effects on phonation, articulation, and prosody associated with unilateral vs. bilateral STN DBS. Next, we will discuss the preliminary outcomes of several potential treatment approaches and strategies including the Lee Silverman voice treatment, change of DBS settings, and the altered auditory feedback. Lastly, we will discuss our experience of using the approaches and strategies helping patients with their speech deficits associated with STN DBS in clinical settings. Both mechanisms and limitations will be explored.

2:35–2:50 Break

2:50

2pSCb3. Effect of subthalamic nucleus deep brain stimulation on tremor, rigidity, muscle strength, and movement. Daniel M. Corcos (Dept. of Kinesiology and Nutrition, Univ. of Illinois at Chicago, 1919 West Taylor St., Chicago, IL 60612)

The goal of this presentation is threefold. First, we will review studies in which we have shown the dramatic benefit of subthalamic nucleus (STN) deep brain stimulation (DBS) on bradykinesia, tremor, and rigidity. We will show that limb tremor is normalized, movement speed is increased, muscle activation patterns resemble those of healthy individuals, and rigidity is substantially reduced. Second, we will show that there is no difference between 90 min and greater than 3 months of STN stimulation for both the UPDRS or motor control measures. This finding confirms that the treatment efficacy that is derived from a short time course of stimulation generalizes to the longer time periods of STN stimulation that patients experience in their daily lives. Finally, we will conclude by presenting the effects of five years of continuous STN stimulation on muscle strength and movement speed. We will show that despite the fact that patients become more parkinsonian as measured by the UPDRS, they become stronger and faster at making simple movements. These results will be discussed in the context of models of therapeutic efficacy that are predicated on the idea that STN DBS reduces neuronal noise and thus both facilitates simple movements and the reduction of tremor.

3:20

2pSCb4. Differential effects of deep brain stimulation (DBS) on speech and limb movements in Parkinson's Disease (PD): Clues to basic mechanisms. Howard Poizner (Inst. for Neural Computa., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093)

A puzzling finding has been that STN DBS improves limb motor function, while at the same time providing little benefit to or even markedly worsening speech. In collaboration with Peter Fox's Research Imaging Institute in San Antonio, we combined objective measures of speech with PET imaging and TMS in a PD patient with speech deficits due to STN DBS. Stimulating the left STN produced deteriorated speech together with hyperactivation of left dorsal premotor cortex (PMd), an area known to project to the left STN. TMS to left PMd with DBS off, produced similar speech impairments as DBS stimulation. These findings are consistent with STN DBS antidromically disrupting left PMd activity, thereby causing speech deterioration. In contrast, STN DBS is known to improve limb motor control. We have been examining the effects of STN DBS on reaching to and grasping objects, on reaching to kinesthetically defined 3-D targets, and on inhibiting a pre-programmed action. We have found that bilateral STN DBS improves all of these behaviors, although to varying degrees. Moreover, EEG recordings during response inhibition suggested that the physiological mechanism for the improved behavioral control involves normalization of brain rhythms that may be involved in transferring information within cortico-basal ganglia circuits. [Work supported in part by NIH grant #2 R01 NS036449 and ONR MURI Award No.: N00014-10-1-0072.]

3:50–4:20 Panel Discussion

TUESDAY AFTERNOON, 1 NOVEMBER 2011 — ROYAL PALM 1/2, 1:00 TO 3:15 P.M.

Session 2pSP

Signal Processing in Acoustics: Target Detection and Sonar Related Issues

Jan Dettmer, Cochair

School of Earth and Ocean Sciences, Univ. of Victoria, Victoria, BC V8W 3P6, Canada

Ravi Menon, Cochair

Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

1:00

2pSP1. Mutual interference signal processing for active sonar. Stephen D. Unruh, Jason M. Aughenbaugh, and James M. Gelb (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

Active sonar for anti-submarine warfare (ASW) can at times be hampered by interference from the transmissions of other vessels—either friend or foe. For practical reasons, there are typically only a limited number of center-frequencies, bandwidths, and useful waveform types available. We explore the cross-correlation of various frequency modulated waveforms with varying characteristics—particularly bandwidths, pulse lengths, and the degree of frequency overlap. The impact on the ambiguity function is also explored, along with the efficacy of several transmitter and receiver filters. Real world data containing interfering transmissions are presented including examples where basic theoretical effects are revealed. This work represents an early stage of research to ultimately explain subtle nuances in real data and to develop novel signal processing techniques to mitigate interference. [Work sponsored by ONR 321US.]

1:15

2pSP2. Characterization of non-Gaussian, multi-static clutter from a mud volcano field. John R. Preston (ARL, Penn State Univ,, P.O. Box 30, MS 6110D, State College, PA 16804) and Douglas A. Abraham (Causa Sci LLC, P.O. Box 627, Ellicott City, MD 21041)

Sonar clutter is one of the primary limitations to active ASW. This work focuses on statistical analysis of clutter-like returns from some multi-static measurements. Non-Gaussian characterizations of multi-static clutter from a mud volcano field are presented. The received data are taken from either the Five Octave Research Array (FORA) or the NURC triplet array that have been used to collect extensive monostatic and bistatic data in a recent sea trial on the Malta Plateau off Sicily called Clutter 07. This work uses data from that sea trial to characterize non-Gaussian behavior of multi-static clutter from a mud volcano field using pulsed sources in the 800–3500 Hz band. Either the Page test or a maximum likelihood procedure is used to isolate the clutter-like returns before processing. K-distributions with their shape and scale parameters are used to describe non-Gaussian behavior together with the models of Abraham and Lyons to infer physical descriptors from the clutter. The ability to geo-reference key statistical measures of clutter allows CFAR processors to adaptively set thresholds and reduce false alarms. Examples are shown to demonstrate this. Also included are presentations of the shape parameter versus bistatic aspect angle and the cumulative density functions for this parameter. [Work supported by ONR code 321US.]

1:30

2pSP3. Understanding the feedback effect through the physics of the diffraction-based sensitivity kernel for a target in shallow water: A small-scale experimental demonstration. Christian Marandet, Philippe Roux, Patrick La Rizza (ISTerre, BP 53, 38 041 Grenoble Cedex 9, France, christian.marandet@gmail.com), and Barbara Nicolas (GIPSA-LAB, 38 042 St Martin d Hres, France)

Using the feedback effect, we experimentally detect a wavelength-sized target in a shallow ultrasonic waveguide between two source-receiver transducers on acoustic feedback. The waveguide represents a 1-km-long, 50-m-deep ocean acoustic channel at the 1/1000 scale. The feedback phenomenon, or Larsen effect, occurs when a source and a receiver are connected both acoustically through the propagation medium and electrically through an amplifier in such a way that the received signal is simultaneously and continuously added to the emitted signal. A resonance is obtained when the emitter and the receiver are in phase. This resonance is very sensitive to any change in the medium which makes it a good observable for target detection. In presence of a target in the waveguide, the numerical gain of the feedback effect has to increase in order to compensate the scattering of the acoustic field from the target. In a separate experiment, the scattered field may also be recorded in a transmission configuration from the same couple of emitter/receiver with an impulse as a source signal. A comparison is made between the two different approaches.

1:45

2pSP4. Acoustic diver deterrent in a shallow harbor using time reversal acoustics. Alexander Sutin and Yegor Sinelnikov (Sound Interventions, Inc., 25 Health Sciences Dr., Ste. 201, Stony Brook, NY 11790, ysinelnikov@yahoo.com)

Protection of domestic harbors against surface and underwater threats is an important task of port security. A significant security risk is associated with scuba divers ability to carry explosives. The diver detection sonars have been developed and there is a need to compliment it with low cost acoustic swimmer deterrent. Previous research demonstrated that time reversal acoustic (TRA) system can focus intensive sound using acoustic noise from a diver. This paper discusses the feasibility of applying TRA principles for the focusing of sonar ping-pulse reflected from a diver. The advantages of the TRA focusing system and relevant operating parameters are demonstrated using the model of shallow sea, where propagation of sonar pulse is affected by numerous reflections from surface and bottom. The developed model was used to estimate and compare an effective zone of the diver deterrence taking into consideration the sonar pulse reflections from diver, bottom and surface. The main objective was to estimate if NAVY ship sonar is capable of producing sufficient and spatially localized sound pressure to disorient and divert the diver away. A secondary objective was to ensure that sound pressure level outside the focal zone in the diver proximity remains not harmful to marine life.

2:00

2pSP5. A multi-beam array technique for acoustical imaging. John F. Brady and Dipen N. Sinha (Los Alamos Natl. Lab., MSD429, Los Alamos, NM 87544, jbrady@lanl.gov)

Most acoustical imaging systems rely on phase-steered or multiplexed transducer arrays, requiring complex electronics and powerful data acquisition. This presentation discusses a 1-D array capable of generating multiple simultaneous directional beams in both transmit and receive modes. The array can image over a 50 deg window in a single pulse-echo cycle while requiring as few as four electronic channels for operation. The advantages of this simple approach compared to techniques previously mentioned will be discussed.

2:15

2pSP6. Solution behaviors of multistatic transmission loss equations. Hisashi Shiba (Radio Application Div. NEC Corp., 1-10, Nisshin-Cho, Fuchu, Tokyo 183-8501, Japan, h-shiba@aj.jp.nec.com)

A sonar arrangement is one of the most important problems in the multistatic sonar. Although it is very useful to proceed numerical simulations for arrangements, they usually consume much computing resources and their results are sometimes difficult to grasp physical meanings. On the other hand, it is beneficial to analyze the acoustical field further from the classical sonar equation view point as a quick looking. One of the most important element is the transmission loss in the sonar equation. In the multistatic configurations, the transmission loss is dependent on the direction from a receiver. The transmission loss is usually notified as a function using transmitter parameters. New descriptions are introduced without transmitter parameters in this presentation. Spreading dominant cases produce quartic transmission loss functions of the target distance form the receiver. They are solved analytically; however, the solutions do not constitute Cassini oval in spite of some preceding researches. Absorption dominant cases are also solved analytically. Mixtures of spreading dominant and absorption dominant are not able to be solved analytically. These solution behaviors are easily surveyed by contour maps of the transmission loss. These maps are useful for sonar arrangements.

2:30

2pSP7. Time-varying filter estimation for the deconvolution of environmental reverberation from active sonar returns. Kevin D. LePage and Ryan Goldhahn (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, SP Italy)

The estimation and removal of the time-varying two-way impulse response to environmental scatterers from broadband reverberation data is considered for increasing the signal-to-noise ratio of sonar returns from targets in the water column. Spectrograms of simulated and real reverberation time series data from active sonars in the mid-frequency range show strong evidence of interference patterns which give clues to the number of important paths to environmental scatterers as well as their depth in the water column. In this talk we consider the estimation of a time dependent deconvolution filter for the removal of these environmental reverberation returns from active sonar data. Issues regarding the degrees of freedom required for the efficient implementation of this filter and the stability of these estimates are considered. Simulation results are shown which demonstrate the potential gain of using this approach to partially null the impact of environmental scatterers in active sonar data.

2:45

2pSP8. Near-field blind deconvolution in a reverberant environment. Shima H. Abadi, Eric S. Haapaniemi, Andrew J. Femminineo, Laura M. Williamson, and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109-2133)

Artificial time reversal (ATR) is a passive technique for blind deconvolution in an unknown multipath environment that relies on generic features of underwater sound fields. ATR has been found to be effective when the source is far from the receiving array and the receiving array properly resolves propagating modes or ray-path arrival angles throughout the bandwidth of the source signal. This presentation describes the results of an experimental investigation into ATR's performance for a near-field source in a highly reverberant environment. The experiments were conducted with nominally 0.1 ms pulses at frequencies from 20 kHz to 150 kHz in a 1.0-m-deep and 1.07-m-diameter cylindrical water tank having a reverberation time of ~10 ms using a single sound projector and a linear receiving array of 16 hydrophones. The correlation coefficient between the original and the ATR-reconstructed signals is presented as a function of receiving-array and broadcast-signal characteristics and compared to equivalent signal reconstruction results from spherical-wave delay-and-sum beamforming. The intended application of this research lies in determining the acoustic signature of cavitation bubbles and other hydroacoustic sound sources in hydrodynamic test facilities. [Sponsored by ONR and by NAVSEA through the Naval Engineering Education Center.]

3:00

2pSP9. The influence of suppression of side lobes on the range of passive sonars. Zvonimir Milosic (MORH Sarajevska 7, IROS Ilica 156 b, Zagreb, zvonimir.milosic@morh.hr)

This paper presents a universal procedure of the de-embedding of mathematical functions of ratios of maximal ranges depending on the level of suppression of side lobes at the directivity pattern of passive sonar antenna. This dependence is founded on the specially made and named as "idealized model" of measured directivity pattern of a sonar antenna with the model of elliptical cross section of the main lattice perpendicular on acoustic axes. Using the given mathematical expressions of directivity index and dependence of suppression of side lobes at the mentioned idealized model of directivity pattern characteristics with equations of hydrolocations and their conditions, there are relatively simple expressions of ratio of maximum ranges in form of defined function mz. Using the given mathematical function mz there is possible to control the value of all important parameters for the de-embedding of the sonar ranges in its specification. In accordance with given model, there are point out the importance of ratio of ranges of maximum value of level of suppression of side lobes at directivity pattern characteristic with elliptical cross section of main lattice of antenna. Mainly, the level of suppression of side lobes is less than −50 dB at contemporary passive sonars today.

Session 2pUW

Underwater Acoustics and Acoustical Oceanography: Theory and Practical Applications for Bottom Loss II

Nicholas P. Chotiros, Cochair
Applied Research Lab., Univ. of Texas, P.O. Box 8029, Austin, TX 78713

Martin Siderius, Cochair
Electrical and Computer Engineering, Portland State Univ., 1900 S.W. 4th Ave., Portland, OR 97201

Roger W. Meredith, Cochair
U.S. Oceanographic Office, Stennis Space Center, MS 39529

Invited Papers

1:00

2pUW1. High frequency bottom loss database generation at the naval oceanographic office. Jacob George, David Harvey, and Jorge Novarini (Naval Oceanogr. Office, 1002 Balch Blvd., Stennis Space Ctr., MS)

The Naval Oceanographic Office (NAVOCEANO) generates the low- and high-frequency bottom loss databases (LFBL and HFBL). In support of these upgrades, NAVOCEANO carries out survey operations to make acoustic and geophysical measurements. From these measurements, bottom loss parameters are extracted via numerical inversions. The HFBL database describes acoustic bottom loss over the range 1.5-4 kHz and is based on a set of nine bottom-loss curves derived from the Marine Geological Survey carried out by NAVOCEANO (1965–1968). The optimum curve at a given location is derived from a process of brute force inversion using measured transmission loss (TL) data. Curves are interpolated to the nearest 1/10th; they are then input to the CASS model to determine which curve optimizes the fit between the CASS prediction and the measured TL data. Optimal curves are determined independently at each of the 1/3 octave frequencies over the HFBL range. Studies of the derived curve values in a recent upgrade area have indicated very high variability. This variability does not tend to correlate with bottom sediment type, and the variability tends to be highest in the higher frequency bands. An implication is that scattering caused by roughness at the water/sediment interface may be driving variability.

1:20

2pUW2. Prediction of marine fine-grain sediment states: Determinants of mine burial and acoustic impedance. Richard H. Bennett (SEAPROBE, Inc., 501 Pine St., Picayune, MS 39466), Conrad W. Curry, Roger W. Meredith (Stennis Space Ctr., MS 39529), and Richard W. Faas (Univ. of Southern MS, MS 39529)

The predicted depth of mines buried in marine muds is generally based on estimates of sediment shear strength (often unreliable). Conversely, sediment states of marine muds are water-dependent, defined empirically by the Atterberg Limits (liquid limit and plastic limit), and allow the sediment to be described as having fluid-like, plastic-like, or semi-solid consistency. When the natural water content and the liquid limit of normally and unconsolidated marine muds are approximately equal at depth below the seafloor, the mud at greater depth is considered to no longer behave as fluid-like, but plastic-like. This relationship provides a predictable conservative minimum mine burial penetration depth. Mine burial depths at two sites were shown to closely agree with predicted burial depths based on the natural water contents and the liquid limits (Bennett *et al.*, SEAPROBE, Inc., Technical Report Number SI-0004-01, p., 89, 2004, funded by ONR). Prediction of selected sediment physical properties using acoustic impedance as a function of depth below seafloor may provide a method to estimate and evaluate sediment states. Comparison of subbottom natural water contents with a database showing known liquid limits for different types of marine muds should make possible prediction of conservative, minimum, mine burial depths.

1:40

2pUW3. Experience with geoacoustic inversion of transmission loss. Paul J. Vidmar (SAIC, 4001 N. Fairfax Dr., Arlington, VA 22203)

This presentation will discuss recent experience with geoacoustic inversion of transmission loss (TL) data from 50 to 5000 Hz from both deep and shallow water regions. For shallow water, a multi-layered geoacoustic profile with range dependent layer thicknesses is used. Layer thickness is not an inversion parameter but is known from ancillary data such as seismic profiling and chirp sonar. Issues related to range dependent sound speed profiles (SVPs) will be discussed as will observation of the time spread of multipath arrivals. For deep water, a single layer geoacoustic profile is assumed. Quality of inversion is assessed using comparison of measured and modeled TL and the estimation error. The estimation error is the dependence of the total rms error—summed over range, frequency, and receivers—on the values of a geoacoustic parameter within the bounds used to constrain inversion. The estimation error identifies geoacoustic parameters that are well constrained or poorly constrained by the TL data. Inversion was carried out using a genetic algorithm model [D. Harvey *et al.*, Oceans '02 MTS/IEEE, **1**, 358–362 (2002)]. Acoustic and environmental data were provided by the Naval Oceanographic Office. [Work funded by the Ocean Bottom Characterization Initiative through PEO C4I, PWM 120.]

2:00

2pUW4. Relating volume scattering from the seafloor to dispersion and attenuation within the seafloor. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Heterogeneities in ocean sediments can produce significant scattering of sound from the seafloor, particularly in soft sediments. First-order volume perturbation models for scattering from the seafloor into the water ignore the effect of this scattering on propagation and dispersion within the sediment. The energy that is scattered, however, contributes to the attenuation of the penetrating sound. This increase in attenuation, and its effect on dispersion, is modeled by applying perturbation theory to sound propagation through a fluid sediment. This allows the application and predictions of this propagation model to be connected directly to previous perturbation approximations of scattering from the sediment. Several sediments for which volume scattering has been previously studied are revisited in the context of sound propagation. The implications of this loss mechanism within the sediment for the scattering of sound from the sediment are also considered. [Work supported by the Office of Naval Research.]

Contributed Papers

2:20

2pUW5. Comparison of parabolic equation and coupled mode solutions to seismo-acoustic problems. Scott D. Frank (Dept. of Mathematics, Marist College, 3399 North Ave., Poughkeepsie NY 12601, scott.frank@marist.edu), Robert I. Odom (Univ. of Washington, Seattle, WA 98105), Minkyu Park (Korea Polar Res. Inst., Incheon, 406-840, Korea), and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Parabolic equation methods that use the single scattering approximation and improved rotated coordinate methods generate accurate and efficient solutions for range-dependent underwater acoustic problems with elastic sediments. Recently these methods have demonstrated the conversion of acoustic energy in a fluid into shear propagation in an underlying elastic layer. Elastic coupled mode theory can also be used in a range-dependent environment and provides an accurate description of the conversion between fluid and elastic modes during propagation. The current parabolic equation approach will be augmented with an elastic version of the self-starter that includes both compressional and shear wave energy. Results from these two approaches will be compared for acoustic and elastic sources in both range-independent and range-dependent underwater environments. Propagation at the fluid–solid interface will be examined as a possible mechanism for the conversion of elastic layer shear energy into acoustic energy in the water column. An elastic source is used to demonstrate that sources of this type can transmit substantial acoustic energy into the water column. [Work supported by ONR.]

2:35

2pUW6. Initial assessment of combustive sound source arrays as airgun alternatives for Arctic under-ice seismic exploration. Juan I. Arvelo, >Jerrold Dietz (Appl. Phys. Lab., The Johns Hopkins Univ. 11100 Johns Hopkins Rd., Laurel, MD 20723-6099), Andrew R. McNeese, Jason D. Sagers, and Preston S. Wilson (The Univ. of Texas at Austin, Austin, TX 78712-0292)

Combustive sound source arrays consisting of submersible combustion chambers filled with a hydrogen/oxygen mixture are employed to assess their effectiveness for seismic exploration applications. The combustive mixture is ignited via spark and radiates acoustic pulses capable of undersea deep sub-bottom sediment penetration. Since electrolytic cells may be employed to generate the hydrogen/oxygen mixture from surrounding seawater, this source is an attractive alternative to airgun arrays for Arctic seismic exploration from under-ice platforms. Combustive sound source array configurations were tested in a central Virginia basin with hydrophones deployed in a line at another nearby basin. Seismic reflections are compared against nearby geologic cross sections of the central Virginia seismic zone. [Funding provided via UAF sub-award under NOAA Grant NA09NOS4000262 and ONR.]

2:50–3:05 Break

3:05

2pUW7. Efficient parabolic equation modeling with shear (SCRAM). Richard L. Campbell, Kevin D. Heaney (Ocean Acoust. and Instrumentation Systems, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, rlcamp.pdx@gmail.com), Alec J. Duncan, and Robert D. McCauley (Curtin Univ., Western Australia)

A novel parabolic equation model has been developed based upon Michael Collins RAM algorithm. By separating the PE solver from the fixed range-depth slice geometry of RAM, efficiency is significantly improved for full-field $\times$ propagation to points on a plan-view grid or along an arbitrary vessel track. The code is written in C with the ability to fully leverage modern many-core computers. In this paper, the SCRAM model is introduced, based on the PE propagator from Collins RAMS variant, extending CRAM to environments with sediments supporting shear wave propagation. Comparison with range-independent wave-integral solutions will be made. Specific application to the problem of propagation over continental shelves with calcerenite seabeds, such as those observed in the seas off the coast of Australia will be examined. Model-data comparisons for measurements taken near Dogan, Australia, will be made.

3:20

2pUW8. The depth dependence of earthquake T-phases at an ocean acoustic observatory. Ralph A. Stephen, S. Thompson Bolmer (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543-1542), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0225), James A. Mercer (Univ. of Washington, Seattle, WA 98105-6698), and Bruce M. Howe (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

T-phases are earthquake signals that have propagated, at least partially, in the ocean sound channel. T-phase hydrophone networks detect much smaller earthquakes over basin scales than land-based networks and they detect many more earthquakes than comparable regional scale seismic land networks. Furthermore, since T-phases travel at lower velocities than seismic phases, they result in much more precise locations of events given the same timing accuracy. T-phases are typically spread over 10's of seconds, and a common problem, however, is precisely identifying the arrival time of an event. T-phase stations usually consist of single hydrophones moored near the sound channel axis and the depth dependence of the T-phase envelope and frequency content is rarely studied. In the North Pacific Ocean, from 2004 to 2005, ambient noise and earthquakes were observed at an ocean acoustic observatory consisting of a vertical hydrophone array (from about 750 m above the seafloor to 375 m from the surface) and three co-located ocean bottom seismometers. This data set provides a unique

opportunity to observe earthquake signals and their characteristics throughout the water column and to provide ground-truth to theoretical predictions on the excitation and propagation mechanisms of T-phases. In at least one case, a T-phases from a distant earthquake was readily observed even at the seafloor, well-below the conjugate depth.

3:35

2pUW9. Time-evolving T-phase arrival structure using simultaneous recordings by large-aperture horizontal and vertical line arrays in PhilSea09. Simon E Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, sfreeman@ucsd.edu), Gerald L. D'Spain (Scripps Inst. of Oceanogr., San Diego, CA 92106), Ralph Steven (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Kevin D. Heany (Oasis Inc., Lexington, MA 02421), Arthur Baggeroer (MIT, Cambridge, MA 02139), Peter Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92037), Jim Mercer (Univ. of Washington, Seattle, WA), Stephen Lynch (Scripps Inst. of Oceanogr., San Diego, CA 92106), and Jim Murray (Oasis Inc., Lexington, MA 02421)

A number of models have been proposed to explain the mechanisms by which seismic phases couple to the deep ocean sound channel in order to create water-borne acoustic tertiary (T) phases. Beamforming conducted on simultaneous recordings by large-aperture horizontal towed and vertical moored line arrays during PhilSea09 shows the temporal evolution of a T-phase arrival consistent with the down-slope modal conversion/propagation model. Towed array calibration is conducted using ship-deployed, controlled multi-tone acoustic sources. Conventional, minimum variance distortionless response, white noise constrained, and dominant mode rejection beamformers are compared in their ability to minimize bias and variance in estimating the azimuthal arrival directions of signals from both the controlled source and the seismic phases recorded by the horizontal array. Horizontal array beamformer-derived azimuth and time-of-arrival range estimates from P, S, and T-phase arrivals at towed and moored receivers indicate the event occurred in a region with appropriate bathymetric relief for down-slope conversion/propagation. The seismic event in question was not recorded by the USGS/NEIC seismometer network. This study thus further showcases the highly sensitive capabilities of in-water hydrophones and the effect of array gain to characterize high-frequency (5–50 Hz) seismic events. [Work supported by the Office of Naval Research.]

3:50

2pUW10. Seismic tremor event intervals from dual-frequency coherence. LeRoy M. Dorman (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA, ldorman@ucsd.edu), Susan Y. Schwartz (Earth & Planetary Sci., UCSC, Santa Cruz, CA 95064), and Michael Tryon (SIO, UCSD, 92093-0220)

Slip occurring at plate boundaries creates seismic tremor as well as "normal" earthquakes. This nonvolcanic tremor appears to consist of swarms of low-frequency earthquakes which lack impulsive P and S arrivals. Tremor is accompanied by slip observed by GPS and can show anomalies in fluid flow. The seismic radiation resembles continuous microseismic noise more than discrete events. We report dual-frequency coherence (DFC) calculations on tremor and normal microseismic background noise observed on Ocean-Bottom Seismographs and land seismic stations around the Nicoya Peninsula, Costa Rica. Both the OBS and land tremor signals show a banded pattern in DFC that is absent in normal noise. The similarity in the DFC patterns between OBS and land tremor signals suggests a common source, eliminating the possibility that DFC is a property of the OBS or seafloor environment. Banded DFC patterns can be generated by repeated events with a repeat time equal to the reciprocal of the offset frequency between bands. If, as is becoming widely accepted, nonvolcanic tremor consists of swarms of low frequency earthquakes (LFEs), DFC analysis may help to reveal LFE periodicities or intervals.

4:05–5:05 Panel Discussion

TUESDAY AFTERNOON, 1 NOVEMBER 2011 — CRESCENT, 1:00 TO 2:00 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

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

R. J. Peppin, Vice Chair ASC S1
Scantek, Inc., 6430 Dobbin Rd., #C, Columbia MD 21045

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, November 1, 2011.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Meeting of Accredited Standards Committee (ASC) S12 Noise

W. J. Murphy, Chair, ASC S12
NIOSH, 4676 Columbia Pkwy., Mail Stop C27, Cincinnati, OH 45226

R. D. Hellweg, Vice Chair, ASC S12
Hellweg Acoustics, 13 Pine Tree Rd., Wellesley, MA 02482

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, November 1, 2011.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

TUESDAY AFTERNOON, 1 NOVEMBER 2011 CRESCENT, 3:45 TO 5:00 P.M.

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

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

M. C. Hastings, Vice Chair ASC S3/SC 1
Georgia Institute of Technology, G.W. Woodruff School of Mechanical Engineering
126 Love Bldg., 771 Ferst Dr., Atlanta GA 30332 0405

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings beginning at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday are as follows:

Acoustical Oceanography	Pacific Salon 2
Architectural Acoustics	Sunrise
Engineering Acoustics	Pacific Salon 6/7
Musical Acoustics	Towne
Physical Acoustics	Royal Palm 3/4
Psychological and Physiological Acoustics	Esquire
Structural Acoustics and Vibration	Sunset

Session 3aAA

Architectural Acoustics and Musical Acoustics: Variable Acoustics on Concert Stages I

Bill Dohn, Cochair
Dohn and Associates, Inc., 630 Quintana Rd., Morro Bay, CA 93442

Michelle C. Vigeant, Cochair
Mechanical Engineering, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117-1599

Chair's Introduction—8:00

Invited Papers

8:05

3aAA1. Orchestra canopies. Magne Skalevik (AKUTEK/BrekkeStrand, Bolstadtunet 7, 3430 Spikkestad, Norway, magne.skalevik@brekkestrand.no)

Looking around, one might get the impression that a canopy has become a "must have" in every new concert hall and a magic recipe in any refurbishment project. There are, however, cases where canopies would be unwanted, unnecessary, insufficient, inadequate, or even detrimental to the acoustics on stage or in the rest of the hall. Some of the highest rated concert halls among musicians and audience, Musikvereinsaal in Vienna, Concertgebouw in Amsterdam, and Boston Symphony Hall, all fail to have a canopy. Still, there are many halls where the use of a canopy is justified. This paper points at the importance of starting by defining the requirements to the acoustics on stage (and in the rest of the hall), and then the question how these requirements could be met. A canopy may be the answer, maybe not. Significant features, acoustic properties, design issues, and parameters for prediction and measurements of canopy performance will be discussed. In short: Orchestra canopies—what difference can they make?

8:25

3aAA2. A new active acoustics system for enhancing musician's environment on concert and recording stages. Wieslaw Woszczyk (McGill Univ., CIRMMT and Schulich School of Music, 555 Sherbrooke St., West, Montreal, QC H3A 1E3, Canada, wieslaw@music.mcgill.ca), Doyuen Ko, and Jonathan Hong (McGill Univ., Montreal, QC H3A 1E3, Canada)

The necessity of variable acoustics in modern performance spaces is widely acknowledged; particularly, there has been a growing interest in active acoustics enhancement systems. Many different systems have been developed and installed in concert halls, public venues, and research facilities worldwide. However, further improvements are required to provide musicians with a desirable acoustic on-stage support equal in quality to the best rooms. An innovative electro-acoustic enhancement system, based on measured high-resolution impulse responses, is developed at the Virtual Acoustics Technology (VAT) lab, in McGill University. The first model is installed in Multi-Media Room (MMR), a large rectangular space (80 ft. $\times$ 60 ft. $\times$ 50 ft.) designed as a film scoring stage. The system uses an array of omni-directional radiators and its signal processing includes 24-channel low-latency convolution at 24 bit/96 KHz, three separate stages of matrix mixing, equalization, and time variation. The objective measurement of the system adjusted to improve on-stage acoustics verifies the enhancement of parameters such as inter-aural cross-correlation coefficient (IACC), early stage support (ST1), and early decay time (EDT) without excessively increasing the overall acoustic energy in the room. The subjective evaluations collected from multiple recording sessions with professional musicians using the system will be presented. [Work supported by NSERC, JMP.]

8:45

3aAA3. Orchestra shell innovation: Vented shell case study. Carl Giegold, Scott Pfeiffer, Dawn Schuette, and Molly Norris (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815 Chicago, IL 60604)

The College Community School District music programs required a venue larger and more conducive to band and choral performance than their 1970s 400-seat theater. The new venue, located in Cedar Rapids, Iowa, seats 1000 in a space designed primarily for music but with theatrical capability as well. An adjustable orchestra shell and variable absorption tailor the room's acoustic response to speech, theater, jazz, symphonic band, choral, and orchestra. Though it must accommodate bands of up to 140 players for audiences of up to a thousand, the hall must also be responsive to small ensembles and intimate to audiences of only a few hundred. To control loudness for the largest bands, the orchestra shell can be vented to th flytower. The design will be presented, as well as measurement data and anecdotal impressions of the shell.

9:05

3aAA4. Venting full-stage portable acoustical shells: The acoustic impact on the stage and audience environments. Ron Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

The effectiveness of acoustical shells in the stage acoustic environment has long been a topic of discussion. Generally, their impact is positive with increased gain (G) both on stage and to the audience area. However, there has also been speculation that containing too much energy on stage negatively impacts clarity (C80). A study was conducted of an installed portable shell that provided an option to

vent energy to the backstage area, thus providing the potential to reduce the overall energy in the stage area. Impulse responses were measured in a number of areas on stage and in the audience area, with vents both open and closed. The results of those metrics will be presented. Binaural recordings were also made at the same locations, providing perceptual information as well.

9:25

3aAA5. Fine-tuning onstage acoustical elements to satisfy... Whom? J. Christopher Jaffe, Jonah Sacks, Benjamin E. Markham, and Robert S. Berens (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

Movable onstage features such as over-stage reflector panels, acoustical shell towers, reverberant chamber doors, seating risers, etc. offer a high degree of variability and control over ensemble balance, loudness, and other parameters. Finding the optimal settings for such features is far from trivial and combines many different considerations: quality of sound to the various audience seating sections, balance and intelligibility of the ensemble to the conductor, self and ensemble hearing for the various instrumental sections, and overall loudness onstage, to name a few. On the occasion of welcoming Maestro Christoph Eschenbach as its new Music Director, the National Symphony Orchestra scheduled a special acoustical rehearsal for the purpose of exploring the range of available settings of the adjustable over-stage reflector panels at the Kennedy Center Concert Hall. The presentation will include a detailed description of the rehearsal and the adjustments made to the reflectors, the authors' observations and findings, and various conclusions that have been drawn. Particular attention will be paid to the range of reactions and feedback from various participants, including the Maestro, musicians in different sections on stage, symphony administrators and musicians listening from the audience, and the authors.

9:45

3aAA6. Active acoustics for concert stages. Roger W. Schwenke (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702)

Orchestra shells are often loud places. So much so that some orchestras have instituted noise monitoring and select their repertoire to limit noise exposure. Orchestra shells are loud places because they are surrounded on five out of six sides by nearby hard reflecting surfaces. This also means that the orchestra only experiences reverberation coming from the direction of the house. Active acoustics offers an alternative to physical orchestra shells, which is more light-weight, and offers more flexible acoustical performance including enveloping reverberation for performers.

10:05–10:20 Break

10:20

3aAA7. A survey of selected 800 to 1800-seat multipurpose hall orchestra shells and eyebrows by McKay Conant Hoover Inc. David A. Conant and William Chu (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Westlake Village, CA 91362, DConant@MCHinc.com)

New and renovated multipurpose halls by MCH Inc. are examined with focus on the room-acoustical design of sound to and among on-stage performers as well as, necessarily, acoustical considerations associated with direction of that sound to audience. Specifically, orchestra shell/recital screen design as well as integrated eyebrow/canopy design is addressed. Halls discussed include Kavli Theater (Thousand Oaks Civic Arts Plaza), Jackson Hall (Mondavi Center for Performing Arts), Ikeda Theater (Mesa Arts Center), Granada Theatre Restoration (Santa Barbara), Balboa Theatre Restoration (San Diego), Scottsdale Center for Performing Arts and the new Valley Performing Arts Center (Los Angeles).

10:40

3aAA8. The use of multi-channel microphone and loudspeaker arrays to evaluate room acoustics. Samuel W. Clapp, Anne E. Guthrie, Jonas Braasch, and Ning Xiang (GraduateProgram in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180, clapps@rpi.edu)

Most room acoustic parameters are calculated with data from omni-directional or figure-of-eight microphones. Using a spherical microphone array to record room impulse responses can yield more information about the spatial characteristics of the sound field, including spatial uniformity and the directions of individual reflections. In this research a spherical array was use to measure room impulse responses in a wide variety of concert halls throughout New York State, both on stage and in the audience, with both the microphone array and a dummy head. The results were analyzed using beamforming techniques to determine spatial information about the sound field, and compared to the results of geometrical acoustics and binaural localization models. Of particular interest was how the spatial data can help to differentiate between different spaces or listener positions that exhibit similar values for conventional metrics. Auralizations were created using both headphone playback and second-order ambisonic playback via a loudspeaker array. These systems were evaluated objectively to compare the reproduction systems with the measured data. Subjects were recruited for listening tests using each reproduction method, and asked to evaluate the halls on both objective measures and subjective preference, and the results of binaural and ambisonic playback were compared.

11:00

3aAA9. Using a spherical microphone array to analyze concert stage acoustics. Terence Caulkins (Arup, 155 Ave. of the Americas, New York, NY 10013, terence.caulkins@arup.com), Anne Guthrie (Arup Rensselaer Polytechnic Inst., NY 12180), Sam Clapp, Jonas Braasch, and Ning Xiang (Rensselaer Polytechnic Inst., NY 12180)

A 16 channel spherical microphone array has been designed and constructed to measure high order Ambisonic sound fields, based on equations for scattering off of a rigid sphere. This microphone has been used to capture and characterize the acoustics of nine different concert hall stages around the state of New York. Musicians have been invited to perform in real-time in auralizations of each hall and participate in a subjective preference test based on their experience. The preference tests have been analyzed using multi-dimensional scaling methods and compared with acoustical data derived from a beamforming analysis of each stage. Additional comparisons with A.C. Gade's omnidirectional parameters (stage support, early ensemble level) and geometric parameters described by J. Dammerud are performed.

11:20

3aAA10. Investigations of stage acoustics at the Sydney Opera House Concert Hall. Timothy E. Gulsrud (Kirkegaard Assoc., 954 Pearl St., Boulder, CO, tgulsrud@kirkegaard.com)

Improvements to onstage hearing conditions were made at the Sydney Opera House Concert Hall during a series of acoustics trials conducted during September 2009. During the acoustic trials, physical changes were made to reflective surfaces around the platform and an active acoustics system was demonstrated. Taken together, the temporary changes had a positive influence on hearing conditions during both rehearsals and performances by the Sydney Symphony Orchestra and Sydney Philharmonia. This paper reviews the strategies used to implement the improvements to stage acoustics, and discusses the unique techniques used to evaluate the stage acoustics both subjectively and objectively.

11:40

3aAA11. Acoustical renovation of Hahn Recital Hall at the Music Academy of the West, Montecito, CA. William Chu and David A. Conant (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Westlake Village, CA 91362, WChu@MCHinc.com)

A 350-seat fixed-acoustics recital hall believed designed by Vern Knudsen in 1971 was substantially renovated in 2008 to meet the current needs of the revered Music Academy of the West. The preponderance of its program was to serve Marilyn Horne's operatic program as well as chamber music but required as well to provide suitable space for occasional full orchestra rehearsals and performance. The renovation design is described as it developed from initial acoustical measurements to the novel platform acoustics and other adjustments that permit optimizing the large platform for multiple needs. Specifics of the acoustical coupling characteristics of the custom, sound transparent/sound reflective and diffusive recital screen, its overhead reflectors and absorptive drapery are discussed as well as accommodation for Met Live presentations.

WEDNESDAY MORNING, 2 NOVEMBER 2011 PACIFIC SALON 1, 8:00 A.M. TO 12:00 NOON

Session 3aAB

Animal Bioacoustics: Acoustics for Saving Endangered Species I

Jay Barlow, Cochair
NOAA Southwest Fisheries Science Center, 3333 N. Torrey Pines Ct., La Jolla, CA 92037

Sofie M. Van Parijs, Cochair
Northeast Fisheries Science Ctr., 166 Water St., Woods Hole, MA 02543-1026

Chair's Introduction—8:00

Invited Papers

8:05

3aAB1. Automatically identifying rare sounds of interest in environments cluttered with biological homophones. Kurt M. Fristrup (Natural Sound and Night Sky Div., Natl. Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

Successful acoustical monitoring for threatened or endangered species must surmount challenges of adequate spatial and temporal coverage in the data collection phase, and efficient and effective processing in the data analysis phase. For terrestrial environments, a diverse array of digital recording options has relaxed the difficulty of obtaining sufficient coverage. However, this capability has amplified the requirement for efficient processing. In many terrestrial environments, the principal processing challenge is distinguishing the sounds of rare species from many other sounds that are similar in time-frequency structure. Some of these biological homophones are generated by species that are much more numerous than the target. Solutions to these problems will be assessed in the context of large scale projects that focused on bird species of special interest.

8:25

3aAB2. Monitoring a small marine cetacean population using passive acoustic techniques: Vaquita, a critically endangered species. Armando Jaramillo-Legorreta and Lorenzo Rojas-Bracho (Instituto Nacional de Ecologia, DGIOECE–Marine Mammals Dept., CICESE, Ensenada, B.C., Mexico)

In 2008, vaquita population was estimated in only 245 individuals. Between 1997 and 2007, a passive acoustic semi-autonomous system was used for monitoring detection rate of this species. The analysis resulted in decline of approximately 58%. Sighting and acoustic data from a 2008 research cruise, compared to the 1997 estimate of abundance, resulted in a decline of approximately 57%. Hence, passive acoustic detection probed reliable to monitor the population. The Mexican Government implemented a recovery plan, which includes monitoring population trends. At current level, the population can grow at maximum rates lower than 4% annually. The severely reduced population level, and scarcity of acoustic detections, made unreliable to continue the use of the methods applied till 2007. An increase of sampling effort was identified as the key to implement a reliable system, only achievable using completely autonomous detectors. Field test of autonomous detectors identified C-POD as very reliable. Using data collected with this equipment it was determined that an annual effort of 5000 C-POD days is needed to detect population increase. A sampling grid of 62 stations was designed. Work to design and test reliable mooring devices was done. Currently, the first of five years of sampling has started.

8:45

3aAB3. Statistical design for acoustic monitoring to detect declines of endangered species and prevent extinction. Jay Barlow (NOAA Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Ct., La Jolla, CA 92037), Armando Jaramillo-Legorreta (Instituto Nacional de Ecologia, c/o CICESE, Km. 107 Carretera Ensenada-Tijuana, Ensenada, BC 22860, Mexico), Barbara L. Taylor (NOAA Southwest Fisheries Sci. Ctr., La Jolla, CA 92037), Lorenzo Rojas-Bracho (Instituto Nacional de Ecologia, *c/o CICESE*, Km. 107 Carretera Ensenada-Tijuana, Ensenada, BC 22860, Mexico), and Nicholas J. C. Tregenza (Beach Cottage Ltd., Long Rock, Cornwall, TR20 8JE, United Kingdom)

As populations become smaller and more endangered, the ability to monitor trends in their abundance also decreases. For vocal species, passive acoustic monitoring can provide a powerful, cost-effective method of monitoring relative abundance. However, the monitoring effort should be based on a statistical design that can detect population declines in time to prevent extinction. This is illustrated with an example of the vaquita (Phocoena sinus) an endangered porpoise in the northern Gulf of California, Mexico. A pilot project showed that porpoise detectors (C-PODs) recorded echo-location clicks from approximately one group of porpoises every two days. Based on this, we estimate that 5000 days of C-POD monitoring per year would be needed to obtain a measure of relative vaquita abundance with a coefficient of variation (CV) of 3%. A power analysis shows that five years of monitoring with this CV would give a high probability of detecting a 5% annual increase or decrease in population size. Visual sighting methods could not detect such small changes with any conceivable level of survey effort. This kind of innovative monitoring is a critical tool in the continuing evaluation of conservation measures.

Contributed Papers

9:05

3aAB4. Towed hydrophone surveys for monitoring trends in the abundance and distribution of the critically endangered Gulf of California porpoise, Vaquita. Rene Swift (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, Fife, KY16 8LB, United Kingdom), Shannon Rankin, Tim Gerrodette, Barbara Taylor (Southwest Fisheries Sci. Ctr., La Jolla, CA 92037), Douglas Gillespie, Jonathan Gordon (Univ. of St. Andrews, St. Andrews, Fife KY16 8LB, United Kingdom), Lorenzo Rojas-Bracho, and Armando Jaramillo-Legorreta (Inst. Nacional de Ecologia/CICESE, Ensenada, B.C. 22860, Mexico)

The vaquita (*Phocoena sinus*) is a critically endangered small cetacean found only in the upper Gulf of California, where fisheries bycatch remains an acute threat. Cost, shallow heavily fished areas, and the vaquitas extreme avoidance of noisy motorized vessels argue against using large vessels typically used for visual line transect surveys. Towed hydrophone surveys, using *Rainbow-Click* semi-automatic detection software, were carried out from a 24' sailing trimaran in autumn 2008. Ultrasonic ($\sim$130 kHz) vaquita echolocation clicks were reliably detected and tracked using classification parameters developed for harbor porpoise. Transects were sailed on 49% of days and 31 groups were detected within the vaquitas known range and in areas not easily surveyed using traditional methods. Although very high levels of ambient noise presented challenges for acoustic monitoring, perpendicular distances were calculated to 30 groups giving an estimated strip half-width of 198 m. The detection algorithm has since been implemented in PAMGUARD software and significantly improved using survey data. Shallow, heavily fished areas remain difficult for estimating and monitoring trends in abundance. Towed arrays proved effective for the former and may remain the only alternative for the latter. Precision is likely to remain low for the quick detection of small rates of increase.

9:20

3aAB5. Acoustic detection of the Ivory-billed Woodpecker (*Campephilus principalis*). Michael D. Collins (P.O. Box 1975, Pearl River, LA 70452, mike@fishcrow.com)

There have recently been independent reports of multiple sightings and auditory detections of the Ivory-billed Woodpecker (*Campephilus principalis*) in remote swamps in Arkansas, Florida, and Louisiana. Putative audio and video recordings have been obtained at each site, but no clear image of this critically endangered (and extremely elusive) species has been obtained in decades. Acoustic detection is possible by "kent" calls, alarm calls, double knocks, and pounding sounds associated with foraging and cavity construction, but each of these clues is problematic. Kents are difficult to hear in the distance—the author did not detect a long series of kents before drifting in a kayak to well within 100 m of the bird. This species does not drum like other woodpeckers, and it is believed to forage relatively quiet (at least part of the time) by using its massive bill to pry bark loose. The Ivory-billed Woodpecker is not one of those species that is easier heard than seen—during 5 yr of fieldwork, the author had ten definite sightings but only two definite auditory detections. These issues and the current status of the fieldwork in the Pearl River basin in Louisiana will be discussed.

9:35

3aAB6. Acoustic risk factors and passive acoustic monitoring of endangered southern resident killer whales (*Orcinus orca*). Marla M. Holt, Candice K. Emmons, M. Bradley Hanson, and Dawn P. Noren (Marine Mammal Ecology Team, NOAA Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. East, Seattle, WA 98112)

Southern resident killer whales (SRKWs) occur along the coastal and inland waters of the northeast Pacific Ocean. They are currently listed as endangered in both the U.S. and Canada. Risk factors that are potentially affecting population recovery include prey availability and quality and acoustic disturbance. This is because SRKWs specialize on Chinook salmon, of which many stocks are depleted, and there is a well developed whale-watching industry that focuses on viewing SRKWs in their core summer habitat. This review paper will highlight acoustic research conducted to address risk factors, recovery goals, and other conservation considerations of this endangered population. These include investigations on acoustic and behavioral responses to anthropogenic sounds. A previous study demonstrated amplitude compensation as vessel noise increased and an ongoing study is investigating potential acoustic effects on behavior, including foraging, using suction cup attached digital acoustic recorders (DTAGs). This paper will also discuss passive acoustic monitoring efforts involving coastal ship surveys and acoustic recorder deployments which aim to better characterize critical habitat, particularly in areas and during seasons when SRKW occurrence is less well defined. All of these investigations are designed to provide critical data necessary to address and refine recovery goals and management actions for SRKWs.

9:50

3aAB7. Passive acoustic recording to build acoustic catalogs to remotely monitor resident individuals and the health of the dolphin population in the Indian River Lagoon system in Florida. Edmund Gerstein (Dept. of Psych., Charles E. Schmidt College of Sci. Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431, gerstein2@aol.com), Beth Brady (Nova Southeastern Univ., Dania Beach, FL 33004), Rebecca Weeks (Florida Atlantic Univ., Dania Beach, FL 33004), Gregory Bossart (Georgia Aquarium, Atlanta, GA 30313), Juli Goldstein, and Stephen McCulloch (Florida Atlantic Univ., Ft. Pierce, FL 34946)

Harbor Branch Oceanographic Institute at Florida Atlantic University together with, NOAA's National Ocean Service/Center for Coastal Environmental Health & Biomolecular Research initiated a research program designed to assess environmental and anthropogenic stressors that may affect the health and long-term viability of bottlenose dolphin populations inhabiting coastal regions of Florida and South Carolina This collaborative program is known as the Health and Environmental Risk Assessment Project. The project involves the capture, sampling, and release of selected wild dolphin stocks to allow comprehensive health screenings by collecting and analyzing a variety of biomedical samples and associated data. During Indian River Lagoon (IRL) dolphin population assessment, the acoustic behavior of 33 individual dolphins was recorded using synchronized DARP buoys configured with directional and omnidirectional hydrophones. Vocalizations were recorded during the capture, holding and health sampling phases. Acoustic levels

and behavior during ABR hearing measurements were also monitored. The whistle contours of 23 individuals have been identified and will be added to the photo-identification and genetic catalog maintained for the IRL population. These signature whistles will be used to train a network of remote acoustic sensors to monitor the distribution, social interactions, and habitat utilization of cataloged dolphins in the IRL. Authorized NMFS permit 14352-01.

10:05–10:25 Break

Invited Paper

10:25

3aAB8. Passive acoustic methods for surveying visually cryptic frog species. David L. Borchers (Dept. Maths & Stats., Univ. of St. Andrews, CREEM, The Observatory, Buchannan Gdns, St. Andrews, KY16 9LZ, Scotland, dlb@st-andrews.ac.uk), John G. Measey, and Res Altwegg (South African Natl. Biodiversity Inst., Cape Town, South Africa)

Passive acoustic survey methods have great potential for assessing cryptic species that are vocally active. The tiny Moss Frogs (genus *Arthroleptella*) that inhabit seepages on remote mountain tops in the Western Cape of South Africa are a case in point. Many species are restricted to individual mountains and most are on the IUCN Red List. Surveys are prohibitively expensive because each individual capture involves three to four person-hours of searching. However, males can be heard calling throughout the winter. Using a portable recorder, we gathered acoustic data from which abundance can be estimated and populations monitored. This involves estimation of both the spatial "footprint" of the acoustic array (i.e., what area it effectively surveys) and animal vocalization rates. Ideally both should be estimated as an integral part of the survey, rather than obtained from outside the survey. By combining recently developed mark-recapture methods that take explicit account of the spatial location of microphones, with data on acoustic signal strength and/or time-difference-of-arrival at different microphones, it is possible to do this. We describe the survey and estimation methods, which have many other potential applications. In the case of these frogs, acoustic survey is hundreds of times more efficient than methods involving physical capture.

Contributed Papers

10:45

3aAB9. Interannual temporal and spatial distribution of bowhead whales in the Western Alaskan Beaufort Sea; 2007–2010. Stephanie L. Grassia, Catherine L. Berchok (NOAA/Natl. Marine Mammal Lab., 7600 Sand Point Way NE, Seattle, WA 98115), and Dana L. Wright (School of Marine Sci., Univ. of Maine, 5706 Aubert Hall, Orono, ME 04469)

Passive acoustic monitoring began offshore of Barrow, Alaska, in 2007 as part of the interdisciplinary BOWFEST (Bowhead Feeding Ecology Study) project. This study area spans from the northern coast of Alaska out to 72°N and between 152° and 157°W. To better understand how bowhead whales use this area, long-term (year-long) and shorter-term (week- to month-long) autonomous passive acoustic recorders were deployed on subsurface moorings. The long-term recorders were deployed at four locations along the 100 m isobath off Barrow, running on a 30%–45% duty cycle at a sampling rate of 8192 Hz. The shorter-term recorders were deployed inshore (<20 m water depth) and sampled at either 12.5 or 40 kHz on a 80%–90% duty cycle (although one sampled continuously at 8192 Hz). Since 2007 we have obtained 18 820 h of recordings from the long-term and 3780 h of recordings from the short-term moorings. All recordings were reviewed manually, with the long-term data analyzed on a 3-h time interval and short-term data analyzed fully. The majority of the bowhead call detections were between April and November. Very few detections were made during the winter months. Comparisons between the offshore recorders and inshore recorders (which were within the main migration corridor) will be discussed.

11:00

3aAB10. Acoustics as a tool in sub-species and population identification for endangered fin whales, *Balaenoptera physalus*. Shannon Rankin, Jay Barlow, Eric Archer (Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Ct., La Jolla, CA 92037, shannon.rankin@noaa.gov), and Benjamin Jones (Tulane Univ., New Orleans LA 70118)

Identification of “stocks” (sub-species and independent populations) is important for understanding and mitigating potential sources of human-caused mortality. This is especially critical for endangered and protected species, such as the large whales. Stock identification for whales has typically been based on ecology, life history, morphology, and genetics. However, for many species, acoustic differences in whale call types may indicate population or sub-species structure. The potential role of acoustics in identifying species and sub-species has been identified in numerous publications; however, this role has yet to be realized for large whales. In an effort to include acoustic data in this process, we are contributing to current efforts to update the status of endangered fin whales, *Balaenoptera physalus*, in the North Pacific. An analysis of North Pacific fin whale populations based on identification of “song” provides hypotheses that can be tested with genetics. Strengths and limitations of acoustic methods will be presented, as will the potential for collaboration on the scale of ocean basins.

11:15

3aAB11. New call type produced by the North Pacific right whale, *Eubalaena japonica*. Jessica L. Crance, Catherine L. Berchok, and Phillip J. Clapham (NOAA/AFSC, Natl. Marine Mammal Lab., 7600 Sand Point Way NE, Seattle, WA 98115)

The North Pacific right whale, (NPRW), is one of the most endangered baleen whales in the world and has been the focus of intensive population monitoring studies. In 2010 during transit through the Bering Sea right whale critical habitat (BSCH), near-24-h acoustic monitoring was conducted using DiFAR-capable sonobuoys. A new call pattern was detected during a focal follow of a NPRW, consisting of a series of pulses with a fundamental frequency at 110 Hz and peak energy at 640 Hz, ending in a 250–150 Hz downsweep. This call pattern was repeated multiple times and detected in the presence of gunshot and upsweep calls. Directional bearings to the call were a perfect match to those of the gunshot call. Although humpback whales were present, bearings to the humpback vocalizations were in the opposite direction (200 deg difference). The only other species detected visually or acoustically in the area were fin whales. This same pattern was detected in October and November on a 2009 long-term recorder in the BSCH, also in the presence of gunshot and upsweep calls. Because directional information from the sonobuoys during the focal follow exactly matched those from the gunshot call, we attribute this call pattern to the NPRW.

11:30

3aAB12. Passive acoustic monitoring of North Atlantic right whales in Cape Cod Bay. Ying-Tsong Lin, Arthur E. Newhall (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, ytlin@whoi.edu), and Mark F. Baumgartner (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Passive acoustic techniques have been applied extensively to marine mammal monitoring, localization, and tracking. An acoustic fieldwork using hydrophone arrays and free-floating buoys was conducted in Cape Cod Bay in the spring of 2011 for monitoring North Atlantic right whales. Three vertical hydrophone arrays were deployed to form a large triangular network with approximately 12 km on each side, and one horizontal array was mounted on the seafloor at one station. The passive hydrophone arrays operated for 25 days and recorded a vast amount of vocalizations made by humpback, fin, sei, minke, and, most importantly, right whales. These data can be used for localizing these endangered animals and tracking their movements in the bay with a large spatial coverage. Acoustic buoys, the real-time acoustic tracking system, were deployed on two one-day cruises for several hours at a time to

triangulate calling whales over small spatial scales, and can be used to ground truth long-range whale localizations derived from the array network. Results from different passive acoustic techniques will be presented and compared. [Work supported by the Ocean Life Institute and the Marine Mammal Center of the Woods Hole Oceanographic Institution.]

11:45

3aAB13. Passive acoustic and visual monitoring of humpback whales (*Megaptera novaeangliae*) in the Olympic Coast National Marine Sanctuary: Importance of quantifying call type. Amanda J. Cummins (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0205), Erin Oleson (NOAA Fisheries, Honolulu, HI 96814), John Calambokidis, Greg Schorr, Erin Falcone (Cascadia Res. Collective, Olympia, WA 98501), Sean Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0205)

Humpback whales (*Megaptera novaeangliae*) produce a variety of vocalizations such as social and feeding calls as well as patterned calls that comprise song. Typically, social and feeding vocalizations do not follow the highly structural format of song. High-frequency acoustic recording packages were deployed in the Olympic Coast National Marine Sanctuary July 2004 through June 2009 while visual surveys were conducted throughout the year at approximately monthly intervals. Humpback whales were detected visually and acoustically; however, there was a mismatch in the peak seasonality of these detections. Visual detections occurred in all seasons but peaked in summer and early fall. Acoustic detections were documented primarily in late summer to early winter. Male humpback whales are known to produce long songs primarily during their winter breeding season. To test whether the detection differences between visual and acoustic surveys could be explained by changes in the whales' vocal behavior, we quantified the relative occurrence of song and non-song calling using a variety of metrics and related the occurrence to the visual survey sightings. We show how identification of the type of acoustic detection is an important consideration and can help address biases introduced by seasonal differences in the production rate of reproduction-related calls.

WEDNESDAY MORNING, 2 NOVEMBER 2011 — ROYAL PALM 5/6, 7:30 TO 11:55 A.M.

Session 3aBA

Biomedical Acoustics and Physical Acoustics: Biomedical Applications of Acoustic Radiation Force

Mostafa Fatemi, Chair
Physiology and Biophysics, Mayo Clinic, 200 First St., SW, Rochester, MN 55905

Chair's Introduction—7:30

Invited Papers

7:35

3aBA1. Production of shear waves with novel radiation force beams. James Greenleaf, Shigao Chen, and Matt Urban (Mayo Clinic, Dept. of Physiol. and Bioengineering, 200 First St., S.W., Rochester, MN 55905)

Propagating shear waves in tissue can be measured with high frame rate Doppler or correlation methods. The measured characteristics of the shear waves, such as speed versus frequency, can be used to deduce material properties such as complex viscoelastic modulus using physics models appropriate to the geometry and properties of the tissue. This inverse problem is characterized by calculating the storage and loss modulus as a function of frequency and requires appropriate tissue motion, which in turn requires optimized dynamic radiation force distributions. We will discuss novel radiation force distributions that provide enhanced tissue motions appropriate to dealing with the inverse problem of determining tissue material properties from ultrasonically measured tissue motion.

7:55

3aBA2. Acoustic radiation force for rapid detection of particles in biological liquids. Lev Ostrovsky (Zel Technologies and Univ. of Colorado, 325 Broadway, Boulder, CO 80305), Aba Priev, and Yechezkel Barenholz (Hebrew Univ. of Hadassah Med. School, Israel)

As known, ultrasonic standing waves can be used to concentrate particles and biological cells into separated bands. Acoustical separation based on plane standing waves is limited to particles of few microns and larger. This presentation concerns using acoustic radiation force (ARF) produced by cylindrical standing waves for detection of high-density submicron-size particles (bacterial cells) in pressure nodes and low-density particles (fat globules) in antinodes. Theoretical calculations show that in a cylindrical ultrasonic chamber, ARF near the central node can exceed the force at the chamber periphery by about 20 times. In a cylindrical standing wave, ARF may induce movement of bacteria with a speed of the order of a few millimeter per second at a frequency of 2 MHz and pressure amplitude of 100 kPa, whereas the speed of bacteria in plane standing wave does not exceed 0.2 mm/s under the same conditions. The cylindrical standing wave system performance was tested for the *E. coli* bacteria in water and for a multi-component system containing fat globules and somatic cells in milk. Dilute suspensions of bacteria or fat globules were concentrated by at least 2 orders of magnitude.

8:15

3aBA3. Parameterization of the scattering and radiation forces and torques on spheres in terms of complex partial wave s-functions: Applications and interpretation. Philip L. Marston and Likun Zhang (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

A wide range of physical responses of spheres in acoustic beams, including radiation forces and torques, are closely related to properties of the far field acoustic scattering. See for example the short reviews in [Marston, J. Acoust. Soc. Am. **125**, 3539–3547 (2009)] for radiation forces and [Zhang and Marston, J. Acoust. Soc. Am. **129**, 1679–1680 (2011)] for torques. When evaluating these properties it

can be advantageous to retain the notation of resonant scattering theory from prior work for plane-wave illumination. That notation involves the complex partial wave, s-function such that the nth partial wave, the amplitude, is proportional to (s-1) and s is unimodular in the absence of absorption. Some applications of this parameterization to the interpretation and understanding of the scattering by, and radiation forces and torques on, spheres in acoustic beams will be examined. These include examples of forces and torques associated with helicoidal beams and simple methods for modulating the torque. This formulation of the scattering should be helpful for cross-disciplinary applications. Some potentially confusing points from the literature on acoustic plane-wave scattering from the 1970s and early 1980s will be clarified. [Work supported by ONR and by NASA.]

8:35

3aBA4. Vibro-acoustic doppler. Alireza Nabavizadeh, Matthew W. Urban, and Mostafa Fatemi (Dept. of Physiol. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St., SW, Rochester, MN 55905)

This paper describes the principles and initial experimental results of a new Doppler method called vibro-acoustic Doppler (VAD). VAD uses acoustic response of a moving object to a highly localized dynamic radiation force of ultrasound to measure the velocity of the object. The low frequency radiation force is exerted by two co-focused ultrasound beams with slightly different frequencies. The acoustic response of the object is detected by a hydrophone. A formula that describes the relation between the Doppler frequency shift of the emitted acoustic field and the velocity of the moving object is reported. To verify the theory, experiments are conducted on a moving string and a fluid flow phantom. Results show that the error in velocity measurement is less than 9.1% for either phantom. An advantage of this method over the traditional ultrasound Doppler is that velocity measurement with VAD is almost independent of the angle between the ultrasound beam and motion direction. It is shows that in the worst case, the error is 10.3% for a ± 30 deg angle variation. Potential biomedical applications of VAD will be discussed.

Contributed Papers

8:55

3aBA5. Young's modulus estimation of bovine lens *ex-vivo* using a laser-induced microbubble under impulsive acoustic radiation force. Sangpil Yoon (Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712, s.yoon@mail.utexas.edu), Salavat Aglyamov, Andrei Karpiouk, and Stanislav Emelianov (Univ. of Texas at Austin, Austin, TX 78712)

According to the most widely accepted theory of presbyopia, the age-related loss of accommodation is attributed to Young's modulus changes in the lens. Previously, we have developed an approach to measure the mechanical properties of viscoelastic medium using microbubbles and acoustic radiation force. In this study, we tested this technique to assess the mechanical properties of bovine lenses ex-vivo. An impulsive acoustic radiation force was applied to laser-induced microbubbles created with nanosecond laser pulses at different locations in the lens. An acoustic radiation force with typical duration less than 150 μs was generated by a 3.5 MHz transducer. For accurate spatio-temporal measurements of the microbubble's displacement, a custom-made ultrasound system consisting of two 25 MHz transducers was built. The first transducer emitted a train of pulses and the other transducer received the train of echoes reflected from the microbubble. The developed system was operating at 400 kHz pulse repetition frequency. The results show good agreement between experimental measurements and the theoretical model of microbubble dynamics. Evaluation of the spatial distribution of elasticity demonstrates that the Young's modulus of the nucleus is higher than that of the cortex for bovine eye lenses of mature animals. [Work supported by NIH Grant EY018081.]

9:10

3aBA6. Measurement of elasticity of thin elastic layers with radiation force and wave propagation methods. Matthew W. Urban, Ivan Z. Nenadic, Miguel Bernal, Bo Qiang, James F. Greenleaf, and Shigao Chen (Dept. of Biomed. Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

In the field of tissue engineering, monitoring the elasticity of developing tissue cultures is important for evaluating their growth and maturation. In this work, we used wave propagation methods to measure the elasticity of thin tissue-mimicking layers to test the feasibility of measuring the elasticity of developing tissue cultures. We developed finite element and analytical models for investigating this problem and found good agreement between the results from both models. We used ultrasound radiation force to induce propagating waves, and the waves are measured using high frame rate ultrasound imaging at 10 kHz. The wave modes were identified and compared with an analytic model for Rayleigh wave propagation in a thin elastic layer bound to a solid substrate. The same gelatin mixture was used to make phantoms 1, 4, and 50 mm thick, where the phantom with 50 mm thickness was used as control and evaluated with shear wave imaging methods. The analytic model was used to fit data from experiments in the large block and thin layers of 1 and 4 mm thick, and the measured shear moduli were 22.7, 21.8, and 22.5 kPa, respectively. These results provide a validation for measurement of elasticity in thin elastic layers.

9:25

3aBA7. Modeling of the radiation force imparted on a spherical kidney stone by a focused beam of an ultrasound array. Oleg Sapozhnikov and Michael Bailey (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Radiation force imparted on an elastic scatterer by a narrow ultrasound beam is different from that imparted by a plane wave, especially when the beam waist is comparable or smaller than the scatterer's diameter. Such a situation exists when a kidney stone is pushed by an acoustic wave emitted by a megahertz frequency ultrasound array. A spherical stone of several millimeters in diameter with elastic properties similar to calcium oxalate monohydrate kidney stones is considered. An acoustic wave is taken in a form of a continuous-wave 2.5 MHz focused ultrasound beam emitted by Philips HDI C4-2 imaging array. To study radiation force, the transducer field is created as a sum of plane waves of various inclinations, and scattering of each plane waves is modeled based on a known solution. Numerical calculations show that within some frequency range both the backscattering from and the radiation force on a kidney stone exceeds the values for absolutely soft or rigid spheres of the same diameter. A vector component of the radiation force can be created in a direction other than the ultrasound propagation by targeting off the stone center. [Work supported by NIH (DK43881, DK092197, and DK086371), RFBR, and NSBRI through NASA NCC 9-58.]

9:40

3aBA8. Acoustic radiation force applied to the treatment of lower pole kidney stones. Michael R. Bailey, Bryan W. Cunitz, Barbrina Dunmire, Yak-Nam Wang, Wei Lu, Julianna C. Simon, Peter J. Kaczkowski, and Oleg A. Sapozhnikov (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Lower pole kidney stones have lower rates of clearance after shock wave lithotripsy or ureteroscopy compared to other stone locations. Residual stone fragments, post treatment, often lead to additional morbidity and secondary surgery. We describe the use of acoustic radiation force created by transcutaneous focused ultrasound to manipulate the location of stone fragments within the collecting system in order to facilitate their passage. Artificial and human stones were placed in the lower pole of kidneys of a porcine animal model. An open architecture, software based diagnostic ultrasound system and scanhead

were modified to produce roughly 100 μs pulses of up to 16 MPa peak positive pressure in water. Stone motion was observed in real-time with simultaneous imaging through the same scanhead and with fluoroscopy. All stones were seen to move. Stone velocities were on the order of 1 cm/s. Stone displacement distance was up to 3 cm, and operators could generally control the direction of stone movement. No evidence of thermal necrosis or mechanical damage of renal tissue was observed. Thus acoustic radiation force can be used to facilitate lower pole stone fragment clearance. [Work supported by NIH DK43881, DK086371, DK092197, and NSBRI through NASA NCC 9-58.]

9:55

3aBA9. Determination of thresholds for renal injury in a porcine model by focused ultrasound. Julianna C. Simon, Yak-Nam Wang (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Andrew P. Evan (Indiana Univ. School of Medicine, Indianapolis, IN 46202), Marla Paun, Frank L. Starr, Lawrence A. Crum, and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105)

Recently, a system that uses focused ultrasound to expel renal stone fragments from the kidney by radiation force was developed by Shah *et al.* [Urol. Res. **38**, 491–495 (2010)]. A worst-case treatment protocol using this system would require a total exposure time of 10 min, with a spatial peak pulse average intensity (I_SPPA) of 3600 W/cm^2 in water (16 MPa peak positive pressure) and a 3% duty cycle. As the system operates above the FDA limits for diagnostic ultrasound, our goal is to verify the safety of the system by determining the threshold for renal injury. A 2-MHz annular array generating I_SPPA up to 29 000 W/cm^2 in water was placed on the surface of *in vivo* porcine kidneys and focused in the proximal parenchyma. Exposures of 10 min duration with varying I_SPPAs and duty cycles were repeated at least 6 times. Mechanical tissue damage and cell viability were evaluated histologically using H&E, PAS, and NADH diaphorase stains. The proportion of samples showing injury was plotted versus duty cycle and I_SPPA. The results indicate that the system to expel renal stones operates below the threshold for kidney injury. [Work supported by NIH DK043881, DK086371, DK092197, and NSBRI through NASA NCC 9–58.]

10:10–10:25 Break

10:25

3aBA10. On the numerical computation of the acoustic radiation force generated by a modulated sound field. Egor Dontsov, Bojan Guzina (Dept. of Civil Eng., Univ. Of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455, donts002@umn.edu), Shigao Chen, and Mostafa Fatemi (Mayo Clinic College of Medicine, MN 55905)

The acoustic radiation force (ARF), which signifies the average transfer of momentum from the sound wave to the propagating medium, is nowadays frequently used to compute the mean displacement field in soft tissues due to the action of a focused ultrasound beam. In situations when the sound field used to generate the ARF is modulated, the latter depends on both spatial and temporal derivatives of the acoustic intensity. Unfortunately, the numerical computation of the nonlinear acoustic solution due to modulated focused ultrasound beam is complicated by the presence of two disparate time scales in the formulation. To deal with the problem, the KZK-type spatial scaling is complemented by its dual-time-scale companion, where the temporal coordinate is split into its "fast" and "slow" components, allowing one to track the ultrasound-scale oscillations and modulation-scale variations separately in the solution. In this case, the nonlinear acoustic solution (and consequently the ARF) can be effectively computed in the "fast" frequency domain and "slow" time domain for any given modulation envelope, transient, or steady-state. The proposed developments are both validated by the experimental results and illustrated via numerical examples that show the effectiveness of the computational scheme.

10:40

3aBA11. The role of constitutive nonlinearities and sound modulation in the generation of the acoustic radiation force. Bojan Guzina and Egor Dontsov (Dept. of Civil Eng., Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455, guzina@wave.ce.umn.edu)

This study investigates the acoustic radiation force (ARF) in soft tissues when the generating ultrasound field is modulated with a low-frequency envelope (10^2-10^4 Hz range). On approximating the soft tissue as that of a nonlinear elastic material with heat conduction and viscosity, the system of nonlinear balance equations, governing both the ultrasound-scale oscillatory motion and the ARF-induced mean motion, is formulated explicitly. To deal with the effects of ultrasound modulation, a dual-time-scale approach featuring the "fast" (ultrasound-scale) and "slow" (modulation-scale) temporal coordinates is deployed. In this setting the governing equations for the mean motion, featuring the ARF as the body force term, are extracted by taking the "fast" time average of the nonlinear balance laws. The ARF is shown to consist of two distinct terms, namely (i) the potential term, which is proportional to the gradient of the ultrasound intensity and (ii) the axial term, which contains both an attenuation-driven component and a modulation-driven component. A salient feature of the new solution for the ARF is that its entries feature specific combinations of the elastic nonlinearity coefficients, which may vary depending on tissue type. For completeness, the proposed formula is illustrated by numerical simulations and compared to the existing expressions.

10:55

3aBA12. Modeling of shear waves generated in a soft solid by a piston. Kyle S. Spratt, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029)

This work is motivated by the transient elastography experiments described by Catheline *et al.* [J. Am. Stat. Assoc. **105**, 2941–2950 (1999)], the purpose of which was to measure the phase speed of shear waves propagating through tissue phantoms. In that work, a small circular piston was used to generate the shear waves, the motion of which was perpendicular to the bounding surface of the sample. The current work is a theoretical investigation of this type of source condition based on the assumption of a linear elastic medium and using an angular spectrum approach to solve for the entire elastic field, both compressional and shear waves, that results from a given velocity distribution at the source plane. Special attention is paid to the velocity field near the source, in particular how the near incompressibility of the tissue-like medium is conducive to the generation of shear waves from such a compressive piston source. For high-frequency excitation, in which the resulting shear wave disturbances are beam-like, the validity of using a parabolic approximation to describe diffraction of the transverse motion of the field in the paraxial region is investigated. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

11:10

3aBA13. Two-dimensional shear elasticity imaging using comb-push acoustic radiation force and algebraic direct inversion of the motion differential equation. Pengfei Song, Armando Manduca, Zhoubo Li, Heng Zhao, Matthew W. Urban, James F. Greenleaf, and Shigao Chen (Dept. of Physio. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St., SW, Rochester, MN 55905, song.pengfei@mayo.edu)

Tissue mechanical properties can be obtained by algebraic direct inversion (ADI) of the shear-wave motion differential equation, which is insensitive to reflection and geometry of the pushing beam. Shear waves normally generated by a focused ultrasound pushing beam have limited spatial extent in depth and are transient in time, leading to noisy and unstable ADI results. A comb-shape acoustic radiation force distribution can generate shear waves with longer spatial extent and time duration, facilitating more robust ADI. For 400 μs, a linear array transducer simultaneously transmitted four unfocused pushing beams (12 on elements for each pushing beam, 8 off elements between beams) into a tissue-mimicking phantom ($\sim$1.5 kPa) with a cylindrical inclusion ($\sim$4 kPa). The ultrasound system (Verasonics Inc.) then immediately switched to flash imaging mode (frame rate = 2 kHz, spatial resolution = 0.31 mm) to measure shear-wave displacements in a 38 mm by 30 mm domain that was used for ADI. The reconstructed 2D shear elasticity map provides accurate shear elasticity estimates (background = 1.5 kPa with 7% variance; inclusion = 3.9 kPa with 15% variance) and excellent contrast between the background and inclusion.

11:25

3aBA14. A mechanism of tissue emulsification by high intensity focused ultrasound. Julianna C. Simon, Oleg A. Sapozhnikov, Vera A. Khokhlova, Yak-Nam Wang, Tatiana D. Khokhlova, Lawrence A. Crum, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu)

High intensity focused ultrasound (HIFU) has been shown to emulsify tissue in histotripsy created by cavitation clouds and millisecond boiling produced by shock wave heating; however, the mechanism by which millimeter-sized boiling bubbles or cavitation bubble clouds emulsify tissue into submicron-sized pieces is not well understood. Here, we experimentally test the hypothesis that acoustic atomization occurs and a miniature acoustic fountain forms (due to radiation force) at the edge of the millimeter-sized boiling bubble. Through high-speed photography, we observed the violent removal of bovine/porcine liver fragments upon focusing a 2-MHz transducer of 70 MPa peak positive and 15 MPa peak negative pressures at the tissue–air interface. Velocity of the fountain projectiles ranged from 5 to 15 m/s. When the focal amplitudes were reduced to 15 MPa peak positive and 7 MPa peak negative pressures, atomization occurred intermittently. The fountain projectiles were collected and examined histologically using H&E and NADH–diaphorase stains. Results show fragments ranging from tens of microns down to submicron in size. The larger fragments contain distinct tissue aggregates with intact cells and the small particles are cell fragments with few to no intact nuclei. [Work supported by NIH DK43881, DK070618, EB007643, DK007742, and NSBRI through NASA NCC 9–58.]

11:40

3aBA15. Vibro-acoustography beam formation with reconfigurable arrays. Matthew W. Urban (Dept. of Physio. and Biomed. Eng., 200 First St., SW, Rochester, MN 55905, urban.matthew@mayo.edu), Hermes A. S. Kamimura, Antonio A. O. Carneiro (Universidade de Sao Paulo, Ribeirao Preto, SP, Brazil), Mostafa Fatemi, and Azra Alizad (200 First St., SW, Rochester, MN 55905)

Vibro-acoustography (VA) is an imaging modality that measures the acoustic response from stimulation produced by the interaction of two ultrasound beams at different frequencies. In this work, we present a numerical study of the use of reconfigurable arrays (RCA) for VA beam formation. A parametric study of the aperture selection, number of channels, number of elements, focal distance, and steering parameters is presented in order to show the feasibility and evaluate the performance of VA imaging based on RCA. Furthermore, an optimization for beam steering based on the channel assignment is proposed for balancing the contribution of the two waves in the steered focus. The point-spread function is calculated based on angular spectrum methods using the Fresnel approximation for rectangular sources. Simulations considering arrays with 50×50 to 200×200 elements with the number of channels varying in the range of 32 to 128 are evaluated to identify the best configuration for VA. We concluded that RCA transducers can produce spatial resolution similar to confocal transducers, steering is possible in elevation and azimuthal planes, and effective setting parameters including number of elements, number of channels, maximum steering, and focal distance are suggested for VA clinical imaging.

WEDNESDAY MORNING, 2 NOVEMBER 2011 PACIFIC SALON 6/7, 8:30 A.M. TO 12:00 NOON

Session 3aEA

Engineering Acoustics and Biomedical Acoustics: Micromachined Acoustic Transducers I

Neal Hall, Cochair
Electrical and Computer Engineering Dept., Univ. of Texas at Austin, 1 University Station, Austin, TX 78712

Won-Kyu Moon, Cochair
Dept. of Mechanical Engineering, Pohang Univ. of Science and Technology, Pohang, 790-784, Korea

Invited Papers

8:30

3aEA1. Capacitive micromachined ultrasonic transducers: A platform technology. Butrus (Pierre) T. Khuri-Yakub (Dept. of Elec. Eng., Stanford Univ., Stanford, CA 94305)

Capacitive micromachined ultrasonic transducers (CMUTs) have been under development as alternative transducers for generating and detecting ultrasound. The advent of silicon micromachining enables the realization of the full potential of these transducers and provides performance that makes CMUTs competitive and superior to piezoelectric transducers. In immersion applications, CMUTs are possible with fractional bandwidth of over 100%, an electromechanical coupling coefficient close to unity; are made in the form of a single element or 1-D or 2-D arrays of tens of thousand of elements, as well as annular arrays. They have been operated in the frequency range of 100 kHz to 50 MHz, and included in systems with a dynamic range of the order of 150 dB/V/Hz. Custom electronics have been developed and integrated with arrays of transducers to form compact imaging catheters. In airborne applications, CMUTs have been used as mass sensors with atto-gram resolution, and in pulse echo systems operation at high temperature (600 °C). This presentation will first review the operation of CMUTs, the technology used to make them, and some airborne and imaging applications.

8:55

3aEA2. Nickel on glass acoustic microsystems. Robert D. White, Zhengxin Zhao, Minchul Shin, Joshua S. Krause, and Shuangqin Liu (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu)

An electroplated nickel-on-glass surface micromachining process has been developed and applied to fabricate condenser MEMS microphone arrays and ultrasound arrays. The major advantage of these arrays over polysilicon surface micromachined sensors is the significant reduction in stray capacitance that can be achieved by using a glass substrate. For 600 μm diameter elements with a 1.5 to 2 μm sense gap, the stray capacitance per element was reduced from approximately 140–1 pF when compared to similar polysilicon

devices fabricated using the PolyMUMPS process. It is important to reduce stray capacitance, since it is the driving factor for two dominant noise sources: preamplifier voltage noise and bias feedthrough noise. In addition, stray capacitance can place a limit on system bandwidth for high frequency applications such as ultrasound. In this paper, we report on cMUT ultrasound arrays for in-air Doppler applications, as well as microphone array-on-a-chip devices for aeroacoustic measurements. Characterization of system performance and comparison to previously reported polysilicon devices [Krause *et al.* and Shin *et al.*, ASA Spring Meeting, 2011] is ongoing. Other properties such as material damping, structure residual stress, temperature sensitivity, and substrate charging are also under investigation. [Work supported by Spirit Aerosystems and Draper Labs.]

9:20

3aEA3. Piezoelectric microspeakers built on various diaphragms. Eun S. Kim (Dept. of Elec. Eng.-Electrophys., Univ. of Southern California, Los Angeles, CA 90089-0271, eskim@usc.edu)

This paper describes and compares four different types of diaphragm-based piezoelectric microspeakers built on (1) a compressively stressed silicon nitride diaphragm, (2) a parylene diaphragm, (3) a dome-shaped silicon nitride diaphragm, and (4) a PZT bimorph diaphragm. The main innovation in the first device is the usage of a wrinkled diaphragm that supports a flat diaphragm, where piezoelectric actuation happens, and allows a large bending displacement. The second device is to exploit the very low elastic modulus of parylene (a polymer material), and is built on a 1.5 μm thick parylene diaphragm with electrodes and piezoelectric ZnO film. Also described in this paper is an acoustic transducer built on a 1.5 μm thick dome-shaped silicon nitride diaphragm (2 mm in radius, with a circular clamped boundary on a silicon substrate) with electrodes and piezoelectric ZnO film. The dome diaphragm is shown to effectively release residual stress through volumetric change of its shape. Finally, this paper describes a microspeaker (composed of 8 mm square PZT bimorph and bulk-micromachined silicon) that shows flat diaphragm displacement from DC to 8 kHz. A bimorph diaphragm is formed by gluing two 127 μm thick PZT sheets and attaching them to a micromachined silicon substrate.

9:45

3aEA4. Piezoelectric thin films for a high frequency ultrasound transducer with integrated electronics. Flavio Griggio, Hyunsoo Kim, Insoo Kim, Thomas N. Jackson, Kyusun Choi, Richard L. Tutwiler, and Susan Trolier-McKinstry (Penn State Univ., 151 Mater. Res. Lab., University Park, PA 16802, STMcKinstry@psu.edu)

Lead zirconate titanate (PZT) piezoelectric films were integrated into prototype one-dimensional array transducers. Linear arrays of diaphragm transducers were prepared using PZT films of 0.5–1.7 μm in thickness and surface micromachining techniques. For this purpose, the PZT and remaining films in the stack were patterned using ion-beam or reactive ion etching and partially released from the underlying silicon substrate by XeF_2 etching. The PZT films were prepared by chemical solution deposition, and have $e_{31,f}$ piezoelectric coefficients of -6 to -12 C/m^2, depending on the crystallographic orientation. Impedance measurements on the fabricated structures showed resonance frequencies between 3 and 70 MHz for fully and partially released structures depending on the transducer dimensions and vibration modes. In-water transmit and receive functionalities have been demonstrated. A bandwidth on receive of 66% has been determined. Because of the small thickness of the piezoelectric element, the elements can be driven at less than 5 V. This enables the ultrasound system to be CMOS compatible, and hence massive miniaturization. A custom designed CMOS chip which enables beamforming on transmit, transducer excitation, amplification, digitization, and data storage was designed and fabricated.

10:10–10:30 Break

Contributed Papers

10:30

3aEA5. Minimizing noise in micromachined piezoelectric microphones. Robert Littrell (Baker-Calling Inc., 1810 14th St., Ste 102, Santa Monica, CA 90404) and Karl Grosh (Univ. of Michigan, Ann Arbor, MI 48109)

Piezoelectric microelectromechanical systems (MEMS) microphones have been researched for over 30 yr because they are relatively easy to build, output a signal without any biasing circuitry, and are relatively linear. The primary impediment to mass utilization of piezoelectric MEMS microphones has been the noise levels of these devices, which have been unacceptably high. The input referred noise of most piezoelectric MEMS microphones is greater than or equal to 55 dB(A) while commercial capacitive MEMS microphones typically have noise floors between 32 and 38 dB(A), roughly ten times lower. In order to achieve competitive noise levels in a piezoelectric MEMS microphone, a systematic approach to mechanical and electrical optimizations must be used. A key microphone metric for this optimization is acoustically generated electrical energy, as opposed to sensitivity. This optimization can be used to determine the minimum achievable noise floor for piezoelectric sensors with relatively few assumptions. Further, this optimization allows for the determination of materials, electrode patterns, layer thicknesses, and mechanical structures that minimize the noise. These models have been used to design and build piezoelectric MEMS microphones with a noise floor of 34 dB(A) and indicate that further improvements can be achieved.

10:45

3aEA6. Ultrasonic sensing using thermal mechanical noise of capacitive micro-machined transducers. Shane Lani, Sarp Satir, Gokce Gurum, Karim G. Sabra, and F. Levent Degertekin (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332-0405)

Monolithic integration of CMUTs and CMOS electronics minimizes interconnect parasitics which allows recording the actual thermal–mechanical component of the ultrasonic noise field. Consequently, an estimate of the pulse-echo response (or Greens function) between two CMUT sensors can be obtained from the cross-correlation of thermal–mechanical noise recorded by these two sensors, as shown by Weaver and Lobkiss [JASA, **113**(5), 2611-21]. This provides a foundation for passive ultrasound imaging using only the thermal–mechanical noise field, without active transmitter elements. We designed and fabricated monolithic a 32 element CMUT-on-CMOS ring array (d = 725 μm) for intravascular imaging with low noise transimpedance amplifiers (TIAs) implemented in 0.35 μm CMOS technology. The bias voltage was set near the collapse value of the CMUT membrane to maximize receiver sensitivity. Demonstration experiments were conducted by immersing the CMUT array in a water bath to sense/image the water–air interface and compact targets from noise signals in the frequency band 14–25 MHz. These experimental results were consistent with the sensor and target locations and were validated using conventional pulse-echo measurements. This totally passive ultrasonic technique could improve

ultrasound imaging of near-field targets in the deadzone created by active transmitters biasing the receivers and may lead to imaging using evanescent waves.

11:00

3aEA7. Recent progress toward a nanostructured piezoelectric microphone. Adam D. Mathias, Jon R. Fox, Jean P. Cortes, Stephen B. Horowitz (Miltec Systems, 678 Discovery Dr. Huntsville, AL 35806), Mohan Sanghadasa, and Paul Ashley (U.S. Army—AMRDEC, Redstone Arsenal, AL 35898)

In an attempt to push the performance limits of piezoelectric MEMS microphones, several variations of micromachined acoustic sensors that contain piezoelectric zinc oxide nanorods embedded in a flexible polymer matrix were designed, fabricated, and characterized. The polymer matrix offers high compliance, high aspect ratio, ultrathin diaphragms, and low residual stress, while the zinc oxide nanorods provide high piezoelectric coupling. The devices, fabricated on a silicon substrate, consist of 100–800 μm diameter circular diaphragms composed of the piezoelectric polymer composite sandwiched between a circular gold bottom electrode and an annular gold top electrode. Electrical, mechanical, and acoustic characterization were performed on the fabricated sensors. Acoustic measurements included frequency response, sensitivity, linearity, and noise floor. Electrical properties such as resistance and capacitance, and piezoelectric properties, such as the effective piezoelectric coefficient, were also measured.

11:15

3aEA8. Electroacoustic parameter extraction of a piezoelectric microelectromechanical systems microphone. Matthew D. Williams, Benjamin A. Griffin, Tiffany N. Reagan, and Mark Sheplak (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, 231 MAE-A, P.O. Box 116250, Gainesville, FL 32611-6250)

A novel suite of parameter extraction experiments were used to assess the accuracy of individual elements of a lumped element model for a microelectromechanical systems based piezoelectric microphone. The MEMS microphone was developed via model-based design utilizing the lumped element model for use in aeroacoustic applications. Laser vibrometer scans of the microphone diaphragm while subjected to electrical or pressure excitation provided experimental predictions for the effective electroacoustic piezoelectric coupling coefficient, diaphragm compliance, and mass. The experimental results were compared with analytical predictions from a piezocomposite diaphragm model for these individual lumped elements. Associated lumped element model predictions were also compared with the results of device characterization experiments. Similar trends in theory and experiments were observed, though comparative error in element values was attributed to uncertainty in model-inputs, most notably thin-film residual stresses in the microphone diaphragm. [This work was sponsored by Boeing Corporation.]

11:30

3aEA9. State space modeling of surface mount silicon microphones. A. Guclu Onaran, Caesar T. Garcia, Alex Liu, Matt Christensen (701 Tillery St., Ste. A-4B, P.O. Box 7, Austin, TX 78702), Michael Kuntzman, Karen Kirk, and Neal A. Hall (Dept. of Elec. and Comput. Eng. ENS 612A 1 University Station CO803 Austin, TX 78712-0240)

Most surface mount microelectromechanical system (MEMS) microphone packages are similar in construction, consisting of a printed circuit board with sound inlet, a MEMS die with a through-wafer etch aligned over the sound inlet, and cap which serves to protect the structure and render an enclosed back volume. From a lumped modeling perspective, this system is a network of acoustical and mechanical elements. Network models (i.e., equivalent circuit models) have proven to be the most common modeling technique for simulating important features of these microphones, including frequency response functions and internal noise floors. While these models have many advantages including their ability to be solved efficiently using modern circuit simulation software, they do not lend themselves well to an understanding of system dynamics as a decomposition of the fundamental mechanical modes of the packaged system. We present a state space model for complete MEMS microphone packages and present frequency response simulations as a superposition of the system's eigenmodes. In addition to offering insight into package behavior, we believe these models are better equipped to address advanced features such as feedback altered dynamics using internal actuation capabilities. Simulations are compared with measurements on a surface mount optical MEMS microphone discussed prior.

11:45

3aEA10. Magnetostrictive micro-loudspeaker. Thorsten S. Albach and Reinhard Lerch (Sensor Technol., Univ. of Erlangen-Nuremberg, Paul-Gordan-Str. 3-5, 91052 Erlangen, Germany, talbach@lse.eei.uni-erlangen.de)

A magnetostrictive MEMS-actuator which performs as a loudspeaker is presented. The design of the device, its theoretical modeling, as well as measurements showing its acoustical performance are outlined. Instead of a closed membrane, two combs of monomorph bending cantilevers are fabricated face to face on a silicon substrate. The small air gaps between the cantilevers lead to an advantageous reduction of the mechanical stiffness. Each cantilever consists of one active (magnetostrictive) layer and further passive layers. The cantilevers deflect simultaneously when a magnetic field is applied. An additional magnetic dc field linearizes the working point. An electro-acoustic lumped element model is developed to calculate the sound pressure level (SPL) generated in a 2 cm^3 ear volume coupler. This model incorporates finite element simulations analyzing the mechanical behavior of the cantilevers. Measurement results validate the model. The effect of the variation of different design parameters on the frequency characteristics is demonstrated. The desired characteristics of a closed membrane for example can be achieved, if the air gaps between the cantilevers are smaller than 3 μm. The prototype micro-loudspeakers have active areas of up to 3.0 mm $\times$ 2.5 mm with air gaps of 10 μm and generate a SPL up to 101 dB at 400 Hz.

Session 3aED

Education in Acoustics and Physical Acoustics: Undergraduate Research Exposition (Poster Session)

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

Mardi C. Hastings, Cochair
G. W. Woodruff School of Mechanical Engineering, Georgia Inst. of Tech., 126 Love Bldg, 771 Ferst Dr., Atlanta, GA 30332-0405

Contributed Papers

All posters will be on display from 10:00 a.m. to 12:00 noon. All authors will be at their posters for the duration of the session.

3aED1. Ultrasonic analysis of breast tissue for pathology classification. Kristina M. Sorensen and Timothy E. Doyle (Dept. of Phys., Utah State Univ., 4415 Old Main Hill, Logan, UT 84322-4415, Kristina.Sorensen@aggiemail.usu.edu)

The effectiveness of breast conservation surgery (BCS) or lumpectomy relies heavily upon pathology to assure negative or cancer free margins. In a study to develop an intraoperative pathology method, surgical specimens from 17 breast cancer patients were tested with high-frequency (HF) ultrasound (20–80 MHz) to search for pathology sensitive features for the detection of cancer in margins during BCS. Pulse-echo and pitch-catch waveforms were obtained using two single-element 50-MHz transducers. Analysis of time-domain waveforms yielded ultrasonic attenuation and sound speed, whereas fast Fourier transforms of the waveforms produced ultrasonic spectra and cepstra for the evaluation of spectral peak density and cepstrum slope. Spectral peak density indicated significantly higher values for carcinomas and precancerous pathologies than for normal tissue. Cepstrum slope exhibited a substantial distinction between benign and adipose tissues when compared with normal and malignant pathologies. The attenuation coefficients were sensitive to fat necrosis, fibroadenoma, and invasive lobular carcinoma. A multivariate analysis of these parameters was used to further distinguish pathologic classification. Evaluation of ultrasonic attenuation, spectra, and cepstra permits differentiation between normal, adipose, benign, and malignant breast pathologies. These results indicate that HF ultrasound may assist in eliminating invasive re-excision for lumpectomy patients. [Work supported by NIH R21CA131798.]

3aED2. Foreign accent production and perception: An acoustic analysis of non-native Japanese. Lucy Gubbins and Kaori Idemaru (Dept. of East Asian Lang. and Lit., Univ. of Oregon, Eugene, OR 97403, idemaru@uoregon.edu)

What are the characteristics of non-native speech and what contributes to the perception of foreign accent by native speakers? In this study, two experiments are conducted to characterize the acoustic features of non-native Japanese production and to examine how these non-native features influence native Japanese perception of foreign accent. In the production experiment, stop consonants and vowel formants were compared between native Japanese speakers and English speakers with 2 and 4 yr of Japanese instruction. The second experiment examines native listener judgments of foreign accent using a visual analog scale. Preliminary analysis reveals that in both stop consonant and vowel production, learners vary considerably from native speakers, specifically in the production of /k/ and /t/ voice onset time, F1 of /e/, and F2 of /u/ and /i/. These features are correlated with judgment ratings by native Japanese listeners. The findings reveal that even after significant experience in the L2 classroom, speakers still struggle to achieve native-like production of various segments, offering insight into specific problem areas which native English speakers might encounter when learning Japanese pronunciation.

3aED3. The effect of musical training on auditory perception. Irene Kannyo and Caroline M. DeLong (Dept. of Psych., Rochester Inst. of Tech., 18 Lomb Memorial Dr., Rochester, NY 14623)

Previous research has shown that musical training affects the type of cues people use to discriminate between auditory stimuli. The current study investigated whether quantity of musical training and musical area of expertise (voice, percussion instrument, non-percussion instrument) affected musical feature perception. Participants with 0–4 yr of experience (13 non-musicians), 5–7 yr of experience (13 intermediate musicians), and 8 yr or more of experience (13 advanced musicians) were presented with pairs of 2.5 s novel music sequences that were identical (no change trials), differed by one musical feature (pitch change, timber change, or rhythm change), and differed by two musical features (pitch and timber change, pitch and rhythm change, or timber and rhythm change). In 64 trials, participants had to report whether they heard a change, as well as classify the specific type of change. Participants in the advanced group ($M = 91.2\%$) and intermediate groups ($M = 85.0\%$) performed significantly better than non-musicians ($M = 70.0\%$). There was no effect of area of musical expertise (voice or instrument) on musical feature change detection. These results suggest that musical training in any area increases the ability to perceive changes in pitch, timber, and rhythm across unfamiliar auditory sequences.

3aED4. Acoustic correlates of stress in Spanish infant-directed speech. Stephanie Jacobson, Nancy Ward, and Megha Sundara (UCLA Phonet. Lab., Dept. of Linguist., Los Angeles, CA 90095)

In infant speech research, children's input in often examined to determine at what age their production is affected by their language experience. In this study, we examined production of stress in Spanish infant-directed speech. Although the correlates of stress in English infant-directed speech have been examined, the same has not been done for Spanish. We analyzed infant-directed speech from eight native Spanish-speaking adults from Central and South America. The infant-directed speech was acquired by recording half-hour sessions of adults interacting in Spanish with 12-month-old infants. The acoustic measures examined to determine stress were: pitch, duration, and intensity. These measures were selected because they have been shown to be the most consistent correlates of stress in Spanish and English adult-directed speech. These measures were taken over a group of six target words: mama, globo, leche, zapato, mira, and agua. These target words allowed us to examine each of the vowels (/a/, /e/, and /o/) in both stressed and unstressed position, as well as in varying word positions. With these data we hope to discover how stress is instantiated in Spanish infant-directed

speech in order to provide a baseline for future investigations of babbling by Spanish-learning infants.

3aED5. Vibrational assessment of ice hockey goalie sticks. Linda J. Hunt (Dept. of Phys., Kettering Univ., 1700 W. University Ave., Flint, MI 48504) and Daniel A. Russell (Penn State Univ., University Park, PA 16802)

While the vast majority of offensive and defensive ice hockey players prefer composite sticks over wood, a majority of goalies prefer wood sticks over composite. To investigate this goalie preference, experimental modal analysis was performed for one wood and two composite goalie sticks in order to extract mode shapes, frequencies, and damping coefficients. Wood and composite sticks were both shown to exhibit modes with antinodes close to the hand location, and frequencies within the range of maximum sensitivity. The sequence of mode shapes was consistent for wood and composite sticks. The wood stick had lower mode frequencies and higher damping coefficients than the composite sticks. Additional testing was performed on a composite goalie stick with and without the addition of tape at the knob end (required by the NHL). This mass loading lowered the frequencies and increased the damping coefficients and moved the nodes of vibration. These differences in mode shapes, frequencies, and damping along with the effects of tape will be discussed in terms of the influence on the perception of feel for goalies preferring wood sticks.

3aED6. Study of the effects of different endpin materials on cello sound characteristics. Clinton R. Fleming, Cassey R. Stypowany, Robert D. Celmer, and Michelle C. Vigeant (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, clntfleming@gmail.com)

The purpose of this study was to investigate the effects of changing a cello's endpin material and boundary conditions on the sound and vibration characteristics of a cello. It was hypothesized that an endpin made of a denser material than stainless steel, which is traditionally used, would improve the tone quality of the cello. In terms of endpin boundary conditions, it was hypothesized that using a shorter endpin with fixed end conditions might also improve the vibration characteristics and sound radiation efficiency of the cello. Objective and subjective tests were conducted to examine the effects of the different endpin materials. Sound power level output and vibration measurements of a cellist playing on different endpins were obtained following ISO 3741. In general, sound power levels and measured vibrations were consistent for all endpins for all notes tested. For the subjective study, volunteer cellists played selected excerpts with the different endpins, not knowing which endpin they were using. This testing showed that the tungsten endpin gives the cello a tone quality with similar warmth to the stock endpin but makes the cello less responsive. The results of both the objective and subjective tests for all endpin materials will be presented.

3aED7. Noise reduction in deployable infrasound arrays through unconventional clover hose layouts. John R. Clay (Dept. of Elec. and Comput. Eng., Univ. of Wyoming, 1000 E. University Ave., Laramie, WY 82071, jclay5@uwyo.edu) and Mihan H. McKenna (US Army ERDC, CEERD-GS-S, Vicksburg, MS 39180)

Infrasound is acoustic energy that lies below the human hearing threshold. This energy is produced from numerous naturally occurring and man-made sources with the ability to propagate hundreds to thousands of miles, depending on the source strength. Infrasound sensors are commonly deployed in networked, multi-sensor arrays for signal localization and characterization. Deployable infrasound sensors, such as Chaparral and IML rely on the use of soaker hoses for noise reduction and long-wavelength spatial averaging. Traditionally, sets of 25–50 ft hoses attach to the ports on each side of the infrasound sensor and are terminated with a cap at the end of the hose. In August of 2010, hose filter geometry testing was conducted using IML ST infrasound sensors designed to minimize the filter footprint by exploring alternatives to the conventional straight hose layouts while maximizing signal-to-noise in desired pass-bands. The optimal arrangement, "Clover Hose Layout," removed the terminator at the end of the hose and looped it to connect to an adjacent port with substantial environmental noise reduction in the 0.5–50 Hz range. Results are presented comparing a suite of hose arrangements including conventional straight hose layouts under calibrated source testing.

3aED8. Acoustical effect of progressive undercutting of percussive aluminum bars. Eric M. Laukkanen and Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N Warner St., Tacoma, WA 98416)

Standard vibraphone bars consist of aluminum beams which are traditionally tuned with an arched undercut, for the purpose of aligning the musical overtones harmonically. The acoustical effect of various progressions of undercuts on aluminum bars was studied using both an aluminum bar and a finite element computer model. The spectral signature of the aluminum bar was examined with a spectrum analyzer, and the corresponding eigenmodes were imaged with an electronic speckle pattern interferometer. These methods were used to analyze the changes in natural frequencies of the bar as matter was removed from various locations. Additionally, the aural character of each cut was captured with an audio recording, and the fundamental tone was normalized over all recordings to make possible a subjective comparison of the timbral differences of differently cut bars.

3aED9. Investigations of the just-noticeable-difference of clarity index (C80). Christopher M. Jasinski, Meghan J. Ahearn, Clothilde B. Giacomoni, Caitlin I. Ormsbee, Matthew J. Schaeffler, and Adam P. Wells (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, jasinski@hartford.edu)

The clarity index for music (C80) is a valuable room acoustic parameter, as it is an objective measure of how listeners perceive clarity. Knowing the just noticeable difference (JND) of C80 is of great importance to concert hall designers. Limited previous research has been conducted to find the C80 JND, and the major studies have limitations, including a small sample size. The reported JND is approximately 1 dB. An ongoing investigation is being conducted at the University of Hartford to establish the validity of these results by experimentally determining the JND, including an investigation on test method. In the first study (Ahearn *et al*., 2009), the subject pool was increased to 51 subjects, using test methods from prior research. The next study (Giacomoni *et al*., 2010) compared two C80 JND test methods, including allowing subjects to switch between signals in real-time. The results revealed the importance of subject training. In the third study (Wells *et al*., 2010), the switch method was used. The resulting JNDs were 1.6, 3.8, and 4.0 dB for each of the studies, respectively, showing that the C80 JND may be significantly higher than previously thought. The testing methods, results, and comparison to the previous studies will be described.

3aED10. Source localization in a reverberant environment using first reflections. Eric S. Haapaniemi, Andrew J. Femminineo, Laura M. Williamson, and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward, Ann Arbor, MI 48109, ehaap@umich.edu)

Matched field processing (MFP) has been shown to be effective for remote sound source localization when the receiving array clearly records direct-path sound. Unfortunately, in imperfectly characterized confined environments, echoes and reverberation commonly degrade localization performance. However, inclusion of first reflections in the requisite MFP field model should improve localization performance and such inclusion may be tractable as well. This presentation describes an acoustic technology development effort focused on source localization in a simple confined environment using both direct-path and once-reflected sounds. Experiments were conducted in a 1.0-m-deep and 1.07-m-diameter cylindrical water tank having a reverberation time of ~10 ms using a single sound projector and a linear receiving array of 16 hydrophones. Measured localization performance is reported for impulsive (100 μs) and longer duration sounds having frequencies from 20 to 150 kHz. Appropriate cropping of the signal coda and inclusion of a simplified characterization of the reflection properties of the tank's walls allows successful integration of first reflections that improve MFP localization results. The intended application of this research is localizing sub-visual cavitation bubbles and other hydroacoustic sound sources in hydrodynamic test facilities. [Work sponsored by NAVSEA through the Naval Engineering Education Center.]

3aED11. Experimental investigation of shock structure in turbulent Coanda jets. Richard J. Shafer and Caroline P. Lubert (Dept. of Mathematics and Statistics, 102 Roop Hall, MSC 1911, Harrisonburg, VA 22807)

Although Coanda surfaces are extremely useful for applications in industrial exhaust, the generation of noise in supersonic flows around such surfaces is not well understood. In order to effectively engineer a solution to this noise problem, the noise generation must be able to be accurately modeled. The two dominating generators of sound are turbulent mixing noise and shock associated noise (SAN). The first step necessary to model SAN is a precise predictor of the shock structure within the jet. A MATLAB algorithm is developed to translate Schlieren photos of the flow into qualitative shock location data. This data will be used in conjunction with the method of characteristics to create a model which predicts shock location around a Coanda surface for several slot widths and exit pressures.

3aED12. Effects of auditory feedback on instrumentalists timber production. Madeline R. Huberth and Timothy A. McKay (Dept. of Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48104, mhuberth@umich.edu)

Previous work has shown that the presence or absence of auditory feedback in musicians has little effect on the performance factors such as note accuracy, timing, and dynamics. This study explored the extent to which an instrumentalist's timber is affected by various feedback conditions. Nine cellists were recorded playing excerpts from the orchestral literature with their hearing completely unimpeded, masked by pink noise, and masked by orchestral recordings of the excerpts requested for the experiment, simulating orchestral playing. Changes in relative harmonic strengths across the three conditions were focused on in analysis. Overall, the results show that the auditory feedback condition has minimal effect on timber production.

3aED13. Dynamic behavior of microwave-cured guitar wood. David-M. Lozupone, Aldean J. Yasay, David A. Dick, Allen R. Beaudry, Andrew T. Sorenson, Stalin L. Vera, and Robert D. Celmer (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117)

An ongoing investigation has been conducted to measure and analyze vibration properties of guitar-body wood samples with different wood finish curing methods for Taylor Guitars. The current method uses ultraviolet radiation to cure a lacquer finish. Virginia Tech's (VT) Material Science & Engineering Department has proposed an alternative method using microwave radiation to cure the finishes, now in its third stage of development. Microwaves (MW) have longer wavelengths than ultraviolet (UV) radiation, allowing greater penetration depth and increased diffusion between wood and lacquer molecules. To investigate the effect of the finish on dynamic behavior, modal analyses have been conducted at the University of Hartford Acoustics Engineering Laboratory using unfinished, UV-cured, and MW-cured guitar wood samples (spruce, maple, ash, and rosewood). Samples were characterized at bending and torsional modes up to 3.2 kHz. Overall results to date demonstrate that both UV-cured and MW-cured samples have higher damping than unfinished samples. Further, the damping provided by the MW method appears to correlate with the number and thickness of filler, finish and top coats. The current results suggest it is possible to produce similar dynamic behavior to UV curing using the MW technique. [Work done in collaboration with Virginia Tech and Taylor Guitars.]

3aED14. Using acoustics to study the effectiveness of fish habitat restoration: A comparison of methods. Michael Bollinger (Biology Dept., The Penn State Univ., State College, PA 16804, mbollinger89@gmail.com), Jennifer L. Miksis-Olds (ARL, Penn State, State College, PA 16804), and Laura E. Madden (Wildlife and Fisheries Sci., Penn State, State College, PA 16804)

Many Pennsylvania reservoirs are created by clearing valleys prior to damming streams. These impoundments are typically devoid of appropriate benthic habitat for fish. The Pennsylvania Fish and Boat Commission has been placing artificial habitat structures in reservoirs across Pennsylvania for 30 years in an effort to increase game-fish production. However, the effectiveness of these efforts is unknown. Active acoustic technology is being considered as an alternative to conventional survey methods for assessing the effectiveness of artificial structure in F.J. Sayers Reservoir in Howard, PA. An autonomous echosounder [(ASL Environmental Sciences, 460 kHz Acoustic Water Column Profiler (AWCP)] was deployed horizontally near artificial rock structures to monitor associated fish activity. The AWCP recorded acoustic backscatter at 1-s intervals continuously for two days. Data was analyzed by (1) manually counting fish tracks on an echogram for each hour and range and (2) calculating integrated volume backscatter values per hour for each range. Both analysis methods showed similar trends in fish activity during the study period. Results agreed with expected behavioral patterns for the species in this body of water; fish activity increased during the night and decreased during the day, with the exception of a mid-day increase on one day.

3aED15. The influence of talker and accent variability on spoken word identification and discrimination. Kierra Villines, Tessa Bent, and Rachael F. Holt (Indiana Univ., 200 S. Jordan Ave., Bloominton, IN 47405, kvillines_08@yahoo.com)

In spoken word identification and memory tasks, stimulus variability from numerous sources impairs performance. The phonetic-relevance hypothesis (Sommers *et al*., 2006) proposes that only acoustic properties that influence phonetic perception (e.g., speaking rate) cause decrements in perception. In the current study, the influence of foreign-accent variability on identification and discrimination of spoken words was evaluated. In experiment 1, word identification in noise was tested in single-talker and two multiple-talker conditions: multiple talkers with the same accent or multiple talkers with different accents. In experiment 2, participants discriminated minimal pairs from a single talker, different talkers with the same accent, and talkers with different accents. Identification and discrimination performance was the highest in the single-talker conditions, but there was no difference between the single-accent and multiple-accent conditions. However, reaction time in the discrimination task was the highest in the multiple accent condition with no difference between the single-talker and single-accent conditions. Thus, the presence of multiple accents does not decrease accuracy beyond the multiple-talker effect, but processing time increases with the addition of multiple accents. These results provide partial support for the phonetic-relevance hypothesis. [Work supported by NIDCD R21DC010027 and Indiana University.]

3aED16. Application of time reversal to sources in non-ideal environments. Blaine Harker and Brian Anderson (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602)

Time reversal (TR) is a technique of localizing acoustical sources using a time reversal mirror (TRM) and is especially useful in reverberant environments. TR is commonly used to find acoustically small sources and in many cases pulsed waveforms are used. Here TR is applied to distributed sources using long duration waveforms. This is done using a straightforward, theoretical, point source propagation computer modeling and data from experimental measurements of jet noise. The quality of the TR focusing versus using different numbers of microphones to constitute the TRM is determined. These results are compared to other imaging techniques and theories of sound radiation from jets.

3aED17. On the directivity of musical instruments: Effects on room acoustics calculations. Kristina M. Sorensen, Nicholas J. Eyring II, Timothy W. Leishman, and Nathan G. W. Eyring (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, twleishman@byu.edu)

Measurements of musical instruments in an anechoic chamber at Brigham Young University are yielding high-resolution directivity data and balloon plots that may be analyzed and visualized as functions of time or frequency. Historically, room acoustics calculations involving directivity of sound sources have relied on well-defined radiation characteristics including principal radiation axes, directivity factors, beamwidths, etc. In recent times, room simulation software packages have incorporated more comprehensive steady state directivity data for loudspeakers that vary as functions of proportional frequency bands. However, in contrast to loudspeakers, many musical instruments have directivity patterns that are not well defined and that may vary more erratically as functions of pitch or other musical characteristics. This presentation explores the measured directivity properties of a few musical instruments and discusses how they might be encapsulated and incorporated into predictive room acoustics calculations.

Session 3aMU

Musical Acoustics: Expressivity in Digital Music Synthesis

James W. Beauchamp, Chair
School of Music, Univ. of Illinois at Urbana-Champaign, 1114 W. Nevada, Urbana, IL 61801

Chair's Introduction—8:00

Invited Papers

8:05

3aMU1. Extracting heterogeneous structured features from music performance. Gang Ren, Dave Headlam, Stephen Roessner, Justin Lundberg, and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Edmund A. Hajim School of Eng. and Appl. Sci., Univ. of Rochester, Rochester, NY 14627)

In music performance, the musician adds artistic expressive elements beyond the information contained in conventional western music scores. As micro-fluctuations these expressive dimensions convey emotions and essential interpretative information and can be measured and compared quantitatively, over large and small scales, and evaluated for their effect on aspects of the performance. We present a heterogeneous expressive music feature description that include both inter-note features that extend over musical phrases composed of several notes, and intra-note features that represent the internal variations within each musical note. The intra-note features include pitch and pitch deviation, dynamic level, timber, articulation and vibrato. The inter-note features include timing and dynamics, as well as timber, pitch deviation, articulation, and vibrato extending across multiple notes and musical phrases. A complete multi-dimensional feature description for every note is unnecessary because there is a hierarchy of note functions in a musical phrase and certain features may be more or less important depending upon their function in the phrase. Thus, we introduce a heterogeneous representation that affords the flexibility to reflect the hierarchical significance of each note or phase segment. Specific visualization and animation schemes of the proposed representation, to facilitate user interaction with the music, are also presented.

8:30

3aMU2. Synthesis of bowing controls applied to violin sound generation. Esteban Maestre (CCRMA–Dept. of Music, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, esteban@ccrma.stanford.edu)

Within instrumental sound synthesis, one of the most challenging aspects of automatic performance resides on the ability to faithfully reproduce the expressive nuances naturally conveyed by a musician when controlling a musical instrument. To that regard, much research effort has been devoted in the seek for obtaining natural-sounding synthetic performances from an annotated input score. Despite continuous improvements on sound synthesis techniques, appropriately mapping score annotations to synthesize controls still remains an interesting research problem, especially for excitation-continuous musical instruments. Along these lines, it is presented here our recent work on modeling bowing control in violin performance and its application to sound synthesis. Nearly, non-intrusive sensing techniques allow for accurate acquisition of relevant timber-related bowing control parameter signals. The temporal contours of bow velocity, bow pressing force, and bow-bridge distance are modeled as sequences of short Bzier cubic curve segments. A database of parametric representations of real performance data is used to construct a generative model able to synthesize bowing controls from an annotated score. Synthetic bowing controls are used to generate realistic performances using a digital waveguide physical model, and a spectral-domain sample-based synthesizer. Obtained results demonstrate the potential of explicitly modeling instrumental control for expressive sound synthesis.

8:55

3aMU3. Where is the expressivity in singing? Johan Sundberg (Dept. of Speech Music Hearing, School of Comput. Sci. and Commun., KTH, SE-10044, Stockholm)

When computers convert music scores to sounding music, performances void of musical expression typically emerge, mostly perceived as musical disasters. This is a striking illustration of the importance of the contributions of musicians. Excellent tools for exploring these contributions are synthesized performances and/or processing of real performances. In the 1970s, the KTH music group started to formulate rules for music performance, now integrated into the KTH Director Musices performance grammar. Some principles underlying its performance rules can be identified: (1) marking of phrase structure, (2) increasing contrasts along acoustic parameters, and (3) emphasizing important and/or less expected notes. These principles can be observed also in phonetics and speech prosody. Also the acoustic codes used for these principles in music performance are similar to those used in speech. Just as pitch is raised in agitated speech, singers have recently been found to increase contrast along the pitch dimension by sharpening the peak note in agitated phrases. In a recent experiment, audio processing software was used to quasi-correct the intonation of the peak tone in a handful of examples sung by a professional baritone singer. A listening test indicated that the sharpening added to the expressivity of the performance.

9:20

3aMU4. Synthesizing melodic expression. Christopher S. Raphael (School of Informatics and Computing, Indiana Univ., 901 10th St., Bloomington, IN 47408)

A method for expressive melody synthesis is presented seeking to capture the prosodic (stress and directional) element of musical interpretation. An expressive performance is represented as a note-level annotation, classifying each note according to a small alphabet of symbols describing the role of the note within a larger context. An audio performance of the melody is represented in terms of two time-varying functions describing the evolving frequency and intensity. A method is presented that transforms the expressive annotation into the frequency and intensity functions, thus giving the audio performance. The problem of expressive rendering is then cast as estimation of the most likely sequence of hidden variables corresponding to the prosodic annotation. Examples are presented on a data set of around 50 folk-like melodies, realized both from hand-marked and estimated annotations.

9:45

3aMU5. Virtual instrument player using probabilistic gesture grammar. Eric Lindemann (Synful, 2975 18th St. Boulder, CO 80304)

Expressive instruments such as violin require subtle control gestures: variations in bow velocity and pressure over the course of a note, changes in vibrato depth and speed, portamento gestures, etc. Music synthesizers that attempt to emulate these instruments are often driven from keyboard controllers or directly from score editor or sequencer software. These synthetic control sources do not generally provide the level of detailed control required by expressive instruments. Even if a synthesizer offers a rich set of realistic continuous control inputs, the effort and skill required by the user to manage these controls, e.g., by drawing expression and vibrato controller curves in a MIDI sequencer, is considerable. Often the user would prefer to treat the synthesizer as a combination instrument and virtual player. The virtual player receives limited score-like control input and infers the more detailed gestural control. This presentation describes a virtual player mechanism that employs a probabilistic note-gesture grammar to aid with control inference. The virtual player uses the grammar to select various gestural patterns that are idiomatically appropriate given the limited control input. The virtual player is used to drive a concatenative music synthesizer employing a rich database of recorded note gestures.

10:10–10:25 Break

10:25

3aMU6. Playing a virtual drum from a real one. Miller Puckette (Dept. of Music (0099), Univ. of California, San Diego, La Jolla, CA 92093-0099)

An electronic musical percussion instrument is described which can be played by tapping or otherwise playing on a small physical object, whose vibrations are picked up and used to activate the virtual one. To do this, a ceramic tile or other small rigid object is attached to a contact microphone. Any resonances of the physical system are filtered out using linear predictive analysis, so that the resulting residual signal approximates the excitation the player introduces into the system. The resulting audio signal could then be forward filtered to recover the tile's own sound, but instead it is fed into a nonlinear reverberator to simulate a variety of real or fanciful percussion instruments. The result is a highly expressive and playable electronic percussion instrument, that is, both easy and inexpensive to build. The effects of various design parameters of the nonlinear reverberator on the resulting sound are discussed.

10:50

3aMU7. Music from "an evil subterranean beast." Chris Chafe (CCRMA, Stanford Univ., Stanford, CA 94305)

Two recent compositions have capitalized on the great expressive range of a custom-designed music synthesis algorithm. The "Animal" plays a part in Tomato Quintet (a music installation) and Phasor (a contrabass and computer piece). The algorithm is played with live controls, respectively, updates from CO_2 sensors and signals from a sensor bow. Nuanced deflections of parameter values elicit characteristic and sometimes quirky behavior. The bounded set of sonic behaviors goes into making up an identifiable personality whose moods or temperaments can be traversed in the music which exploits it. "The computerized sounds were spacey and sometimes menacing, sounding at times like Chafe was trying to tame an evil subterranean beast." (H. Ying, Global Times, 2011). Animal's algorithm is comprised of the logistic map with two parallel waveguides in its feedback path. Animal can be categorized as a "meta-physical" model or modeling abstraction as seen in earlier integrations of families of physical models (P. Cook, ICMC, 1992) and recombinations of physical model components in physically impossible ways (C. Burns, Composition, 2003). Abstraction opens the door to inclusion of mathematical "parts" from other domains. In Animal's case, the logistic map has been borrowed from population biology.

Contributed Papers

11:15

3aMU8. Nonlinear coupling and tension effects in a real-time physical model of a banjo. Rolf Bader and Florian Pfeifle (Musicological Inst., Univ. of Hamburg, Neue Rabenstrasse 13, 20354 Hamburg, Germany, Florian.Pfeifle@haw-hamburg.de)

A real-time physical model of the Banjo as proposed by Pfeifle and Bader [J. Acoust. Soci. Am. (2009)] is extended with nonlinear effects on the membrane resulting from the added force of the bridge and the thereby arising anisotropic tension distribution on the membrane. It is shown that this effect directly influences the vibrational behavior of the physical model and yields a more realistic sound. In a second step a non-linear excitation mechanism of the string is presented, modeling the interaction between a standard metal banjo fingerpick and the string of the banjo. The pressure and velocity of the fingerpick are used as control parameters for the model. The whole model is implemented on a FPGA and can be played and controlled in real-time.

11:30

3aMU9. An active learning-based interface for synthesizer programming. Mark B. Cartwright and Bryan Pardo (Dept. of EECS, Northwestern Univ., 2145 Sheridan Rd., Tech. L359, Evanston, IL 60208, mcartwright@u.northwestern.edu)

Commercial software synthesizer programming interfaces are usually complex and tedious. They discourage novice users from exploring timbers outside the confines of "factory presets," and they take expert users out of their flow state. We present a new interface for programming synthesizers that enables both novices and experts to quickly find their target sound in the large, generative timber spaces of synthesizers. The interface does not utilize knobs and sliders that directly control synthesis parameters. It instead utilizes a query-by-example based approach combined with relevance feedback and active learning to create an interactive, personalized search that allows for exploration. We analyze the effectiveness of this new interface through a user study with four populations of varying types of experience in which we compare this approach to a traditional synthesizer interface. [Work supported by the National Science Foundation Graduate Research Fellowship.]

WEDNESDAY MORNING, 2 NOVEMBER 2011 GARDEN SALON 2, 7:40 A.M. TO 12:00 NOON

Session 3aNS

Noise and Committee on Standards: Impact of New Environmental Protection Agency (EPA) Regulation on Hearing Protection

William J. Murphy, Chair
National Inst. for Occupational Safety and Health, 4676 Columbia Pkwy., Cincinnati, OH 45226

Chair's Introduction—7:40

Invited Papers

7:45

3aNS1. Hearing protector labeling—Yes or no? Ken Feith (13404 Query Mill Rd., North Potomac, MD 20878, feithk@comcast.net)

The federal government has been accused of requiring needless product labeling in order to protect citizens from themselves. While there is probably some truth in these accusations, the key point of labeling is lost in the noise of criticism—at least in the case of hearing protector devices. If we were to consider the one word that most influences our life actions we would find it to be the word "decisions." A moment's thought would reveal that from the minute we wake in the morning until we achieve a deep state of sleep, we are confronted with the need to make decisions. Every move, every action, is determined by either conscious or subconscious decision making. To that end, we need to consider those elements that influence our decision making process—understandable and accurate information. Thus, the role of federal labeling of select products is to provide an understandable and accurate means for making decisions that may have an impact on our ability to function effectively in our work, maintain our good health, avoid personal injury, and the list goes on. This presentation reveals the secrets behind regulatory labeling.

8:15

3aNS2. Evaluation of hearing protection devices with high-amplitude impulse noise. Karl Buck, Sebastien DeMezzo, and Pascal Hamery (ISL, APC, 5 r. du Gn. Cassagnou, BP 70034, 68301 St. Louis, France, karl.buck@isl.eu)

Presently, the evaluation of HPDs (hearing protection devices) is mainly based on audiometric threshold methods. However, in the military environment soldiers may be exposed to impulse noise levels of 190 dB and higher if large caliber weapons or IEDs (improvised explosive devices) are considered. In this case it seems reasonable to expect that an evaluation done at hearing threshold might not represent the protection which will be encountered at these highest levels. Therefore, it is necessary to evaluate the HPDs with signals close to the effective exposure. ISL has developed procedures using explosive charges creating very high levels of impulse noise (up to 195 dB peak) for the evaluation of HPDs. ISL also has developed an artificial head that can withstand this type of exposure and has sufficient self-insertion loss for the measurement of double hearing protection. The presentation will give a description of the techniques used at the ISL. It will present some results showing the possibilities to determine the performance of different types of HPDs when subjected to extreme impulse noise levels. Moreover, the new version of the ISL artificial head, compatible ANSI-ASA-S12.42-2010, and possible problems when using it will be presented.

8:45

3aNS3. Comparison of three acoustics test fixtures for impulse peak insertion loss. William J. Murphy (Hearing Loss Prevention Team, NIOSH, 4676 Columbia Parkway MS C-27, Cincinnati, OH 45226, wjm4@cdc.gov), Gregory A. Flamme (Western Michigan Univ., Kalamazoo, MI 49008-5243), Deanna K. Meinke, Donald S. Finan (Univ. of Northern Colorado, Greeley, CO 80639), James Lankford (Northern Illinois Univ., DeKalb, IL 60115), Amir Khan (NIOSH, Cincinnati, OH 45226-1998), Jacob Sondergaard (G.R.A.S. Sound and Vib., North Olmsted, OH 44070), and Michael Stewart (Central Michigan Univ., Mount Pleasant, MI 48859)

Acoustic test fixtures (ATF) for testing the impulse peak insertion loss (IPIL) of a hearing protector are described by American National Standard ANSI S12.42-2010. The self-insertion loss, ear simulator design (canals, microphone, and temperature), hardness of the area surrounding the pinna, and the anthropometric shape of the head has been specified in the standard. The IPILs of four protector conditions were evaluated with three ATFs during an outdoor field study using firearm noise. The Etymotic Research ER20 musicians' earplug and electronic (EB1 earplugs), the Peltor Tactical Pro earmuffs, and a combination of the TacticalPro and ER20 protectors were

3a WED. AM

tested at 130, 150, and 170 dB peak sound pressure level with the Institute de Saint Louis heated and unheated fixture and the GRAS 45CB heated ATF. IPILs exhibited good agreement across all three fixtures for earplugs. Significant differences were observed between the fixtures for the earmuff-only condition. These differences were more evident for the double-protection condition. [Portions of this work were supported by the U.S. EPA Interagency Agreement DW75921973-01-0.]

9:15

3aNS4. Insertion-loss and transfer-function performance of two new acoustical test fixtures complying with ANSI S12.42-2010, relative to performance of prior test fixtures and to real-ear data. Elliott H. Berger, Ronald W. Kieper, and Michael E. Stergar (3M Occupational Health & Environ. Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268-1657)

The most recent American National Standards Institute (ANSI) standard for the measurement of the insertion loss of hearing protection devices (HPDs), ANSI/American Standards Association (ASA) S12.42-2010, specifies a new-concept acoustical test fixture (ATF). It is similar to some existing ATFs but differs in terms of the required earcanal length, inclusion of a simulated flesh lining the earcanal, and a heater to bring the test fixture to approximate body temperature. These features were deemed necessary to develop a device that provides insertion loss data with reasonable correspondence to performance on human heads, as the ATF is the preferred method in the standard for tests on certain electronic earplugs and for all impulse testing. Within a year of the issuance of the standard, at least two ATFs [one produced by G.R.A.S. Sound and Vibration and the other produced by the Institute of St. Louis] became available. The studies reported herein will provide an initial evaluation of these two heads compared to prior art, based on ATF insertion-loss measurements for a sample of passive earplugs and earmuffs versus real-ear attenuation at threshold per ANSI S3.19-1974. Additionally, the ATFs' transfer function of the open ear in a diffuse field will also be reported.

9:45

3aNS5. Attenuation performance of active noise reduction headsets using American National Standard s12.42 and the proposed U.S. Environmental Protection Agency Hearing Protector Labeling Regulation. Richard L. McKinley, Hillary L. Gallagher (Air Force Res. Lab., 2610 Seventh St. WPAFB, OH 45433, richard.mckinley@wpafb.usaf.mil), Melissa A. Theis (Oak Ridge Inst. for Sci. and Education, TN 37831), and Paul C. Schley (Ball Aerosp. and Technologies Corp., Fairborn, OH 45324)

The attenuation performance and noise reduction rating (NRR) of six commercially available active noise reduction (ANR) headsets was assessed using the proposed environmental protection agency (EPA) regulation. The passive attenuation results were collected using American National Standard Institute (ANSI) S12.6 method for measuring real-ear attenuation at threshold (REAT) of hearing protectors while the active attenuations results were collected using ANSI S12.42 methods for the measurement of insertion loss of hearing protection devices in continuous or impulsive noise using microphone-in-real-ear (MIRE) or acoustic test fixture procedures. ANSI/ASA S12.68 methods of estimating effective A-weighted sound pressure levels when hearing protectors are worn was used to compute noise reduction metrics including the noise reduction statistic A-weighted (NRSA) and the graphical noise reduction statistic (NRSG). The proposed NRR labels for the ANR headsets were computer per the guidance in the draft U.S. EPA regulation. The presentation will include the baseline passive, active, and total attenuation, the NRSA and the Graphical NRSG, and the proposed EPA labels for passive attenuation and total attenuation while in an active mode.

10:15–10:30 Break

Contributed Papers

10:30

3aNS6. 1.Comparison of the HPDLAB and REATMASTER software/hardware systems for ANSI S12.6 testing. David C. Byrne (NIOSH–Taft Labs., 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226, DByrne@cdc.gov), Caryn C. Perry (Univ. of Cincinnati, Cincinnati, OH 45267), and William J. Murphy (NIOSH–Taft Labs., Cincinnati, OH 45226)

The American National Standard Methods for Measuring the Real-Ear Attenuation of Hearing Protectors (ANSI S12.6-2008) requires a Békésy procedure for testing occluded and unoccluded thresholds. Since 2002, the National Institute for Occupational Safety and Health (NIOSH) has used the custom-designed HPDLAB software operating Tucker-Davis Technologies System 3 hardware. ViAcoustics, Nelson Acoustics, NASA, and NIOSH researchers recently developed REATMASTER which runs on National Instruments hardware in the LABVIEW environment. Ten subjects were trained by the experimenter on how to fit a passive earmuff and were qualified according to the requirements of ANSI S12.6-2008. The laboratory was configured such that diffuse sound field thresholds were tested with either the HPDLAB or REATMASTER hardware by flipping a toggle switch. The earmuff was not touched or re-positioned between test trials with the two different hardware/software systems. The test sequence for the order of open and occluded measurements was counterbalanced across occluded conditions and hardware system. Results from this testing were used to validate the REATMASTER system for its ability to produce accurate threshold data. Preliminary results indicate no significant differences between the two systems.

10:45

3aNS7. Calibration details for the impulse peak insertion loss measurement. William J. Murphy and Julia A. Vernon (Hearing Loss Prevention Team, NIOSH, 4676 Columbia Parkway MS C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The American National Standard ANSI S12.42-2010 specifies the measurement of hearing protector performance in the presence of impulse noise. A series of calibration impulses are recorded from an acoustic test fixture (ATF) and a field microphone for peak sound pressure levels of 130, 150, and 170 dB. The averaged acoustic transfer function between the ATF and field microphone is calculated as follows:

$$\overline{H}_{\overline{ATF}-\overline{FF}}(f) = \overline{\langle FFT(P_{\mathrm{ATF},i}(t))/FFT(P_{FF,i}(t))\rangle}.$$

The transfer function is computed for each of the ranges of impulse levels and is applied to the field microphone measurements to estimate the unoccluded fixture levels of the ATF when hearing protection is being tested. This method allows a comparison between occluded and unoccluded waveforms. The calibration transfer function is affected by the time-alignment of the field impulse peaks, time-windowing of the impulses, and compensation for any dc bias. Time-alignment significantly affected the accuracy of predicting individual calibration levels with $\overline{H}_{\overline{\mathrm{ATF}}-\overline{\mathrm{FF}}}$. The prediction error variance was less at 170 dB than at 130 dB impulses. The time-window was varied from 2.5 to 100 ms preceding the peak of the field impulse. [Portions

of this work were supported by the U.S. EPA Interagency Agreement DW75921973-01-0.]

11:00

3aNS8. Measuring, rating, and comparing the real ear attenuation at threshold of four earplugs. William J. Murphy, Mark R. Stephenson, and David C. Byrne (Hearing Loss Prevention Team, NIOSH, 4676 Columbia Parkway MS C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The effect of training instruction, whether presented as the manufacturer's printed instructions, a short video training session, specific to the product, or as a one-on-one training session, was evaluated using four hearing protection devices with eight groups of subjects. The Howard Leight Fusion and Airsoft premolded earplugs and the Moldex PuraFit and EAR Classic foam earplugs were tested. Naïve subjects were recruited and tested using three different forms of training: written, video, and individual training. The differences between group averages for A-weighted attenuation were not statistically significant when compared between the video or the written instruction conditions, regardless of presentation order. The experimenter-trained A-weighted attenuations were significantly greater than the written and video instruction for most of the protectors and groups. For each earplug, the noise reduction statistic for A-weighting (NRSA) and the associated confidence intervals were calculated for the 90th and 10th percentiles of protection. Across subject groups for each protector, the differences between NRSA ratings were found to be not statistically significant. Several comparisons evaluating the order of testing, the type of testing, and statistical tests of the performance across the groups are presented. [Portions of this work were supported by the U.S. EPA Interagency Agreement DW75921973-01-0.]

11:15

3aNS9. ANSI S12.42-2010 measurements of impulse peak insertion loss for passive hearing protectors. Kevin Michael (Michael & Assoc., Inc., 2766 W. College Ave., St. 1, State College, PA 16801, kevin@michaelassociates.com) and Jeff G. Schmitt (ViAcoust., Austin, TX 78745)

Until recently, a standardized measurement procedure to determine peak insertion loss for hearing protectors was not available. This has led to confusion and uncertainty for hearing protector users who commonly use the devices in impulse noise, such as gunfire. Released in 2010, ANSI S12.42-2010 defines a test method and analysis procedure for measuring hearing protector impulse peak insertion loss. The required test fixture has recently become commercially available and laboratories are gaining experience making these measurements. Impulse peak insertion loss data will be presented for a variety of hearing protector types along with a description of the measurement procedure.

11:30

3aNS10. Real-ear attenuation of custom-fit earplugs with the communications earplug (CEP). Elmaree Gordon and Efrem R. Reeves (USAARL, Acoust. Branch, 6901 Farrel Rd., P.O. Box 620577, Fort Rucker, AL 36362-0577, elmaree.gordon@us.army.mil)

Hearing loss is one of the most common occupational injuries in the Department of Defense. One reason is limited access to adequate hearing protection during combat operations. Even if traditional pre-formed foam earplugs are used during combat, approximately 15% of the military population remains unprotected because the pre-formed earplugs do not adequately fit extremely small ear canals, extremely large ear canals, or ear canals with sharp turns. Custom-fit earplugs provide a potential solution for this hard-to-fit population. However, producing traditional wax-dipped custom-fit earplugs in a combat environment is not always practical. Fortunately, methodology exists to scan ear impressions to create a digital data set, transmit this data to a remote manufacturer via the Internet, and fabricate a set of custom-fit earplugs. The purpose of this study was to investigate how the attenuation provided by custom-fit earplugs created using the digital scanning technique compares to the attenuation provided by custom-fit earplugs created using the traditional wax-dipped technique when used with the CEP. Results show the digitally scanned custom-fit earplugs provide significantly poorer attenuation than the traditional wax-dipped custom-fit earplugs. Comply Canal Tip foam earplugs were also evaluated and shown to provide significantly greater attenuation than custom-fit earplugs manufactured by either method.

11:45

3aNS11. Noise surveys and hearing protection at the Royal Canadian Mint plant. (Jimmy) Jianping Yan (Corporate Eng., Royal Canadian Mint, 320 Sussex Dr., Ottawa, ON, K1A 0G8, Canada)

Royal Canadian Mint is a Crown Corporation to produce circulation and non-circulation coins for Canada and other countries. Noise measurements have been conducted regularly for production environment and before any new equipment operation in Royal Canadian Mint, and to protect our employee from hearing loss is one of our health and safety issues applied in whole plant. The formula used in the noise measurement is introduced in this paper, and the noise calculations are also discussed for the continuous noise sources, intermittent noise sources, and the combinations of individual noise sources. Some noise survey results are listed and analyzed, and the related hearing protective methods in our plant are also described.

WEDNESDAY MORNING, 2 NOVEMBER 2011 ROYAL PALM 3/4, 7:45 A.M. TO 12:00 NOON

Session 3aPA

Physical Acoustics: Theoretical and Computational Advances

Bonnie Schnitta, Chair

SoundSense, LLC, 46 Newtown Ln., Ste. 1, East Hampton, NY 11937

Contributed Papers

7:45

3aPA1. Formulation and applications of an integral-equation approach for solving scattering problems involving an object consisting of a set of piecewise homogeneous material regions. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Res., 739 Calle Sequoia, Thousand Oaks, CA 91360)

The paper presents the formulation and selected applications of the surface integral equation approach for finding pressure, displacement, and traction fields in a complex object consisting of a set of piece-wise homogeneous regions characterized by different Lame' material parameters. Representative applications of the approach are presented, which involve finding pressure field distribution inside human inner ear treated as an inclusion embedded in the surrounding inhomogeneous material. (the embedding region is treated with volumetric integral equations with suitable coupling to the inclusion). The method uses a set of coupled elastodynamics integral equations for two unknown fields: displacement and traction at each interface, which allow us to find displacement and traction field distribution inside a general complex object composed of sub-regions with different material parameters. The matrix elements of strongly singular kernels appearing in the integral equations were converted, through suitable integration by parts, to equivalent forms involving weakly singular integrands having at most 1/r singularity. Applications of the method are presented in simulation of acoustic and elastic fields in a human head and in the cochlea region. [This work is supported by AFOSR.]

8:00

3aPA2. Analysis and extension of scattering from rigid infinite wedge. Ambika Bhatta, Timothy Pflanz, Charles Thompson, and Kavitha Chandra (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 Univ. Ave., Lowell, MA 01854)

A new representation of the scattered pressure of a point source due to infinite rigid wedge is derived by simplifying the Bessel functions in the modal solution and by implementing the image based representation of the source. The solution comprises of six impulses contributing to the reflected and transmitted pressure and an integral for the diffracted pressure. The inverse Fourier transform of the solution yields the time domain pressure and the asymptotic solution for higher frequency is evaluated and compared with solution obtained by steepest descent method. The analytical approach based on wedge diffraction is extended to evaluate the scattered pressure from finite rigid polygon with three vertices and edges. The computational results for the scattered pressure will be explained in detail.

8:15

3aPA3. Acoustic scattering strength of turbulence generated waves in a shallow water flow. V. Kirill Horoshenkov, Andrew Nichols, J. Simon Tait (School of Eng., Univ. of Bradford, Bradford, West Yorkshire, BD7 1DP, United Kingdom), and A. German Maximov (N. N. Andreyev Acoust. Inst., Moscow, Moscow 117 036, Russian Federation)

This work examines an airborne method of ultrasonic characterization of the dynamically rough water surface in an shallow water flow. A new experimental setup has been developed and tested in a 12 m long, 450 mm wide laboratory flume. This equipment includes a source of a 45 kHz airborne wave, microphone array, and an array of wave probes. It has been shown experimentally and theoretically that the directivity of the ultrasonic wave scattered by the dynamically rough water surface can be related directly to the roughness correlation function which can be measured non-acoustically. It is shown that the form of the roughness correlation function is determined unambiguously by the scale and intensity of the turbulence processes in the flow. The proposed acoustic method is accurate, remote, and rapid. It can be used to relate the spectral and statistical characteristics of the rough flow surface to the scale and intensity of the turbulent flow structures which cause the water surface to appear rough.

8:30

3aPA4. Distributed point source method and its applications in solving acoustic wave scattering problems. Tribikram Kundu, Ehsan Kabiri Rahani, and Talieh Hajzargerbashi (Dept. of Civil Eng. Eng. Mech., Univ. of Arizona, 1209 E. 2nd St., Tucson, AZ 85721)

A recently developed semi-analytical technique called distributed point source method is used for solving various wave scattering problems. Scattering of focused ultrasonic fields by air bubbles or cavities in fluid and solid media is investigated in this presentation. Results for both single and multiple cavity geometries are presented. It is investigated when two cavities in close proximity can be distinguished and when it is not possible. The interaction effect between two cavities prohibits simple linear superposition of single cavity solutions to obtain the solution for the two cavities placed in close proximity. Therefore, although some analytical and semi-analytical solutions are available for the single cavity in a focused ultrasonic field, those solutions cannot be simply superimposed for solving the two-cavity problem even in a linear elastic material. The comparison between the ultrasonic energies reflected from two small cavities versus a single big cavity is also investigated. Detection and characterization of cavities is important for both materials science and medical applications because air bubbles in molten metal as well as in blood must be first detected and characterized before taking necessary actions.

8:45

3aPA5. Effective medium theory applied to bubble scattering well below resonant frequency. Dale I. McElhone, Robert W. Smith, and R. Lee Culver (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

When ensonifying bubble assemblages using acoustic frequencies well below the resonant frequencies of the individual bubbles, resonant scattering theory for individual bubbles would suggest very low backscatter, whereas an "effective medium theory" predicts a result for backscattering that is orders of magnitude higher. For this case, a number of authors have suggested resonant scattering associated with the size of the bubble clusters, and backscatter measurements from bubble clouds have supported this hypothesis. Scattering can also be considered from a much larger bubble structure: a ship wake. At the stern, ship wakes contain a wide range of bubble sizes. Buoyancy quickly drives the largest bubbles to the surface, but small bubbles can persist for large distances. Ensonification at mid frequencies (1 kHz–10 kHz) therefore cannot rely on resonant bubble scattering. Using a generic wake model developed from the literature, acoustic backscatter can be modeled at frequencies below those of the resonant bubbles. The approach discretizes the spatial bubble distribution and sums the backscatter coherently, following the work of Commander and Prosperetti [JASA (1989)]. Ongoing work includes a lab-scale comparison of the backscattering approach applied to the ship wake model. Work sponsored by the Office of Naval Research, Code 321US.

9:00

3aPA6. Image reconstruction using singular-value decomposition of scattering operators. W. Jiang, J. Astheimer (Dept. of ECE, Univ. of Rochester, NY 14627, wejiang@ece.rochester.edu), and R. Waag (Dept. of ECE, Dept. of Imaging Sci., Univ. of Rochester, NY 14627)

Efficient inverse scattering algorithms for nonradial lossy objects are presented using singular-value decomposition to form reduced rank representations of the scattering operator. These algorithms extend eigenfunction methods that are not applicable to nonradial lossy scattering objects because the scattering operators for these objects are not normal. A method of local reconstruction by segregation of scattering contributions from different local regions is also presented. Scattering from each region is isolated by forming a reduced rank representation of the scattering operator that has domain and range spaces comprised of far-field patterns with retransmitted fields that focus on the local region. Accurate methods for the estimation of the boundary, average sound speed, and average attenuation slope of the scattering object are also given. These methods yield initial approximations of scattering objects that reduce the number of distorted Born iterations needed for high quality reconstruction. Calculated scattering from two lossy elliptical objects is used to evaluate the performance of the proposed methods. In both cases, the reconstruction procedures result in high-quality quantitative images of tissue parameters with sub-wavelength resolution.

9:15

3aPA7. Observations of Biot compressional waves propagating in snow. Donald G. Albert, Stephen N. Decato, and Frank E. Perron, Jr. (ERDC-CRREL, Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755)

Many laboratory measurements of compressional waves propagating in a porous material have been conducted, but similar measurements are less frequent outdoors. Here, we present velocity measurements of Biot compressional waves of the first and second kind propagating through a thick natural snow cover. Vertical sledge hammer blows and blank pistol shots were used to generate horizontally propagating pulses that were recorded by acoustic and seismic sensors. For both sources, seismic P1 compressional, shear, and Rayleigh waves are visible on the measured time series, followed by the airborne acoustic arrival. The acoustic arrival generates additional P1 ice frame waves and Biot P2 pore waves that propagate downward from the snow surface at measured velocities of 460 and 95 m/s, respectively. Theoretical calculations for a rigid porous medium show that the Biot P2 wave can account for the distortion and attenuation observed in the atmospheric acoustic wave. Separate small-scale laboratory measurements demonstrate that the P2 wave in snow is dispersed (the velocity increases with increasing

frequency) in agreement with the trend predicted by Biot's theory, but with some parameter adjustments needed to obtain the best agreement. [Work funded by U.S. Army basic research program.]

9:30

3aPA8. Refraction of the cyclonic microbarom signal by the cyclonic winds. P. Blom, R. Waxler, W. G. Frazier, C. Talmadge, and C. Hetzer (NCPA, Univ. of Mississippi, University, MS 38677, psblom@olemiss.edu)

Non-linear interaction of the ocean surface and atmosphere is known to produce narrow-band, low frequency, continuous acoustic and seismic radiation termed microbaroms and microseisms, respectively. The microbarom signal typically has an amplitude of a few microbars and a peak at 0.2 Hz. The microbaroms are generated by counter-propagating surface waves of equal period. The microbarom source location associated with a hurricane is believed to be due to the interaction of the waves produced by the cyclonic winds with the background ocean wave field and is generally located many kilometers from the eye of the storm, along a perpendicular to the direction of the ambient winds. Following up on a suggestion of Bedard and coworkers, propagation of the microbarom signal through the storm wind structure has been investigated using geometric acoustics in an inhomogeneous moving medium. Strong refraction of the signal is predicted. To observe this refraction we have deployed infrasound arrays along the US eastern seaboard. Predicted and measured back azimuths for propagation through the wind structure have been compared to the data recorded during the 2010 and 2011 Atlantic hurricane seasons.

9:45

3aPA9. Numerical examination of the impact of random terrain elevation and impedance variations on sound-field coherence. D. Ketih Wilson, Santosh Parakkal, Sergey N. Vecherin (U.S. Army Engineer Res. and Dev. Lab, 72 Lyme Rd., Hanover, NH 03755-1290), and Vladimir E. Ostashev (Univ. of Colorado, 325 Broadway, Boulder, CO 80305)

Like turbulent fluctuations in the atmosphere, variations in terrain elevations (surface roughness) and ground impedance may be regarded as diminishing the coherence of the sound field and introducing uncertainty into predictions. Here we examine, using a numerical approach, the characteristics and relative importance of terrain-elevation, ground-impedance, and turbulent variations on sound-field coherence. The calculations utilize parabolic equation solutions for a refractive atmosphere, with a Beilis–Tappert transform to incorporate terrain elevation variations and a phase-screen method for the turbulence. The relative importance of terrain variability and turbulence is studied for different refractive conditions. In particular, we examine whether surface roughness has a more significant impact on coherence in downward or upward refracting conditions. Interdependencies between the various mechanisms of coherence degradation are also investigated. The numerical calculations are compared to a recently developed theory for waveguide propagation conditions, which predicts existence of an effective spectrum with independent, additive contributions for turbulence and roughness. Limitations of the Markov approximation for modeling the elevation and impedance variations are also examined.

10:00–10:15 Break

10:15

3aPA10. Detection of turbulence aloft by infrasonic wind noise measurements on the ground. Jericho Cain and Richard Raspet (Jamie Whitten Natl. Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

Wind turbines can be damaged by the inflow of high amplitude wind turbulence. The goal of this project is to determine if potentially damaging turbulent structures at hub height can be detected by infrasonic wind noise measurements at the ground. A large eddy simulation is used to model the atmospheric turbulence structures of interest and the resulting pressure fluctuations on the ground. Preliminary studies of the relationship between the time development of the pressure structures on the ground and the velocity fluctuations in the atmosphere will be described. If elevated turbulent structures can be detected on the ground, the predictions will be used to design an optimized pressure sensing array for experimental tests.

10:30

3aPA11. A linearized contrast source method for full-wave modeling of nonlinear acoustic wave fields in homogeneous media. Martin D. Verweij, Libertario Demi, and Koen W. A. van Dongen (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, m.d.verweij@tudelft.nl)

Many novel medical echography modalities, like super harmonic imaging (SHI), employ imaging of higher harmonics arising from nonlinear propagation. Design optimization of dedicated imaging probes for these modalities requires accurate computation of the higher harmonic beam profiles. The existing iterative nonlinear contrast source method can perform such simulations. With this method, the nonlinear term of the Westervelt equation is considered to represent a distributed contrast source in a linear background medium. The full transient nonlinear acoustic wave field follows from the Neumann iterative solution of the corresponding integral equation. Each iteration involves the spatiotemporal convolution of the background Green's function with an estimate of the contrast source, and appropriate filtering enables a discretization approaching two points per smallest wavelength or period. To further reduce the computational load and to anticipate extension of the method, in this presentation, the effect of linearizing the nonlinear contrast source is investigated. This enables the replacement of the Neumann scheme by more efficient schemes. Numerical simulations show that for design purposes the effect of linearization on the harmonic components involved with SHI remains sufficiently small. Moreover, it is shown that a significant decrease in computational costs may be achieved by using a Bi-CGSTAB scheme.

10:45

3aPA12. Angular spectrum representations of focused acoustic beams and a comparison with the complex source point description. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814)

One approach to describing the wave fields of focused acoustic beams is to use an angular spectrum representation. This has the advantage of facilitating a simple evaluation of the scattering by a sphere placed at the focal point of the beam in terms of the scattering by a superposition of Bessel beam components. This was recently demonstrated for a quasi-Gaussian beam [Marston, J. Acoust. Soc. Am. **129**, 1773-1782 (2011)]. This approach also has the advantage of separating the propagating spectral components from the evanescent spectral components [Marston, J. Acoust. Soc. Am. (submitted)]. While, for most practical acoustic sources, the evanescent spectral components may be neglected, the spectral representation is helpful for explaining why in principle evanescent spectral components can be important in the production of extremely tightly focused beams. A tightly focused beam with a Gaussian profile at the focal plane is not as easily describable using the more widely investigated complex source point description of the focused wave field. In addition to a Gaussian beam, the propagating-component angular spectrum representation is easily applied to the lowest radial-order Laguerre-Gauss helicoidal acoustic beam in the usual case where the beam is not tightly focused. [Work supported by ONR.]

11:00

3aPA13. Attenuation of elastic waves due to wave-induced vorticity diffusion in porous media. Tobias M. Mueller (CSIRO Earth Sci. and Resource Eng., 26 Dick Perry Ave., Kensington, 6151 WA, Australia, tobias.mueller@csiro.au) and Pratap N. Sahay (CICESE, 22860 Baja California, Mexico)

A theory for attenuation of elastic waves due to wave-induced vorticity diffusion in the presence of randomly correlated pore-scale heterogeneities in porous media is developed. It is shown that the vorticity field is associated with a viscous wave in the pore space, the so-called slow shear wave. The latter is linked to the porous medium acoustics through incorporation of the fluid strain rate tensor of a Newtonian fluid in the poroelastic constitutive relations. The method of statistical smoothing in random media is used to derive dynamic-equivalent elastic wave numbers accounting for the conversion scattering process into the slow shear wave. The result is a model for wave attenuation and dispersion associated with the transition from viscosity- to inertia-dominated flow regime in porous media. It is also shown that the momentum flux transfer from the slow compressional into the slow shear wave is a proxy for the dynamic permeability in porous media. A dynamic permeability model is constructed that consists of an integral over the covariance function of the random pore-scale heterogeneities modulated by the slow shear wave. In a smooth pore-throat limit, the results reproduce the dynamic permeability model proposed by Johnson *et al*. [J. Fluid Mech. **176**, 379 (1987).]

11:15

3aPA14. Application of the Beilis and Tappert parabolic equation to long-range sound propagation over irregular terrain. Santosh Parakkal, Kenneth E. Gilbert, and Xiao Di (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

The Beilis and Tappert parabolic equation method is studied in the applications of sound propagation over irregular terrain. The exact ground impedance condition for porous ground is derived and applied to propagation over hills with slopes from 5° to 22°. It is found that for slopes less than approximately 20°; the flat-ground impedance condition is sufficiently accurate. For slopes greater than about 20°, the limiting factor on numerical accuracy becomes the narrow-angle approximation used in the Beilis and Tappert method. The generalized polar coordinate parabolic equation method (Polar-PE method) was developed partly to provide a very solid numerical benchmark to the Beilis and Tappert PE method.

11:30

3aPA15. A linearized contrast source method for full-wave modeling of nonlinear acoustic wave fields in media with strong and inhomogeneous attenuation. Libertario Demi, Martin D. Verweij, and Koen W. A. van Dongen (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, l.demi@tudelft.nl)

The iterative nonlinear contrast source (INCS) method is a numerical method originally developed for modeling transient nonlinear acoustic wave fields in homogeneous (i.e., spatially independent) lossless or lossy media. This method is based on the Westervelt equation and assumes that the nonlinear term describes a distributed contrast source in a linear background medium. Next, the equivalent nonlinear integral equation is iteratively solved by means of a Neumann scheme. In medical diagnostic ultrasound, the observed acoustic nonlinearity is weak, resulting in a weak nonlinear contrast source and a fast convergence of the scheme. To model spatially dependent attenuation, the Westervelt equation has recently been extended with an attenuative contrast source. This presentation deals with the situation of strong local attenuation. In that case, the attenuative contrast source is large and the Neumann scheme may fail to converge. By linearizing the nonlinear contrast source, the integral equation becomes linear and can be solved with more robust schemes, e.g., Bi-CGSTAB. Numerical simulations show that this approach yields convergent results for nonlinear acoustic fields in realistic media with strong local attenuation. Moreover, it is shown that in practical situations, the error due to the linearization remains sufficiently small for the first few harmonics.

11:45

3aPA16. A full wave method for modeling nonlinear acoustic wave fields in media with inhomogeneous wave speed, attenuation, and parameter of nonlinearity. Koen W. A. van Dongen, Libertario Demi, and Martin D. Verweij (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, k.w.a.vandongen@tudelft.nl)

The iterative nonlinear contrast source (INCS) method is a numerical method that has originally been developed for modeling transient nonlinear acoustic wave fields in homogeneous (i.e., spatially independent) lossless or lossy media. Starting from the Westervelt equation, the INCS method considers the nonlinear term of the latter as a nonlinear contrast source in an otherwise linear wave equation and iteratively solves the corresponding nonlinear integral equation using a Neumann scheme. By adding an attenuative contrast source, the method has recently been extended to deal with spatially dependent attenuation, and a spatially dependent parameter of nonlinearity has also been introduced. This presentation is about the introduction of an additional contrast source that account for a spatially dependent wave speed. In this case, convergence problems arise with the Neumann scheme. This problem is solved by replacing the Neumann scheme with a conjugate gradient scheme in which the error functional is based on the complete nonlinear integral equation. Results show that this approach yields very accurate results for the nonlinear wave field (tested up to the fifth harmonic) in media with a spatially dependent wave speed, attenuation, and parameter of nonlinearity, as encountered in realistic medical ultrasound applications.

Session 3aSCa

Speech Communication: The Voice Source: From Physiology to Acoustics

Abeer Alwan, Cochair
Dept. of Electrical Engineering, Univ. of California, Los Angeles, 405 Hilgard Ave., Los Angeles, CA 90095

Patricia A. Keating, Cochair
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Jody E. Kreiman, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

3aSCa1. Use of laryngeal high-speed videoendoscopy systems to study voice production mechanisms in human subjects. Daryush D. Mehta (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, One Bowdoin Sq., 11th Flr., Boston, MA 02114, daryush.mehta@alum.mit.edu), Matías Zañartu (Univ. Técnica Federico Santa María, Valparaíso, Chile), Thomas F. Quatieri (Massachusetts Inst. of Technol., Lexington, MA 02421), Dimitar D. Deliyski (Cincinnati Children's Hospital Med. Ctr., Cincinnati, OH 45229), and Robert E. Hillman (Voice Rehabilitation & Massachusetts General Hospital, Boston, MA 02114)

Advances in laryngeal high-speed videoendoscopy (HSV) are making it possible to investigate critical relationships between vocal fold physiology and acoustic voice production in human subjects. Our group has developed HSV systems for clinical research with synchronous acquisition of acoustics, electroglottography, neck skin acceleration, and, in the most comprehensive setup, airflow and intraoral pressure. Key results will be presented from examinations of source-filter coupling and studies of the acoustic impact of vocal fold vibratory asymmetry in subjects with and without voice disorders. Findings hold potential clinical significance by revealing acoustic-HSV relationships not observable using standard stroboscopic imaging, as well as contributing to the direct evaluation and eventual improvement of voice production models. The work of T. F. Quatieri was supported by the Department of Defense under Air Force contract FA8721-05-C-0002. The work of other authors was supported by grants from the NIH National Institute on Deafness and Other Communication Disorders (T32 DC00038 and R01 DC007640) and by the Institute of Laryngology and Voice Restoration.

8:25

3aSCa2. From high-speed imaging to perception: In search of a perceptually relevant voice source model. Abeer Alwan, Patricia Keating, and Jody Kreiman (Elec. Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, alwan@ee.ucla.edu)

The goal of this project is to develop a new source model from high-speed recordings of vocal fold vibrations and simultaneous audio recordings. By analyzing area waveforms and acoustics and performing perception experiments, we will better parameterize the new source model and also uncover those aspects of the model that are perceptually salient, leading to improved TTS systems. We recorded acoustic and high-speed video signals from eight speakers producing the vowel /i/. Each speaker produced four phonation types (breathy, modal, pressed, and creaky) and three pitches (low, normal, and high). Glottal area waveforms were extracted from the high-speed recordings, semi-automatically, on a frame-by-frame basis. The acoustic results indicated that phonation types are differentiated by a number of spectral and noise measures, including H1*-H2* and the harmonics-to-noise ratio. In addition, the different pitch levels were responsible for changes in quality within each phonation type, but such changes were subject to inter-speaker variability. Perceptual results highlight the importance of lower-frequency harmonics in voice quality perception. These studies are critical to our computational modeling efforts, because they help us understand which aspects of the source model are perceptually important. [Work supported in part by NSF and NIH.]

8:45

3aSCa3. The affect of epilarynx tube dimensions on glottal airflow. Ingo R. Titze (Natl. Ctr. for Voice and Speech, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306)

Since the early 1980s, it has been known that an inertive supraglottal acoustic load skews the peak of the glottal flow pulse. It also lowers the phonation threshold pressure. The skewing in turn affects the vocal intensity by increasing the rate of flow declination. High frequencies in the source spectrum are emphasized. More recent is the discovery that the airway between the vocal folds and the aryepiglottic folds, often called the epilarynx tube, provides much of the inertance in a vowel production. This presentation shows how epilarynx tube geometry skews the glottal flow pulse and thereby may affect voice quality.

9:05

3aSCa4. An *ex vivo* perfused human larynx for studies of human phonation. Gerald S. Berke, Scott Howard, and Abie Mendelsohn (Head and Neck Surgery, UCLA School of Medicine, 62-132 CHS, Los Angeles, CA 90095-6214, gberke@mednet.ucla.edu)

For many years, non-human animal models and excised human larynges have been used to study the physiology and biomechanics of human laryngeal function. While these studies have provided a wealth of information, experimental data concerning actual human laryngeal function have been limited by the inability to parametrically study the complete set of variables that controls vocal fold function in a living human. This presentation describes the development of a perfused *ex vivo* human larynx. In this model, a human larynx is harvested at the time of organ donation and perfused with the donor's blood to maintain viability of its nerves and muscles, which can then be stimulated as in an *in vivo* preparation. This model allows parametric, experimental control of the larynx (including stimulation of the thyroarytenoid muscle) while it is still virtually living. Applicability of this new laryngeal model to studies of physiology, biomechanics, and to treatment of human disease will be discussed.

9:25

3aSCa5. Voice quality as a correlate of stop coda voicing in developing speech. Helen M. Hanson (Elec. and Comput. Eng. Dept., Union College, 807 Union St., Schenectady, NY 12308, hansonh@union.edu) and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA 02139)

Previous work reported that two children from the Imbrie Corpus (2005) tend to produce noise at the end of the nuclear vowel in words with [-voice] stop codas (duck, cup), but not in words with [+voice] codas (tub, bug) [JASA **123**, 3320]. This noise resembles the preaspiration reported for adult speech in some languages. In this paper the earlier work is extended to the remaining eight children in this corpus of speech from 2- to 3-year old children learning American English. Furthermore, for nine subjects we compare their behavior to that of their primary female caregiver. Many of the children produce unvoiced coda stops that are coded as being strongly preaspirated. Their caregivers also produce vowel-final noise preceding [-voice] stops, but these tend to be coded as being weakly preaspirated. For both child and adult subjects preaspiration is more likely to occur for /k/ than for /p/. A few subjects produce irregular voicing rather than noise before [-voice] coda stops. More detailed acoustic measures such as durations and relative intensities will be presented. In addition, the acoustics are used to infer the physiology behind these voice quality differences between children and adults. [Work support from NIH R01DC00075, R01HD057606.]

9:45

3aSCa6. Hemilarynx experiments: Statistical analysis of vibrational parameters over changing glottal conditions. M. Doellinger (Dept. for Phoniatrics and Pedaudiology, Univ. Hospital Erlangen, Bohlenplatz 21, 91054 Erlangen, Germany, michael.doellinger@uk-erlangen.de) and D. A. Berry (The Laryngeal Dynam. Lab., Div. of Head & Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794)

In vitro (humans, animals) or *in vivo* (animals) hemilarynx experiments are still the only way to visualize and analyze 3-D vibrations of the entire medial surface of the vocal fold. However, to date, only a few studies have been performed. For three different larynges during sustained phonation, vibrational output was statistically analyzed as a function of glottal airflow (800 to 1800 ml/s), adductory forces (10, 50, 100 g), and two different anterior–posterior pre-stress conditions (10, 50 g). Using a homogeneous grid of 30 microsutures sewn along the medial surface of the vocal fold, global parameters (empirical eigenfunctions/EEF, f0, subglottal pressure, and sound intensity level) were computed. Similarly, local parameters (displacements, velocities, and accelerations) were analyzed at the 30 different suture locations. The recordings were obtained using a digital high-speed camera (2000 and 4000 fps). Increased airflow resulted in significant statistical changes in all parameter values except the empirical eigenfunctions. For increased adduction and a lower pre-stress (10 g), the local parameters increased more than for the higher pre-stress condition (50 g). For the two pre-stress conditions and an increased adductional force, most global parameters exhibited significant changes (f0, sound intensity, EEF2), while other global parameters (subglottal pressure) did not.

10:05–10:20 Break

10:20

3aSCa7. Electroglottographic and acoustic measures of phonation across languages. Patricia Keating and Jian-Jing Kuang (Dept. Linguist., UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu)

Many less-commonly studied languages use phonation contrastively: vowels, and therefore words, can differ in phonation type. Most such languages contrast two or three phonation categories. In other languages, a lexical tone (contrastive pitch) may typically have a redundant non-modal phonation. We have collected electroglottographic and audio recordings from speakers of six languages with one or both of these uses of phonation (Bo, Gujarati, Hani, White Hmong, Mandarin, Yi) and analyzed them with EGGWORKS. EGGWORKS is a free program which automates several EGG measures and works together with VoiceSauce, a free program which automates many acoustic voice quality measures. Comparisons of the EGG and acoustic measures within and across languages, and their correlations, will be presented. We have also applied functional data analysis to EGG pulse shapes and will describe the relations of the factors underlying the shapes to the EGG and acoustic measures. [Work supported by NSF.]

10:40

3aSCa8. The two-source problem: Producing voiced obstruents. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu)

The relationship between the physiological parameters and the acoustic output of the voice source is complex, and even more so when voiced obstruents are considered. Standard models assume that the subglottal pressure is divided across the glottis and the supraglottal constriction, which explains both the weaker noise source of voiced compared to voiceless fricatives and the tendency of voiced stops and fricatives to devoice. However, such models do not take into account small adjustments in vocal fold posture that may, for

example, decrease the phonation threshold pressure, or in the constriction that may act to optimize noise production. Larger adjustments are made for voiced fricatives, such as lowering the larynx and advancing the tongue root, that may also enhance a speaker's ability to produce both voice and noise sources at a given subglottal pressure. In this talk results from a wide range of studies will be considered to arrive at a more nuanced understanding of the interaction of two source mechanisms in voiced obstruents.

Contributed Papers

11:00

3aSCa9. The formation of flow separation vortices inside the glottis during vocal fold closing. Liran Oren, Sid Khosla, and Ephraim Gutmark (Dept. Otolaryngol., Univ. of Cincinnati, P.O. Box 670528, Cincinnati, OH 45267, orenl@mail.uc.edu)

Our previous work highly suggests that glottal airflow contains certain vortical structures, especially flow separation vortices (FSV) that significantly contribute to acoustics. We have previously identified these FSV directly above the folds but not between the folds. We have now developed a technique that allows us to see both intraglottal velocity fields and the medial wall of the folds during vocal fold closing. In this study, we used the new technique in an excised canine larynx to simultaneously measure intraglottal velocity fields (using high speed particle image velocimetry) and the medial surface of the folds. We observed the formation of FSV inside the glottis. Intraglottal pressures were calculated from measured intraglottal velocity fields using the Navier–Stokes equations. The maximum negative intraglottal pressure on the medial surface of one fold was −6 cm H2O. The circulation of the transglottal flow was highly correlated to the maximum closing speed (MCS) (r=0.98, $p<0.001$), and had good correlation to the spectral slope (r=-0.8, $p<0.01$) and HNR (Harmonic to Noise Ratio) (r=0.85, $p<0.005$). These results substantiate our hypothesis and our new technique is significant because it will allow us to experimentally characterize the intraglottal flow-structure relationship during vocal fold closing.

11:15

3aSCa10. Simulation of the coupling between vocal-fold vibration and time-varying vocal tract. Yosuke Tanabe (Automotive Product Res. Lab., Hitachi America, Ltd., 34500 Grand River Ave., Farmington Hills, MI 48335, yosuke.tanabe@hitachi-automotive.us), Parham Mokhtari, Hironori Takemoto (Natl. Inst. of Information and Commun. Technol., Kyoto 619-0288, Japan), and Tatsuya Kitamura (Konan Univ., 8-9-1, Hyogo 658-8501, Japan)

Acoustic coupling between the voiced sound source and the time-varying acoustic load during phonation was simulated by combining the vocal-fold model of [S. Adachi *et al.*, J. Acoust. Soc. Am **117**(5) (2005)] with the vocal-tract model of [P. Mokhtari *et al.*, Speech Commun. **50**, 179–190 (2008)]. The combined simulation model enables to analyze the dynamic behavior of the vocal folds due to the time-varying shape of the vocal tract. An example of vocal-fold behavior is shown for sustained phonation of the Japanese vowel sequence "aiueo." [This research was partly supported by Kakenhi (Grant Nos. 21500184, 21300071).]

11:30

3aSCa11. Restraining mechanisms in regulating glottal closure during phonation. Zhaoyan Zhang (UCLA School of Medicine, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA, 90095-1794)

Recent experimental studies showed that isotropic vocal fold models were often blown wide apart and thus not able to maintain adduction, resulting in voice production with noticeable breathy quality. This study showed that the capability of the vocal fold to resist deformation against airflow and maintain position can be improved by stiffening the anterior-posterior tension in either the body-layer or the cover-layer of the vocal folds, which presumably can be achieved through the contraction of the thyroarytenoid and cricothyroid muscles, respectively. Experiments in both physical models and excised larynges showed that, when these restraining mechanisms were activated, the vocal folds were better able to maintain effective adduction, resulting in voice production with much clearer quality and reduced breathiness. [Work supported by NIH.]

11:45

3aSCa12. The effects of age on speech production in a non-pathological system: A case study of 48 years. Eric J. Hunter, Wesley R. Brown, Patrick Pead, and Megan Engar (Natl. Cntr for Voice and Speech, Univ. of Utah, 136 S. Main St., #320, Salt Lake City, UT, 84101, eric.hunter@utah.edu)

The current study examines 36 recordings (mean: 30 min) of a male speaker, spanning the years 1958 to 2007. These recordings provide a rare opportunity to track a single individual's age-related voice and speech mechanism changes (48–98 y/o). Aging effects became noticeable between the ages of 68–74, indicating a fundamental change in the body's maintenance of the speech mechanism, implying the likely onset of other aging symptoms such as swallowing problems. Voice F_0, which usually begins to drop during puberty, continued to decrease for this individual until 70 y/o; from 70 to 98 y/o, average vocal F_0 then increased from 140 to 160 Hz. Rate of speech (syll/min) began to decrease precipitately at 78 y/o (175 to 165 per/min.), while length of speech breathing reduced from 70 y/o (35 to 25 phonemes per breath group). The results of this case study can be used as a baseline for future studies. The aging of the voice and speech mechanism affects breathing, swallowing, and communication. The world's 50+ population is the fastest growing segment, affecting society by its sheer number as well as by historically high life spans. Healthcare practitioners must understand and accommodate the needs of this population.

Session 3aSCb

Speech Communication: Production, Processing, Acquisition, and Hearing Among Typical and Atypical Populations (Poster Session)

Sam Tilsen, Chair
Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

3aSCb1. Speaker adaptation in infancy: The role of lexical knowledge. Marieke van Heugten and Elizabeth K. Johnson (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd. N., Mississauga, ON L5L1C6, Canada, marieke.vanheugten@utoronto.ca)

The acoustic realization of words varies greatly between speakers. While adults easily adapt to speaker idiosyncrasies, infants do not possess mature signal-to-word mapping abilities. As a result, the variability in the speech signal has been claimed to impede their word recognition. This study examines whether exposure to a speaker may allow infants to better accommodate that speaker's accent. Using the Headturn Preference Procedure, 15-month-olds were presented with lists containing either familiar (e.g., ball) or unfamiliar words (e.g., bog). In experiment 1, these words were produced in infants' own accent (Canadian English); in experiment 2, they were produced in a foreign accent (Australian English). Comparable to previous work (Best *et al*., 2009), only infants presented with their own accent preferred to listen to familiar over unfamiliar words. Thus, without access to speaker characteristics, word recognition is limited to familiar accents. In experiment 3, the same Australian-accented stimuli were preceded by exposure to the Australian speaker. Speaker adaptation tended to correlate with the infants' vocabulary size, with greater vocabularies being indicative to more robust adaptation. We are currently testing whether vocabulary size is a mediating factor caused by general processing abilities or whether speaker adaptation is lexically driven.

3aSCb2. Social accountability influences phonetic alignment. Holger Mitterer (Max Planck Inst. for Psycholinguistics, P.O. Box 310, 6500 AH Nijmegen, The Netherlands, holger.mitterer@mpi.nl)

Speakers tend to take over the articulatory habits of their interlocutors [e.g., Pardo, JASA (2006)]. This phonetic alignment could be the consequence of either a social mechanism or a direct and automatic link between speech perception and production. The latter assumes that social variables should have little influence on phonetic alignment. To test that participants were engaged in a "cloze task" (i.e., Stimulus: "In fantasy movies, silver bullets are used to kill ..." Response: "werewolves") with either one or four interlocutors. Given findings with the Asch-conformity paradigm in social psychology, multiple consistent speakers should exert a stronger force on the participant to align. To control the speech style of the interlocutors, their questions and answers were pre-recorded in either a formal or a casual speech style. The stimuli's speech style was then manipulated between participants and was consistent throughout the experiment for a given participant. Surprisingly, participants aligned less with the speech style if there were multiple interlocutors. This may reflect a "diffusion of responsibility:" Participants may find it more important to align when they interact with only one person than with a larger group.

3aSCb3. The influence of socioindexical expectations on speech perception in noise. Kevin B. McGowan (Dept. of Linguist., Rice Univ., 6100 Main St., Houston, TX 77005, clunis@umich.edu)

Previous research has shown that listener percepts can be altered by the manipulation of listener expectations of speaker identity [e.g., Niedzielski (1999), Hay *et al.* (2006)]. This study examines the extent to which socioindexical information can be *useful* to listeners. English speakers with little or no Chinese experience and Heritage Mandarin English speakers transcribed Chinese-accented speech in noise. In a between-subjects design, listener expectations about speaker identity were manipulated by presenting an Asian face, a socioindexically uninformative silhouette, or a Caucasian face as the purported speaker of high and low predictability sentences. Consistent with previous findings, listeners presented with an Asian face were significantly more accurate transcribers than those presented with a Caucasian face. Intriguingly, the silhouette condition patterned with the Asian face for experienced listeners but with the Caucasian face for inexperienced listeners. Unexpectedly, inexperienced listeners, while overall less accurate than experienced, Heritage listeners, saw the same magnitude improvement with the Asian face. Furthermore, transcription errors are inconsistent with suggestions (e.g., Staum Casasanto, 2009; Johnson, 2006) that listeners will alter base activations of prelexical or lexical forms to accommodate an expected accent. Implications for theories of speech perception and word recognition will be explored.

3aSCb4. Attention modulates the time-course of talker-specificity effects in lexical retrieval. Rachel M. Theodore and Sheila E. Blumstein (Dept. of Cognit., Linguistic & Psychol. Sci., Brown Univ., P.O. Box 1821, Providence, RI 02912, rachel_theodore@brown.edu)

Listeners retain in memory phonetic characteristics associated with individual talkers. It has been proposed that talker-specificity effects, while robust, emerge late in processing, reflecting increased time to activate lower-frequency instantiations for individual talkers. Here we test the hypothesis that attention to talker characteristics modulates the time-course of talker-specificity effects. Two groups of listeners participated in encoding and recognition phases. During encoding, all listeners heard words produced by two talkers, but only half of them were required to identify talker gender. During recognition, all listeners heard a series of words and indicated whether each word had been presented during encoding. Critically, some of the words were previously presented in the same voice and some in a different voice. Results showed no difference in mean hit rate or reaction time between the two groups, indicating that the encoding manipulation did not globally affect recognition memory or response latency. However, the latency data showed an interaction such that a talker-specificity effect emerged only for listeners who identified talker during encoding. These results indicate that the talker-specificity effect reflects an attention shift,

not increased processing time, consistent with a dynamic perceptual system where attention to specific aspects of the signal can influence representational frequency.

3aSCb5. Lexical-neighborhood and competing-task effects on word recognition and word recall by native and non-native listeners. Catherine L. Rogers, April M. Frenton, and Heather A. Hoefer (Dept. of Comm. Sci. and Dis., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620)

In spoken-word recognition, high-frequency words with few and less frequently occurring minimal-pair neighbors (lexically easy words) are recognized more accurately than low-frequency words with many and more frequently occurring neighbors (lexically hard words). This easy-hard word effect has been found to be larger for non-native listeners with a relatively late age of immersion in an English-speaking environment. Previous research found no effect of a competing digit-recall task on spoken-word recognition and no effect of listener group or listening environment on digit-recall accuracy. The present study compares word recognition by native English-speaking listeners and non-native listeners with either earlier (age 10 or earlier) or later (age 14 or later) ages of immersion in an English-speaking environment. Spoken word lists composed of equal numbers of lexically easy and hard target words were presented in an open-set word-identification task. Spoken words were presented in quiet and in moderate background noise and preceded by a list of zero, three, or six visually presented words, which listeners were asked to recall following the spoken word-recognition task (competing task). The size of the easy-hard word effect, spoken-word recognition accuracy competing-task word-recall accuracy, and word-entry response time will be compared across listener groups and listening conditions.

3aSCb6. Reading between the turns: Social perceptions of turn-taking cues in conversation. Marisa Tice and Tania Henetz (Dept. of Linguist., Margaret Jacks Hall, Bldg. 460, Stanford Univ., Stanford, CA 94305-2150, middyp@stanford.edu)

The perception of turn-taking cues provides an alluring new avenue for speech perception research since it requires complex cognitive and social processes that speakers must perform online during conversation. This study investigates whether inter-speaker gap duration, overlap, and backchanneling play a role in the social perception of speech during conversation. In a matched-guise task, participants heard clips of spontaneous, dyadic conversation. Clips were phonetically manipulated to control for the three turn-taking cues of interest by stretching and shrinking gap durations and replacing the naturally occurring backchanneling with silence. Participants then rated the speakers and conversations on a range of social measures, including dominance, formality, and similarity. Initial results indicate that listeners' social judgments are sensitive to these cues. For example, the removal of backchanneling in a male–male conversation resulted in ratings of higher speaker interest and lower speaker similarity, running contrary to some previous descriptions of backchannels as supportive signals (Yngve, 1970). This effect may interact with gender, such that backchanneling is interpreted differently in female-female or mixed-gender dyads, supporting Pearson *et al.*'s (2008) finding that turn-timing effects interact with (mis)matches in speaker race. It is clear these judgments follow from a complex interaction of subtle acoustic cues, social information, and context. [This work was supported by a National Science Foundation Graduate Research Fellowship to M. Tice.]

3aSCb7. Use of waveform mixing to synthesize a continuum of vowel nasality in natural speech. William F. Styler IV, Rebecca Scarborough, and Georgia Zellou (Dept. of Linguist., Univ. of Colorado, 295 UCB, Boulder, CO 80309-0295)

Studies of the perception of vowel nasality often use synthesized stimuli to produce controlled gradience in nasality. To investigate the perception of nasality in natural speech, a method was developed wherein vowels differing naturally in nasality (e.g., from CVC and NVN words) are mixed to yield tokens with various degrees of nasality. First, monosyllables (e.g., CVC, NVC, CVN, NVN) matched for vowel quality and consonant place of articulation were recorded. The vowels from two tokens were excised, matched for amplitude, duration and pitch contour, and then overlaid sample-by-sample according to a specified ratio. The resulting vowel was spliced back into the desired consonantal context. Iterating this process over a series of ratios produced natural-sounding tokens along a continuum of vowel nasality. Acoustic measurements of the nasality of output tokens [using A1-P0 (Chen, 1997)] confirmed a relation between the ratio used and the nasality of the output. Stimuli created in this manner were used in a perception experiment (Scarborough *et al.*, 2011) where degree of nasality affected perception without any slowdown in response time for nasality-altered vs. unaltered tokens. This demonstrates the quality of the nasal-altered stimuli and suggests the potential usefulness of this process for other speech perception studies.

3aSCb8. Formant frequencies, vowel identity, and the perceived relative tallness of synthetic speakers. Santiago Barreda and Terrance M. Nearey (Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB, Canada T6G 2E7, sbarreda@ualberta.ca)

Listeners can make consistent judgments regarding the tallness of speakers [Rendall *et al.*, J. Exp. Psych: Human Percep. Perform. **33**, 1208 (2007)]. These judgments are informed by the f0 and formant frequencies (FFs) of a speaker's voice. However, FFs are also cues to vowel identity, such that a small speaker producing an /u/ might have lower average FFs than a larger speaker producing an /æ/. Do listeners use absolute FFs to judge speaker tallness, or do they "correct" for phonetic identity and consider the FFs of a vowel relative to those expected for that category? To test this, a series of synthetic vowels (/i æ u/) with different FF scalings and different numbers of formants (2–5) were created. These scalings were intended to replicate speakers of different vocal tract lengths (i.e., sizes). Participants were presented with pairs of vowels and asked to indicate which vowel sounded like it had been produced by a taller speaker. Results indicate that listeners consider both absolute and phonetically "corrected" FF information and that formants higher than F3 greatly reduce listener's reliance on absolute F1 and F2 information in making speaker size judgments.

3aSCb9. Acoustic variability affects asymmetry in infant speech discrimination. Stephanie Archer (Dept. of Linguist., Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2E 0L7 Canada) and Suzanne L. Curtin (Univ. of Calgary, NW, Calgary, AB))

From birth, infants are capable of discriminating speech sounds that occur cross-linguistically, but by their first year, infants have difficulty discriminating most non-native contrasts (Werker and Tees, 1984). Yet, language experience is not the only factor affecting discrimination. Further research into language-specific perception reveals asymmetries occur in the discrimination of vowels according to perceptual space (Polka and Bohn, 2003) and consonants based on frequency in the input (Anderson *et al.*, 2003). It might be that acoustic variability also causes asymmetries in infant speech perception. Seventy-six 6- and 9-month olds participated in a discrimination task comparing bilabial and velar stop - /l/ onsets to unattested coronal stop - /l/ onsets (e.g. /kla/ - /tla/; /pla/ - /tla/) in both voiced and voiceless conditions. Infants successfully discriminated coronal from bilabial onsets ($p<0.05$), but not velar ($p>0.05$), with no effects of age or voicing. Adult productions of the attested clusters were subjected to an acoustic analysis. Of the four tested characteristics, higher standard deviations were found in velar onsets (/kla/ > /pla/: F2 onset and liquid duration; /gla/ > /bla/: F2 slope, VOT, and liquid duration). This suggests that acoustic variability in the input affects infants' speech discrimination.

3aSCb10. Adapting to foreign-accented speech: The role of delay in testing. Marijt J. Witteman, Neil P. Bardhan, Andrea Weber (MPI for Psycholinguistics, Wundtlaan 1, 6500AH Nijmegen, The Netherlands, marijt.witteman@mpi.nl), and James M. McQueen (Behavioural Sci. Inst., Radboud Univ. Nijmegen, 6500HE, The Netherlands)

Understanding speech usually seems easy, but it can become noticeably harder when the speaker has a foreign accent. This is because foreign accents add considerable variation to speech. Research on foreign-accented speech shows that participants are able to adapt quickly to this type of variation. Less is known, however, about longer-term maintenance of adaptation. The current study focused on long-term adaptation by exposing native listeners to foreign-accented speech on Day 1, and testing them on comprehension of the accent one day later. Comprehension was thus not tested immediately, but only after a 24 h period. On Day 1, native Dutch listeners listened to the speech of a Hebrew learner of Dutch while performing a phoneme monitoring task that did not depend on the talker's accent. In particular, shortening of the long vowel /i/ into /ɪ/ (e.g., lief [li:f], 'sweet', pronounced as [lɪf]) was examined. These mispronunciations did not create lexical ambiguities in Dutch. On Day 2, listeners participated in a cross-modal priming task to test their comprehension of the accent. The results will be contrasted with results from an experiment without delayed testing and related to accounts of how listeners maintain adaptation to foreign-accented speech.

3aSCb11. The interrelation between perceived and produced vowel spaces in Brazilian and European Portuguese. Katerina Chladkova (Amsterdam Ctr. for Lang. and Commun., Univ. of Amsterdam, Spuistraat 210, 1012VT Amsterdam, The Netherlands, k.chladkova@uva.nl) and Paola Escudero (MARCS Auditory Labs., Univ. of Western Sydney, Milperra, NSW 2214, Australia)

Previous research has shown that the interrelation between the perception and production of vowels is not straightforward. For instance, it has been shown that the vowel spaces of Spanish and Czech may have different structures in production than in perception (Boersma and Chladkova, 2011). This study compares vowel perception and production of 40 speakers in Brazilian- (BP) and European Portuguese (EP). Productions were elicited in a sentence-reading task and were reported in Escudero *et al.* (2009). Perception was tested by means of a phoneme identification task with tokens sampled from the whole vowel space. Perceived and produced vowel spaces were compared through the location of vowel categories (the vowels' F1 and F2 averaged across speakers/listeners) and their boundaries with neighboring categories. The results show that the perceived vowel space of BP listeners differs from that of EP listeners. Specifically, low-mid vowels have a higher F1 in BP than in EP, a finding that is in line with BP and EP vowel production differences. However, BP low-mid vowels are less peripheral than EP vowels along the F2 dimension only in perception. Other dialectal differences as well as the degree of correspondence between the production and perception of Portuguese vowels are discussed.

3aSCb12. Phonetic imitation by school-age children. Kuniko Nielsen (Dept. of Linguist., Oakland Univ., Rochester, MI 48309, nielsen@oakland.edu)

Phonological representations such as phoneme and feature are often assumed in prevailing linguistic theories. However, little is known about how these representations are formed during the course of language acquisition. The current study aims to investigate developmental changes of phonological representations by examining how children imitate fine phonetic details of recently heard speech. Previous studies have shown that adult speakers implicitly imitate the phonetic properties of recently heard speech (e.g., Goldinger, 1998; Pardo, 2009). Recently, Nielsen (2011) showed that phonetic imitation was generalized at phoneme and feature levels, providing support for sub-lexical representations. The current study extends these findings and examines whether school-age children manifest similar patterns of phonetic imitation. The experiment employed the imitation paradigm in which participants' speech is compared before and after they are exposed to model speech with extended VOT on the target phoneme /p/. A preliminary analysis of data revealed that participants produced longer VOTs after being exposed to model speech with extended VOTs. The change was generalized to new instances of the target phoneme /p/ and the new phoneme /k/, indicating that sub-lexical units were also involved in phonetic imitation by children.

3aSCb13. Classification of foreign accents. Eriko Atagi and Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloominton, IN 47405, eatagi@indiana.edu)

Regional dialects and foreign accents account for a significant source of between-talker variability. The auditory free classification task—in which listeners freely group talkers based on audio samples—has been a useful tool for examining listeners' cognitive representations of regional dialects (e.g., Clopper and Pisoni, 2007) and was employed here to examine perceptual representation of foreign accents. This task assesses listeners' perception of variation without imposing category structure. In the present study, native listeners completed four free classification tasks in which they grouped 24 talkers from six native language backgrounds by perceived native language. The four tasks varied in overall sentence intelligibility and whether four native talkers were also included, to explore how accent level and presence of native language exemplars influence classification. Categorization performance was less accurate with lower intelligibility sentences, even when sentence content was known, but more accurate when native talkers were included. Thus, listeners attend to phonological features that characterize differences across foreign accents, and are more attentive to these features when accented and native speech can be directly compared. Moreover, perception of lower intelligibility sentences may impose a higher cognitive load, thus restricting the cognitive resources available for classification. [Work supported by NIDCD R21DC010027.]

3aSCb14. Does second language phonological acquisition affect native language word recognition? Melinda D. Woodley (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720-2650, melinda.woodley@berkeley.edu), Marisa Tice, and Meghan Sumner (Stanford Univ., Stanford, CA 94305-2150)

Research with monolingual adults has shown that words whose initial stop consonants have canonical VOTs show larger semantic priming effects than words beginning with phonetically non-canonical stops; the prime [k^h] (long-lag) facilitates recognition of a semantically related target (*queen*) more than [k] (short-lag), though both facilitate target recognition (Andruski *et al.*, 1994; van Alphen and McQueen, 2006). The present study asks how the acquisition of a second (L2) phonological system affects word recognition in the native language (L1). A longitudinal semantic priming study ([k]... *queen*) is being administered to American students in Paris at the beginning, middle, and end of their 4–6 weeks of immersion class enrollment. Two potential patterns may emerge: (1) exposure to instances of French short-lag /k/ may lead to increased facilitation of short-lag [k] primes, indicating an early established link between shared L1 and L2 phonological categories or (2) since stops in the short-lag VOT range are categorized differently in French versus English ([k] = /k/ vs. /g/), increased ambiguity introduced by the L2 system may incur a greater cost for non-canonical variants, causing [k] to become a less effective prime. Both findings have implications for theories of speech perception and representation.

3aSCb15. When do which sounds tell you who says what? A phonetic investigation of the familiar talker advantage in word recognition. Stephen J. Winters (Dept. of Linguist., Univ. of Calgary, SS 820, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, swinters@ucalgary.ca)

This study investigated whether voice quality, which is a highly salient cue to talker identity, facilitates the process of word recognition for familiar talkers. Two groups of listeners were trained to identify the same set of talkers. One group of listeners heard these talkers speaking in a variety of voice qualities (modal, creaky, and breathy), while another group of listeners heard each talker speaking in only one voice quality. Both groups of listeners were then tested on their ability to identify words spoken by these familiar talkers—in a variety of voice qualities—and also words spoken by a group of unfamiliar talkers. Results showed that voice quality had a significant effect on word recognition scores. Both groups also exhibited better word recognition scores for familiar talkers. However, the familiar talker advantage in word recognition did not depend on voice quality; the familiar voice qualities of familiar talkers did not produce better word recognition scores than the unfamiliar voice qualities of familiar talkers. These combined results suggest that any integration of linguistic and indexical information in word recognition emerges from a subset of the phonetic features of the speech signal and is not, therefore, a strictly general property of speech perception.

3aSCb16. All representations are not equal: Differential activation of words and phonological variants across accents. Meghan Sumner and Reiko Kataoka (Dept. of Linguist., Stanford Univ., Stanford, CA 94305-2150, sumner@stanford.edu)

Given the massive variation in natural speech, how listeners recognize words is a central issue for linguistic theory. Listener sensitivity to acoustic fluctuations in speech has provided us with one piece of the puzzle: detailed, specific representations. We provide another piece: differential activation of these representations. We conducted a semantic priming with General American listeners of positively viewed non-rhotic British English accents and negatively viewed non-rhotic New York City accents (slender'THIN; slend-UH(BE)'THIN; slend-UH(NYC)'THIN). Controlling participant experience and self-reported familiarity with both accents, only the words produced in a British accent primed semantically related targets. There is no *a priori* reason to expect this pattern, as the phonological variant and lexical items are controlled. We suggest that two variants equally experienced in number (raw frequency) are perceived differentially because positive and negative attitudes influence the activation lexical representations. Additionally, by examining particular participant groups from non-rhotic regions, we show an oscillation between effects of raw frequency and listener attitudes. This work (1) provides a broader understanding of factors influencing activation, (2) increases our understanding of how frequency effects are modulated by other factors (subjective listener attitudes), and (3) illuminates the interactive nature of linguistic and social factors in speech perception.

3aSCb17. Recognition memory for speech of varying intelligibility. Kristin J. Van Engen, Bharat Chandrasekaran, Lauren Ayres, Natalie Czimskey (Dept. of Linguist., Univ. of Texas, 1 Univ. Station B5100, Austin, TX 78712, k-van@mail.utexas.edu), and Rajka Smiljanic (Dept. of Linguist., Univ. of Texas, 1 Univ. Station B5100, Austin, TX 78712)

Previous research has demonstrated that the internal representation of spoken words includes both a phonetic description and information about the source characteristics of specific talkers. Spoken utterances also vary greatly in intelligibility. However, relatively little is known about how such variability affects the encoding of speech signals in memory. In the current study, we hypothesize that increased intelligibility leads to better recognition memory for speech. We test this hypothesis by comparing recognition memory for clear and conversational speech produced by native and non-native speakers of English. In experiment 1, listeners were exposed to semantically anomalous sentences produced by a native speaker of English in clear and conversational speech styles. In experiment 2, listeners heard sentences in both styles produced by a Croatian-accented speaker of English. In each experiment, after initial exposure, recognition memory was tested with an old-new identification task. Preliminary results suggest that native-accented speech and clear speech may facilitate recognition memory compared to foreign-accented and conversational speech. The neural mechanisms involved in processing speech of varying intelligibility are currently being investigated using fMRI. The results of this project will expand our understanding of how different sources of variability in the speech signal affect speech processing and memory representations.

3aSCb18. Perceptual accommodation to rate, talker, and context in Mandarin. Joan A. Sereno, Hyunjung Lee, and Allard Jongman (Dept. of Linguist., Univ. of Kansas, Lawrence, KS 66045, sereno@ku.edu)

The present research examines how perceptual constancy is achieved by exploring three fundamental sources of acoustic variability: changes in the rate at which a sentence is spoken, differences due to which speaker produced the sentence, and variations resulting from the sentence context. These sources of variability are investigated by examining the processing of Mandarin tone (specifically Tone 2 and Tone 3). Acoustic analyses (fundamental frequency and duration) of tone productions of 12 native speakers were conducted. These data were then used to construct stimuli to evaluate perceptual adjustments to speaking rate, talker, and context. 14 target stimuli were created varying in temporal, spectral, and both temporal and spectral characteristics. These target stimuli were preceded by six different precursor sentences representing different speaking rates, speakers, and contexts. Participants were asked for their tone judgments. Analyses show interactions between precursor type and target manipulation. Specifically, normalization primarily occurs when precursor and target vary along the same acoustic dimension. However, variation in coarticulatory context violated this pattern. These experiments will be discussed in terms of the use of temporal and spectral cues and its interaction with listener normalization of speaking rate and talker. [Research supported by NSF.]

3aSCb19. Perceptual shifting versus perceptual learning: How listeners accommodate accented speech. Ed King and Meghan Sumner (Dept. of Linguist., Stanford Univ., Bldg. 460, Rm. 127, Stanford, CA 94305, etking@stanford.edu)

Variation is ubiquitous in speech. Differences among individuals ensure that no two speakers pronounce the same sound the same way. Listeners nonetheless understand variable pronunciations as instances of the same category, adapting to familiar variation (e.g., accents they have heard before) or learning novel variants (e.g., unfamiliar accents). Perceptual learning, in which listeners hear features ambiguous between two phonemes and subsequently shift their perceived phoneme boundary, is claimed as a mechanism for accent learning (Eisner and McQueen, 2005). Work examining adaptation to familiar variation has found that listener expectations about speaker dialect influence speech perception (Hay *et al.*, 2010). This study investigates adaptation to a familiar accent, British English, by testing American listeners' identification of "er" on an [e]-[ɜ] continuum before and after hearing a series of British-accented words without the phoneme /ɜ/. The perceptual learning paradigm predicts that training should shift the 50% "er"-identification crossover toward [ɜ], with listeners accepting more [ɜ]-like sounds as "er" after hearing British English. This approach is novel because none of the training words contain [ɜ], suggesting that, unlike traditional perceptual learning, adaptation to a phoneme in a familiar accent does not require immediate exposure to that phoneme.

3aSCb20. Non-symmetric time resolution for spectral feature trajectories. Stephen A Zahorian, Jiang Wu, and Montri Karnjanadecha (Dept. of Elec. and Comput. Eng., Binghamton Univ., Binghamton, NY 13902)

In a study presented at the fall 2010 meeting of the Acoustical Society of America (Zahorian *et al.*, "Time/frequency resolution of acoustic features for automatic speech recognition"), we demonstrated that spectral/temporal evolution features which emphasize temporal aspects of acoustic features, with relatively low spectral resolution, are effective for phonetic recognition in continuous speech. These features are computed using discrete cosine transform coefficients for spectral information from 8 ms frames and discrete cosine series coefficients (DCSCs) for their temporal evolution, over overlapping intervals (blocks) longer than 200 ms. These features are presented as an alternative to mel-frequency cepstral coefficients, and their delta terms, for automatic speech recognition (ASR). In the present work, it is shown that these features are even more effective for ASR, using a non-symmetric time window which is tilted toward the beginning of each block when computing DCSCs. This non-symmetry can be implemented by combining two Gaussian windows with different standard deviations. This work also supports the hypothesis that the left context is somewhat more informative to phonetic identity than is the right context. Experimental results for automatic phone recognition are given for various conditions using the TIMIT and NTIMIT databases.

3aSCb21. Listening to a novel foreign accent, with long lasting effects. Neil P. Bardhan and Andrea Weber (Adaptive Listening Group, P.O. Box 310, 6500AH Nijmegen, The Netherlands, neil.bardhan@mpi.nl)

In conversation, listeners frequently encounter speakers with foreign accents. Previous research on foreign-accented speech has primarily examined the short-term effects of exposure and the relative ease that listeners have with adapting to an accent. The present study examines the stability of this adaptation, with seven full days between testing sessions. On both days, subjects performed a cross-modal priming task in which they heard several minutes of an unfamiliar accent of their native language: a form of Hebrew-accented Dutch in which long /i:/ was shortened to /I/. During this task on Day 1, recognition of accented forms was not facilitated, compared to that of canonical forms. A week later, when tested on new words, facilitatory priming occurred, comparable to that seen for canonically produced items tested in both sessions. These results suggest that accented forms can be learned from brief exposure and the stable effects of this can be seen a week later.

3aSCb22. Hearing vocalizer race from voiced laughter sounds. Anais F. Stenson, Noel B. Martin, R. Toby Amoss, and Michael J. Owren (Dept. of Psych., Georgia State Univ., P.O. Box 5010, Atlanta, GA 30302-5010, anaisfern@gmail.com)

Previous experiments testing whether an individual's race can be identified from speech sounds have produced inconsistent results. Current work examined this question using spontaneous laughter sounds to help isolate voice quality from dialect effects. In Experiment 1, six laughs from each of six black and six white vocalizers were edited to create 72 bouts of three to four contiguous bursts. These bouts were then further edited to create 72 single bursts. Fourteen undergraduate participants heard both sets of laughs in separate blocks. Sounds were presented twice on each trial and categorized as "Black" or "White." Mean percentage-correct was 61.1 (SD = 10.1) with multi-burst versions, and 55.8 (SD = 12.7) with single bursts. Both

outcomes were statistically higher than chance performance in *t*-test comparisons. Experiment 2 tested 12 new listeners in a balanced, same-different design with paired, bout-length stimuli from same- or different-race vocalizers. There were 120 stimuli in all, combining 5 laughs from each of 5 laughers of each race. Mean d' was 0.92 (SD = 0.44), which was statistically different from chance. Taken together, these experiments suggest modest, but reliable sensitivity to vocalizer race from voice-quality alone.

3aSCb23. Compensation for altered feedback is sensitive to speaking style. Shira Katseff (NZ Inst. of Lang., Brain, and Behaviour, Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, shira.katseff@canterbury.ac.nz) and John F. Houde (UCSF, HSE800, San Francisco, CA 94143-0444)

Many speakers oppose alterations to auditory feedback, using a higher pitch when they hear their pitch lowered or a lower first formant when they hear that formant raised. But there is substantial variation in compensation within and across individuals. This case study asks whether a speaker's individual vowel space influences compensation for altered auditory feedback. A generalized linear regression model was used to investigate the influence of vowel density on trial-to-trial formant movement. First, two density maps, showing "hot spot" regions where the subject produced many vowels and "cold spot" regions where the subject produced few vowels, were constructed for (1) casual speech (from a 30-min mock interview) and (2) citation form speech (from 360 HVD words). Maps were produced for one non-phonetician speaker of California English. Afterward, the speaker participated in Five altered feedback experiments on Five separate days. The analysis showed that the subject's between-trial formant changes were not correlated with spontaneous speech vowel density but were positively correlated with citation form vowel density. This pair of results suggests that speech motor decisions are influenced by speech style, accessing citation form vowels more readily when producing citation form speech.

3aSCb24. Perception of speaker sex in children's voices. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX 75083-0688) and Terrance M. Nearey (Univ. of AB, Edmonton, AB T6G 2E7, Canada)

To determine how accurately adult listeners can identify speaker sex in children's voices, we presented vowels in /hVd/ syllables produced in isolation and in a brief carrier sentence by five boys and five girls from each of four age groups (5–8, 9–12, 13–16, and 17–18 yr). Speaker sex identification improved with increasing age of the speaker, but even in the youngest group performance was significantly above chance. Strong biases were noted, with older boys more accurately classified than older girls, while the reverse pattern was found at younger ages. Listeners sometimes reported difficulty distinguishing the older girls from younger pre-adolescent boys. Recognition of speaker sex from isolated syllables was more accurate and listeners were more confident of their responses when informed of the age of the speaker. Performance was also higher when syllables were embedded in a carrier phrase, and in this condition the advantage provided by knowledge of the speaker's age was reduced. The results indicate that the perception of speaker sex can be informed by knowledge of the speaker's age, and that sentence context provides additional cues to speaker sex in children's speech.

3aSCb25. Fundamental frequency and perceptual adaptation to gender and emotion. Daniel J. Hubbard and Peter F. Assmann (School of Behavioral and Brain Sci. GR4.1, Univ. of Texas at Dallas, P.O. Box 830688, Richardson, TX 75083, dhubbard@utdallas.edu)

Previous work has documented perceptual adaptation to nonlinguistic properties in speech using voice gender and emotion categories. Exposure to voice gender and expressive speech adaptors produced a response shift away from the adaptation category. This project extends those findings by examining the contribution of fundamental frequency (F0) to perceptual speech aftereffects following adaptation to VCV syllables. Voice recordings from six speakers were processed using the STRAIGHT vocoder and an auditory morphing technique to synthesize gender (experiment 1) and expressive (experiment 2) speech sound continua ranging from one category endpoint to the other (female to male; angry to happy). Continuum endpoints served as adaptors for F0 present and F0 removed conditions. F0 removed stimuli were created by replacing the periodic excitation source with broadband noise. Aftereffects were found only in the F0 present condition, resulting in a decreased likelihood to identify test stimuli as part of the adaptation category. Aftereffects did not appear when F0 was removed. The findings highlight the important role F0 plays in perceptual adaptation to gender and expressive properties of speech, and further identifies a common acoustic basis for speech adaptation among several different classes of stimuli.

3aSCb26. Investigating the roles of vocal tract size and phoneme content in context effects. A. Davi Vitela and Andrew J. Lotto (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, adv1@email.arizona.edu)

In this series of experiments, we examined the role of specific vocal tract and phoneme contextual information on vowel categorization. An articulatory synthesizer was used to create vowels produced by specific vocal tract lengths (VT) of different sizes [Story, J. Acoust. Soc. Am. **117**, 3231-3254 (2005)]. Contexts of repeated vowels (/a/ or /i/) produced by long and short VTs preceded a target series that changed perceptually from /bit/-/bet/. If listeners are extracting information about the VT itself, similar context effects should be elicited by either vowels produced by the same VT. If listeners are comparing the ambiguous target to the placement of a known vowel, one might predict similar results for the /aa/ condition or the /ii/. However, a complete cross-over interaction was obtained such that the effect of the context could not be predicted by the VT size or the vowel identity, but instead on the acoustic characteristics of F1. Follow-up experiments used contexts consisting of cross-cutting vowels from both VTs or two different vowels from the same VT. The results suggest that the order of the appended stimuli matters, demonstrating that listeners are not averaging information equally across the entire context. [Work supported by NIH-NIDCD.]

3aSCb27. Free classification of dysarthric speech. Kaitlin L. Lansford, Rebecca E. Norton, Julie M. Liss, and Rene L. Utianski (Dept. of Speech and Hearing Sci., Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287, kaitlin.lansford@asu.edu)

The "gold standard" for classification of motor speech disorders, based on the work of Darley *et al.* [J. Speech Lang. Hear. Res. **12**, 246–269 (1969)], presumes that the underlying pathophysiology of each dysarthria subtype is responsible for the associated perceptual features. However, this approach to differential diagnosis suffers significant limitations in that (1) there is substantial overlap of perceptual characteristics associated with the dysarthria classes and (2) characteristics of a given dysarthria vary with severity. It is therefore difficult to perceptually classify speakers reliably without knowledge of the underlying etiology. In the present investigation, 20 speech pathology graduate students grouped 33 speakers with dysarthria into clusters according to the speakers' perceptual similarity using an unconstrained free classification task. The resulting similarity data underwent cluster analysis to identify subsets of perceptually similar speakers. Multidimensional scaling (MDS) revealed the dimensions salient to judgments of perceptual similarity. To interpret the dimensions derived by the MDS, correlation analyses comparing the speakers' spatial distances along the dimensions to obtained segmental and suprasegmental acoustic metrics was conducted. The results of these analyses and how they support the establishment of a non-etiology based, perceptually relevant taxonomy of dysarthric speech will be discussed. [This investigation was supported by 1 F31 DC010093-01A2 awarded to K.L.]

3aSCb28. Temporal and spectral characteristics of fricatives in dysarthria. Heejin Kim, Mark Hasegawa-Johnson (Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, hkim17@illinois.edu), and Adrienne Perlman (Univ. of Illinois at Urbana-Champaign, Champaign, IL 61820)

Imprecise production of fricatives has been noted in various types of dysarthria, but their acoustic properties have not been fully investigated. Using durational measures and spectral moment analysis, this study examines the acoustic characteristics of alveolar and post-alveolar voiceless fricatives produced by speakers with cerebral palsy (CP) in three different intelligibility levels. The following questions are addressed: (1) Are fricatives longer in CP-associated dysarthric speech, compared to controls? (2) What is the evidence of reduced acoustic distinctions between alveolar vs. post-alveolar fricatives? and (3) Are the intelligibility levels associated with acoustic measures of fricatives? Duration and the first three spectral moments were obtained from word initial fricatives produced by 18 American English native speakers (9 speakers diagnosed with spastic CP and 9

controls). Results showed that speakers with CP exhibited significantly longer duration of fricatives and a reduced distinction between alveolar versus post-alveolar fricatives compared to control speakers. A reduced place distinction in dysarthric speech was mostly due to lower first moments and higher third moments compared to normal speech. The group difference was greater for alveolar fricatives than for post-alveolar fricatives. Furthermore, as the intelligibility level decreased, durational increase and the degree of place overlap were consistently greater.

3aSCb29. The role of linguistic and indexical information in improved recognition of dysarthric speech. S. A. Borrie, M. J. McAuliffe (Dept. of Commun. Disord., Univ. of Canterbury, Private Bag 4800, Christchurch, New Zealand, steph.borrie@gmail.com), J. M. Liss (Arizona State Univ., Tempe, AZ 85257-0102), G. A. O.Beirne (Univ. of Canterbury, Christchurch, New Zealand), and T. Anderson (Van der Veer Inst. for Parkinson's and Brain Res., Christchurch, New Zealand)

This investigation examined perceptual learning of dysarthric speech. Forty listeners were randomly assigned to one of two identification training tasks, aimed at highlighting either the linguistic (word identification task) or indexical (speaker identification task) properties of the neurologically degraded signal. Immediately following familiarization, all listeners completed an identical phrase transcription task. Analysis of post-training listener transcripts revealed remarkably similar intelligibility improvements for listeners trained to attend either the linguistic or the indexical properties of the signal. Perceptual learning effects were also evaluated with regard to underlying error patterns indicative of segmental and suprasegmental processing. Comparisons revealed no significant difference at either level of perceptual processing for the two training groups. The findings of this study suggest that elements within both the linguistic and indexical properties of the dysarthric signal are learnable and interact to promote improved processing of this type and severity of speech degradation. Furthermore, error pattern analysis indicates that similar cognitive–perceptual mechanisms may underlie the processing of indexical and linguistic information. Thus, the current study extends support for the development of a model of perceptual processing in which the learning of indexical properties is encoded and retained alongside linguistic properties of the signal.

3aSCb30. Automated rhythmic discrimination of dysarthria types. Rene L. Utianski, Visar Berisha, Julie M. Liss, and Kaitlin Lansford (Dept. of Speech and Hearing Sci. Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287, rutiansk@asu.edu)

This study examines whether features of rhythm, extracted via an automated program, can match the success of hand-extracted metrics in discriminating among control and different types of dysarthric speech. Previously, Liss *et al*. [J. Speech, Language, and Hearing Res., **52**(5), 1334–1352 (2009)] demonstrated that rhythm metrics could successfully separate individuals with perceptually distinct rhythm patterns, with the majority of classification functions more than 80% successful in classifying dysarthria type. However, the hand extracted rhythm measurements are labor intensive and require expertise in acoustic analysis to achieve valid and reliable measures. In this study, digitized speech was automatically segmented into vocalic and voiceless intervals using an autocorrelation-based algorithm, and, from these intervals, rhythm metrics were computed. Discriminant function analyses were performed and revealed the majority of functions were more than 90% successful in classification, matching, or exceeding the success of the hand- extracted measurements. Removing subjectivity, automating this procedure, and verifying its comparability with traditional methods improve the viability of rhythm metrics as a clinical tool in speech therapy intervention. Additionally, the ease of manipulation of the automated program will allow for development of metrics that capture severity and individual speaker differences that will inform models of speech perception.

3aSCb31. The effect of listener age on the recognition of dysarthric speech. Megan J. McAuliffe, Elizabeth M. R. Gibson, Sarah E. Kerr (Dept. of Commun. Disord. and New Zealand Inst. of Lang., Brain & Behaviour, Univ. of Canterbury, Christchurch, New Zealand, megan.mcauliffe@canterbury.ac.nz), and Tim Anderson (Van der Veer Inst. for Parkinson's and Brain Res., Christchurch, New Zealand)

A growing number of studies have attempted to address the cognitive–linguistic source of intelligibility deficits in dysarthria. These have focused predominantly on the ability of young adults with normal hearing to decipher dysarthric speech. However, dysarthria is commonly associated with aging; older listeners often form primary communication partners. It is also known that older adults exhibit difficulty understanding speech that has been temporally or spectrally degraded. It follows that the spectral and temporal degradations present in dysarthric speech may pose a greater perceptual challenge for older, as opposed to younger, listeners. Twenty younger listeners (mean age = 20 yr) and 15 older listeners with good hearing for their age (mean age = 65 yr) transcribed the speech of individuals with moderate hypokinetic dysarthria. Percent words correct (intelligibility) was calculated and underlying error patterns at the suprasegmental and segmental levels of processing were examined. While the younger and older listener groups achieved similar intelligibility scores, the younger group showed greater reliance on syllabic strength cues to inform word boundary decisions. Similar levels of attention to segmental cues were observed across both groups. These results suggest that the recognition of dysarthric speech may be comparable across younger and older listeners.

3aSCb32. Movements of the tongue, lips, and jaw for selected vocalic nuclei in speakers with dysarthria. Gary Weismer (Dept. Commun. Disord., UW-Madison, Goodnight Hall, Madison, WI 53706)

Speech movement data in persons with dysarthria are relatively rare, and much of this sparse data is restricted to the lips and jaw. The purpose of this presentation is to provide a survey of speech movements for selected vocalic nuclei, in speakers with dysarthria due to Parkinson's disease and Amyotrophic Lateral Sclerosis, and to compare these movements to those of healthy controls. The speech materials used in this analysis are those collected over a ten-year period at the x-ray microbeam facility at the University of Wisconsin-Madison. The selected vocalic nuclei are extracted from connected speech samples, including both read sentences and an extended reading passage; multiple repetitions of sentence data are available to estimate the variability of the selected movements. Both qualitative and quantitative analyses of these speech movements, with emphasis on lingual motions, will be employed to generate hypotheses concerning the speech movement deficit in the dysarthrias. This inductive analysis approach will pay special attention to across-speaker variability, among both healthy controls and speakers with dysarthria. A working hypothesis is that the lingual motions for speakers with dysarthria are no more variable across repetitions than those in healthy controls, but show deficits in scale and form.

3aSCb33. A continuing study of the temporal structure of the speech of a person with dementia. Linda Carozza (Dept. of Commun. Sci. and Disord., St. John's Univ., 300 Howard Ave., Staten Island, NY 10301), Margaret Quinn, Julia Nack, and Fredericka Bell-Berti (St. John's Univ., Queens, NY 11439)

The question of motor speech degeneration in the course of dementing illness is a relatively unexplored area; rather, extensive research has focused on cognitive and language processes in dementia. The potential for early dissociation of motor functions of language at the level of speech production has not been explored; an interaction between motor speech and language production and perception changes should inform our understanding of the deterioration in dementia. We have previously reported results of a preliminary study of the temporal structure in the speech of persons with dementia. In a continuing longitudinal study of one of our subjects, we found that after 6 months, utterance-level timing patterns were more disrupted than segment- and syllable-level patterns in his speech. This report explores changes over time in the temporal structure of speech of that participant, who was recorded 12 months after the second recording session. The results of this third recording session, in which we will examine utterance-level timing patterns (phrase-final lengthening and compensatory shortening) and segment- and syllable-level timing patterns (VOT and the effects of final consonant voicing on vowel duration) will be presented.

3aSCb34. Prosodic features and evaluation of pitch controllable electrolarynx. Minako Koike, Satoshi Horiguchi (School of Medicine, Kitasato Univ., 1-15-1 Kitasato, Minami-ku, Sagamihara, Kanagawa, 252-0374, Japan, minako@kitasato-u.ac.jp), Hideki Kasuya (Utsunomiya Univ., Utsunomiya 321-8585, Japan), Yoshinobu Kikuchi (Int. Univ. of Health and Welfare, Otawara 324-8501, Japan), Makihiko Suzuki, and Makito Okamoto (Kitasato Univ., Sagamihara 252-0374, Japan)

The prosodic features and perceptual evaluation of an electrolarynx (EL) with pitch-control function were examined. We previously constructed a prototype electrolarynx of which F0 (hereafter "pitch") can be adjusted by up-down or left-right thumb movement, and it was recently developed into a commercial product (Yourtone II, manufactured by DENCOM). Users can choose the pitch-fixed type with jitter (PF-EL) or the pitch-controllable type without jitter (PC-EL). One laryngectomized male speaker, who was an EL user and whose mother tongue was Japanese, served as the subject. He practiced controlling the pitch of PC-EL in Japanese (Tokyo dialect) for two weeks, and then his utterances when using PF-EL and PC-EL were recorded. Twenty normal listeners evaluated his EL speech tokens and were asked to rate on a visual analog scale how close the speech tokens were to normal speech. The results indicated that although the pitch range of PC-EL was narrower than that of normal speech and the production of Tokyo-dialect Japanese was not perfect, most of the PC-EL speech was rated more highly than PF-EL speech. [Work supported by KAKENHI (20500163 and 22500147).]

3aSCb35. A principal components analysis of tongue motion patterns in normal and glossectomy speakers. Julie Langguth (Dept. of Orthodontics, Univ. of Maryland Dental School, 650 W. Baltimore St., Baltimore, MD 21201, juliemikhail@gmail.com), Jonghye Woo, Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD 21201), Hegang Chen (Univ. of MD School of Medicine, Baltimore, MD 21201), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD 21218)

This study determines whether a small excision of the tongue, due to removal of a lateral T1 tumor, will cause a change in tongue motion patterns during speech. Principal components analysis (PCA) compared the motion patterns of ten control subjects to those of three glossectomy patients. The patients were 7–39 months post-glossectomy removal of T1N0M0 tumors and had primary closures. Cine- and tagged- magnetic resonance imaging (MRI) data included three sagittal slices, which were used to compare the midsagittal and bi-lateral motions of the tongue during speech. The data were internal tissue point motions extracted from the tagged MRI images using harmonic phase MRI (HARP-MRI) (NessAiver and Prince, 2003). The motion patterns of the tongue (velocity field) were observed during the motion from "ee" to "s" which may be difficult for glossectomy patients as it requires subtle changes in the tongue tip. The results demonstrated normal variability across the control subjects. The three patients retained some components of normal tongue motion; however, there were also notable differences from the controls. The glossectomy patients plotted at the extremes of the PC loadings, and the details revealed how they differed from the controls, from each other, and used individual compensatory strategies. [Work supported by NIH-R01CA133015.]

3aSCb36. Acoustic characteristics of diphthongs in Parkinson's disease. Kris Tjaden (Dept. of Commun. Disord. & Sci., Univ. at Buffalo, 3435 Main St., Buffalo, NY 14214, tjaden@buffalo.edu)

The second formant frequency (F2) is sensitive to dysarthria. For example, vowel F2 slope has been shown to correlate with intelligibility, such that persons who are less intelligible or who have more severe speech impairment also tend to have shallower F2 slopes (Weismer *et al.*, 1992). This result is consistent with studies suggesting the perceptual importance of transitions (Hillenbrand and Nearey, 1999), although the extent to which F2 slope explains reduced intelligibility in dysarthria is unclear. F2 slope also may serve as an acoustic metric of effort (Moon and Lindblom, 1994; Wouters and Macon, 2002). Therapeutic techniques for dysarthria, including Clear and Loud speech, are thought to be beneficial because they involve a scaling up of effort. Clear speech emphasizes hyperarticulation or increased articulatory effort, while Loud speech focuses on increasing respiratory–phonatory effort, although it has been suggested that increased loudness involves a system-wide scaling up of effort (Fox *et al.*, 2006). Empirical evidence that Clear or Loud speech is associated with increased articulatory effort in dysarthria is limited. The current study sought to further evaluate this suggestion by investigating F2 slope characteristics for vowels produced in Habitual, Clear and Loud conditions by individuals with dysarthria secondary to Parkinson's disease.

3aSCb37. Music masking speech in cochlear implant simulations. Shaikat Hossain and Peter F. Assmann (Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX 75083)

While the question of how speech is segregated among competing talkers has received quite a lot of attention in the cochlear implant (CI) literature, an equally important question has been largely neglected: To what extent does music mask speech? Recent studies have shown that bimodal stimulation, using a hearing aid to boost low frequency acoustic information in the ear with residual hearing, can markedly improve CI users' ability to separate multiple talkers, recognize speech in noisy conditions and discriminate between melodies. In this study, normal hearing listeners completed a speech recognition task in musical maskers under two different processing conditions: bimodal and normal CI simulations. Results showed that listeners performed significantly better under the bimodal simulation in all three SNR conditions (0, -3, -6 dB). It is thought that the low frequency acoustic information provided listeners with a release from masking as compared to the normal CI condition, particularly at low SNRs. The amount of masking release varied as a function of instrument type and SNR. The findings of this study may be especially relevant for the perceptual separation of instrumentation and lyrics in popular music which remains to be a highly challenging task for CI users.

3aSCb38. Envelope and temporal fine structure perceptual weights for sentences: Effect of age, hearing loss, and amplification. Daniel Fogerty and Larry E. Humes (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, dfogerty@indiana.edu)

The speech signal may be divided into spectral frequency bands, each band containing temporal properties of the envelope and fine structure. This study measured the perceptual weighting strategies for the envelope and fine structure in each of three frequency bands for sentence materials in young normal-hearing listeners, older normal-hearing listeners, aided older hearing-impaired listeners, and noise-matched young normal-hearing listeners. A novel processing method was designed to vary the availability of each acoustic property independently through noisy signal extraction. Thus, the full speech stimulus was presented with noise used to mask the different auditory channels. Perceptual weights were determined by correlating a listener's performance with the signal-to-noise ratio of each acoustic property on a trial-by-trial basis. Preliminary results demonstrate that fine structure perceptual weights remain stable across the four listener groups. However, a different weighting typography was observed between the listener groups for envelope cues. Preliminary results suggest that spectral shaping used to preserve the audibility of the speech stimulus may alter the allocation of perceptual resources. The relative perceptual weighting of envelope cues may also change with age. These effects were largely limited to envelope weights. [Work supported by NIA R01 AG008293.]

3aSCb39. Contributions of static and dynamic spectral cues to vowel identification by cochlear implant users. Gail S. Donaldson, Catherine L. Rogers, Emily S. Cardenas, Benjamin A. Russell, and Nada H. Hanna (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, gdonalds@usf.edu)

Relatively little is known about cochlear implant (CI) users' ability to make use of static versus dynamic spectral cues in vowel perception tasks. The present study measured vowel identification in CI users and young normal hearing (YNH) listeners using naturally produced /dVd/ stimuli (*deed, did, Dade, dead, dad, dud,* and *Dodd*). Vowel identifcation was tested for (1) the unmodified syllables, (2) syllables modified to retain only 60 or 80 ms of the vowel center (*center-only* conditions), and (3) syllables modified to retain only 30 or 40 ms of the initial and final vowel transitions, with vowel duration neutralized (*edges-only* conditions). YNH listeners achieved high levels of performance for the unmodified stimuli (avg. 99.8%) and for the center-only stimuli (90.8%); their performance dropped to more moderate levels (68.1%) for the edges-only stimuli. CI users demonstrated moderate performance for the unmodified stimuli (avg. 72.0%) but demonstrated substantially poorer performance for both the center-only (41.1%) and edges-only stimuli (27.8%). Findings suggest that CI users (1) have difficulty identifying vowels in syllables when one or more cues are absent and (2) rely more strongly on quasi-static cues from the vowel center as compared to dynamic cues at the syllable edges.

3aSCb40. Spectral contrast analyses of vowels processed through a multichannel simulated hearing aid. Amyn M. Amlani (Dept. of Speech and Hear Sci, Univ. of North Texas, P.O. Box 305010, Denton, TX 76201, amlaniam@unt.edu), Sneha V. Bharadwaj (Texas Woman's Univ., Denton, TX 76204), Shirin Jivani, and Jody Pogue (Univ. of North Texas, Denton, TX 76201)

Hearing aids are designed to separate the audible frequency response into independent channels so that gain can be modulated correspondingly, based on the degree and configuration of the hearing loss. However, increasing the number of channels and employing fast-acting compression might lead to the spectral flattening of speech segments. In this study, we performed spectral contrast analyses on three vowels /i, oo, u/ spoken by two male and two female talkers in a /CVC/ context. The initial consonant consisted of /p, b, t, d, k, g, s, sh/ and the final consonant consisted of /t/. Each /CVC/ word was spoken in the phrase, "I said /CVC/, again" Each phrase was then processed by a simulated hearing aid having 2, 4, 8, and 16 independent channels according to an amplification scheme (linear, compression [fast-fast, slow-slow, fast-slow]). Findings revealed that spectral contrast decreased (1) as the number of channels increased, (2) when a fast attack time was employed, and (3) when the talker was a male. Further, significant spectral flattening and shifts in vowel formants occurred for /i, oo/ compared to /u/. We discuss these findings relative to the speech-intelligibility performance of impaired listeners.

3aSCb41. An additive blending sound presentation system with air and bone conducted sound. Kenzo Itoh (Dept. of Comput. Sci. and Information System, Iwate Prefectural Univ., Iwate 020-0193, Japan, itoh@iwate-pu.ac.jp)

The sound of our voice reaches the inner ear by two paths: the first being air and the second being sound conducted by bones. When you speak, what you hear as your voice is the blending of two signals: one signal comes from the auditory canal and the other signal comes from bone conduction. This paper proposes a new sound presentation system with an additive blending effect with air and bone conducted signals. The system is composed of normal headphones and/plus a bone conduction transducer. The sound quality is evaluated by subjective tests using three kinds of musical sources: Piano, Vocal, and Orchestra. Two experimental conditions are used. One condition is normal headphones as air only (AIR), and the other is normal headphones and a bone conduction transducer simultaneously and in phase (BONE). The subjects select the better quality after comparing AIR and BONE conditions. The result of the experiment shows that the most subjects chose BONE. The proposed system can be used for aged persons or persons with hearing loss.

3aSCb42. Perceptual deficits in Thai sensorineural hearing loss patients. C. Tantibundhit, C. Onsuwan, N. Rueangwit, N. Saimai, T. Saimai (Dept. of Elec. and Comput. Eng., Thammasat Univ., Thailand, tchartur@engr.tu.ac.th), N. Thatphithakkul, K. Kosawat, S. Thatphithakkul, and P. Chootrakool (Natl. Electron. and Comput. Technol. Ctr., Thailand)

This study explored differences in CVV perception in two groups of Thai listeners: with normal hearing and with sensorineural hearing loss (with/without hearing aids). All participants chose one response in each of 210 Thai stimulus rhyming pairs, e.g., /taa/-/naa/. The rhyming monosyllabic words share an /aa/ vowel and mid tone, but differ in their initial phonemes (symmetrically distributed across 21 phonemes). While all stimuli for the normal hearing group were embedded in 4 signal-to-noise ratio levels, clean stimuli were presented to the patients. Comparisons of confusion patterns and perceptual distance were made. In both groups, /r/ is the most confusable phoneme, while /w/ is among the least. Perceptual representations of initial phonemes show five individual clusters: glide, glottal constriction, nasality, aspirated obstruent, and a combination of liquid and unaspirated obstruent. Patients' perceptual difficulty could be attributed to the nasality grouping, which is normally well separated, shifting closer to the glottal constrictions and aspirated obstruents. Hearing aids seem to improve perception of all phonemes by 10%, with /kh/ and /h/ showing the highest improvement rate, and /d/ the lowest. The instruments are beneficial in moving the nasality cluster further away from the nearby groupings.

3aSCb43. Living sound identification system using smartphone for persons with hearing loss. Ashita Sarudate and Kenzo Itoh (Dept. of Comp. Sci. and Info. Sys., Iwate Prefectural Univ., Iwate 020-0193, Japan, g236g003@s.iwate-pu.ac.jp)

In today's advanced information society that overflows with a variety of sounds, people with hearing loss find it very difficult to obtain information on sounds within the home. Although there are many systems to aid handicapped persons, support systems for the hearing impaired do not provide adequate performance. In light of this, the various living sounds identification system for persons with hearing loss was proposed. This system adopts the method of pre-storing signal characteristics so that it can discriminate important living sounds in the home with a high degree of accuracy. In order to construct a real-time processing system, basic signal processing was subjected to frame-by-frame analysis. In addition, so as to be able to detect signals precisely in a noisy environment, signal part detection—based on auditory perception—was adopted. In an experiment in a simulated life space, the system was able to discriminate ten living sounds with almost 100% accuracy. Therefore, the proposed method was very applicable. In addition, the method used by us to present the sound identification results using a mobile phone is considered to be of value. This paper proposed the living sound identification system for use with a smartphone and discussed the utility of this system.

3aSCb44. Talker and gender effects in induced, simulated, and perceived stress in speech. James Harnsberger, Christian Betzen, Kristen Perry, and Harry Hollien (Dept. Linguist., Univ. of Florida, Gainesville, FL 32611, jharns@ufl.edu)

Prior modeling on the speech correlates of psychological stress have typically reported averaged data in which the problem of quantifying degree of stress was not adequately addressed. Harnsberger, Kahan, and Hollien (2006) reported a speech database in which stress was quantified both with (two) physiological measures and (two) self-report scales. In this study, these materials were assessed in two stress perception experiments, involving ratings and classification, and the subsequent analysis was organized by gender and individual talker. Several observations were made. First, both induced and simulated stress samples were perceptibly different from baseline, unstressed samples. Second, up to one third of talkers in a given gender group differed from the overall trend, showing instead a propensity to be rated as less stressed during high degrees of physiological arousal (reflected also in self-report scales); essentially, sounding calmer under stress than the baseline condition. Third, female talkers showed the expected positive correlation between the degree of physiological shift under stress versus the shift in self-reported stress ratings while, in male subjects, higher degrees of physiological arousal were underreported in self-report (a "tough guy" response). Finally, acoustic modeling efforts were conducted by gender and stress response type, for induced, simulated, and perceived stress.

3aSCb45. Reliability and validity of a simple user interface (EarPrint^{TM}) for the personalization of mobile audio. Meena Ramani, Nick Michael, and Chaslav Pavlovic (Sound ID and SoundHawk 2595 East Bayshore Rd., Palo Alto, CA 94303)

People hear differently and have different aesthetic criteria. Due to a number of these reasons, individuals may prefer different processing parameters for both speech and music. For example, some individuals may prefer more low frequency emphasis and longer reverberation time while listening to music. Alternatively, certain individuals with hearing loss may prefer higher compression ratios at high frequencies. To perform this "personalization," a number of psychometric procedures can be used. They vary in terms of accuracy, complexity, testing time, etc. For consumer devices, such as headsets, personal amplifiers, personal music players, etc., we have developed a new methodology, called EarPrint^{TM}, which is fast, intuitive, and requires minimum instructions. The procedure is based on reducing a multidimensional parametric space into a 2-D surface representation where the user perceives a consistent quality increment as he/she moves his/her finger along a straight line. In this paper, we evaluate how valid and reliable the EarPrint^{TM} results are when compared to more rigorous psychometric procedures (e.g., category scaling or magnitude estimates.) Fifteen subjects were tested and the results were analyzed in terms of validity, dependence on listening material and reliability.

Session 3aSPa

Signal Processing in Acoustics: Array Processing and Source Localization Techniques

Caglar Yardim, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Edmund J. Sullivan, Cochair
EJS Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871-1032

Contributed Papers

8:00

3aSPa1. Broadband model-based tracking. Edmund Sullivan (Prometheus Inc., 46 Lawton Brook Ln., Portsmouth, RI 02871)

Joint range and bearing estimation using a Kalman-based recursive processor has already been achieved for the narrow band case. The signal is modeled as having a circular wavefront curvature, and the range is estimated as the radius of this curvature. The bearing is also jointly estimated along with the range. Since there is a direct estimation of the range, a maneuver of the towed array receiver is not necessary. To extend this procedure to the broadband case, a preprocessed measurement is computed using a sequence of contiguous Fourier transforms. The phases of the associated frequency components are then aligned, providing one effective frequency. This allows an effective time delay to be computed for each Fourier transform, thus allowing a sequential estimation of the bearing and range to be carried out. A further advantage of this approach is that, since the array is in motion, the Doppler of the effective frequency can be included in the signal model, thereby allowing the additional bearing information in the Doppler to be exploited. This extra bearing information significantly decreases the variances on the estimated parameters, providing a synthetic aperture effect. This permits the use of a relatively short array aperture.

8:15

3aSPa2. Tracking system developed using an optical fiber-based distributed acoustic sensor. Georgios Efstathopoulos, Daniel Finfer, Yousif Kamil, Sergey Shatalin, Tom Parker, and Mahmoud Farhadiroushan (Silixa Ltd., Silixa House, 230 Centennial Park, Elstree, Hertfordshire WD6 3SN, United Kingdom)

This work will present a system for acoustic source tracking and imaging developed using a novel distributed acoustic sensor. This sensor, the intelligent Distributed Acoustic Sensor (iDAS), uses standard off-the-shelf optical fiber as a transducer to capture the audio-frequency acoustic field (both phase and amplitude) simultaneously along the entire length of a fiber with a spatial resolution of about 0.5 m. iDAS was used to track the bearing of acoustic sources in a reverberant environment, with localization being performed in two stages. First, the received signal was preprocessed for the mitigating effects of calibration errors and multipaths. Then tracking was performed via particle filters. These filters can track multiple acoustical sources and are efficient enough to run in nearly real time. The output from this system is presented in the form of an acoustic camera-type video.

8:30

3aSPa3. Tracking of small vessels with passive acoustic sensors. Helen H. Ou, Pasang Sherpa, and Lisa M. Zurk (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, ohui@pdx.edu)

This paper introduces a small boat vessel tracking method in marine environment using autonomous, low-complexity acoustic sensors. A cross-correlation of time series between sensors generates a coarse localization, and an application of the extended Kalman filter (EKF) gives the vessel track. Since each sensor has a local clock that operates asynchronously, the time series received by multiple sensors are first synchronized with one another by measuring a sequence of impulses played at the beginning of the recording. The algorithm detects a moving boat by the increase in the broadband sound level and confirms it by extracting the amplitudes at harmonic frequencies due to the propeller movement. To calculate the time delay of arrival, the boat-present signals are divided into small segments that are cross-correlated with signals received by other sensors. Thresholding and clustering are introduced to extract multiple tracks from the cross-correlation. The EKF is trained using the estimated time delays to provide the vessel track. The algorithm has been successful with Bellhop simulated boat signals and field data collected on the Willamette River and Columbia River in Oregon. Limiting factors such as network topology and sensor movement during deployment are also investigated. [Work sponsored by the Nature Conservancy.]

8:45

3aSPa4. Double multiple signal characterization active large band (D-MUSICAL) for extraction of observable in acoustic tomography. Le Touzé Grégoire, Nicolas Barbara (Gipsa-Lab, 961 rue de la Houille Blanche, BP 46 F - 38402 Grenoble Cedex, gregoire.le-touze@gipsa-lab.grenoble-inp.fr), Roux Philippe (ISTerre, BP 53 38041 Grenoble Cedex 9, France), and Mars Jérôme (Gipsa-Lab., 961 rue de la Houille Blanche, BP 46 F - 38402 Grenoble, Cedex)

In the ocean, the acoustic propagation is characterized by different arrivals. Each arrival is characterized by its observables: amplitude, arrival time, emission, and reception angles. To achieve ocean acoustic tomography, a method to separate arrivals and extract observables has been developed. Two vertical arrays are used: a source and a receive array. Until now, the double-beamforming algorithm has been used to extract observables. It allows to go from the initial (time–source depth–receiver depth) domain to the beamformed domain (time–source angle–receive angle). This algorithm is efficient thanks to the 3-D domain of estimation but shows resolution limitation in the angle domain. To improve wave separation, we develop and apply a high-resolution algorithm based on MUSIC principle, in the same 3-D domain. The proposed method, called Double-MUSICAL, is the extension on the emission angle of the 2-D MUSIC Acitve Large band method (MUSICAL). Its principle is to separate the recording space onto the two orthogonal signal and noise spaces. We present results on real data recorded in an ultrasonic tank and show that the developed method better detects and separates the first arrivals than double beamforming.

9:00

3aSPa5. A direction-selective filter with a rational beam pattern. Dean J. Schmidlin (El Roi Analytical Services, 2629 US 70 East, Unit E-2, Valdese, NC 28690-9005, djschmidlin@charter.net)

Presented in this paper is a direction-selective filter whose beam pattern is a rational function of a direction cosine. First, a plane-wave sinusoidal

pressure function is converted from a 4-D function to a 2-D one by restricting the spatial points to lie on a radial line extending out from the origin in a prescribed look direction. The 2-D pressure function is then input into a linear filter characterized by a second-order partial differential equation with constant coefficients. The natural and forced responses are determined from which expressions for the beam pattern of the filter and the time constant of the natural response are found. The beam pattern is the reciprocal of a second-degree polynomial function of the direction cosine of the plane wave. It is shown that the frequency range over which the integrity of the beam pattern is maintained is a function of the natural response of the filter. An example is presented which illustrates the directivity index that is achievable in contrast to that of a vector or dyadic sensor, both of which have beam patterns that are polynomial functions of a direction cosine.

9:15

3aSPa6. Multiple sources localization with microphone array: A subarray approach. Kai-Chung Tam (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong), Siu-Kit Lau (Construction Univ. of Nebraska-Lincoln), and Shiu-Keung Tang (The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong)

The problems of location and distance estimations with planar microphone array for multiple point sources are addressed in the present study. Though traditional array processing methods show the reliability in estimation of direction-of arrival (DOA) of the sources, such techniques are incapable to provide source distances which requires for three dimensional (3-D) target coordinates. With planar array processing, azimuth and elevation angles are always obtainable but no longer to extend into more details in 3-D analysis. A new approach to deal with this problem is proposed by estimating multiple sources' DOA with multiple planar sub-arrays. A least square solution based on the estimated DOA of the sources is derived to give the source distances with minimum two sub-arrays in arbitrary microphone configurations. The benefits of the proposed method include a extension of the capability of planar array processing to a higher dimensional analysis. Moreover, its adaptability with any DOA estimation methods makes it convenient to use. The new approach is examined with conventional delay and sum beamforming and MUSIC methods under noisy conditions by both simulations and experiments. [K.C.T. is supported by a studentship from the Hong Kong Polytechnic University.]

9:30

3aSPa7. A comparative analysis of numerical optimization algorithm for array pattern synthesis. Yingmin Wang, Xiao Wei, and Yucai Wang (College of Marine Eng., Northwestern Polytechnical Univ., 127 You Yi Xi Lu, Xi'an 710072, People's Republic of China)

The problem of array pattern synthesis with desired shapes has received much attention over years. There is a number of intelligent optimization methods that were adopted for pattern synthesis, and to some content, several successful design cases are reported. In this paper, based on a special sensor array, a comparative study of those methods is given. Those methods include simulated annealing, genetic technique, and adaptive searching procedures. The effectiveness of the algorithms is demonstrated by the sidelobe level and the width of the main beam. For a given array, those techniques are investigated on their convergence speed and residue error energy. A set of design results are given in details and conclusions are included.

9:45

3aSPa8. Analysis of truncation and sampling errors in the spherical harmonic representation of soundfields. Stefanie Brown and Deep Sen (Acoust. Lab., School of Elec. Eng. and Telecommun., Univ. of New South Wales, Sydney, Australia, stefanie@unsw.edu.au)

The use of microphone arrays to record soundfields for subsequent immersive reproduction, real time or off-line beamforming applications are becoming increasingly prevalent. The representation and analysis of such recorded soundfields using finite-order spherical harmonic coefficients is elegant and has a number of advantages. The focus of this paper is to quantify the accuracy of such a representation which is limited by errors introduced by several types of non-idealities: manufacturing non-idealities of the array such as microphone positional errors, coupling between microphones, etc.; physical non-idealities due to the discrete spatial sampling of the soundfield which introduces spatial aliasing and the inherent truncation of the modal representation of the soundfield to a finite order; and mathematical errors due to the computational technique used to derive the modal coefficients. This paper investigates the latter two classes of errors and quantifies the significant advantages in accuracy achievable when the physical structure of the microphone array allows the use of the orthogonal properties of the spherical harmonic basis functions to extract the modal coefficients. In addition, we extend previously published analysis of the truncation error, which was limited to incident plane waves, to spherical waves.

10:00

3aSPa9. Analysis on novel tangible acoustic interfaces approaches and new thoughts. Xu Wang (School of Honors, HIT, Harbin, 150001 China, yiwang20071@hotmail.com) and Hong Jun Yang (Univ. of INHA, Incheon)

To solve the problems in novel tangible acoustic interfaces approaches [time delay of arrival (TDOA) and location pattern match (LPM)], a new method (which is amplify contrast method) of setting threshold so as to determine the arrival point will be applied, and three new thoughts on LPM, which are improved LPM, time match, and amplitude attenuation match, would be proposed. Traditional TDOA is based on measuring time difference and wave velocity. Traditional LPM is based on time reversal theory and cross correlation. New thoughts are based on amplification on signal, classification of signal source, match time, and uniqueness of damping. Amplify contrast method improves the accuracy of determining the arrival point.

Session 3aSPb

Signal Processing in Acoustics, Underwater Acoustics, and Biomedical Acoustics: Fusion of Acoustic Signals with Data from Various Sensor Modalities

Grace A. Clark, Cochair
Electronics Engineering, Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550

Colin W. Jemmott, Cochair
Applied Operations Research, 420 Stevens Ave., Ste. 230, Solana Beach, CA 92075

Chair's Introduction—10:30

Invited Papers

10:35

3aSPb1. Multiview, multimodal fusion of acoustic and optical data for classifying marine animals. Paul L. D. Roberts, Jules S. Jaffe (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla CA 92093-0238), and Mohan M. Trivedi (Univ. of California at San Diego, La Jolla, CA 92093-0434)

Multiview, multimodal acoustic and optical sensor data can be used to improve our ability to remotely classify marine fish and zooplankton. A key advantage of these data is that by capturing more views of a subject, they are more robust to unknown orientation or pose. In this regard, these data offer significant advantages over singleview, unimodal sources. However, they require additional processing to combine data sources together to render a final classification. Here, a fusion algorithm based on combining data sources at the classification probability level (decision fusion) using confidence-weighting with feedback is investigated. This algorithm uses support vector machines as the underlying classifiers can incorporate any number of views or modalities on the fly and is in general independent of the underlying data source. In addition, the algorithm is able to favor views or modalities that are more discriminant without any *a priori* knowledge. The algorithm is tested on an array of problems related to classifying fish from broadband acoustic scattering and classifying zooplankton from broadband acoustic scattering and optical images. In all cases, the algorithm yields significant (more than 50%) reductions in error by fusing multiview, multimodal data.

10:55

3aSPb2. Exploiting multiple sensor modalities for classification of underwater munitions: A Dempster–Shafer theoretic approach. Thanuka L. Wickramarathne, Kamal Premaratne, Shahriar Negahdaripour (Dept. of Elec. and Comput. Eng., Univ. of Miami, Coral Gables, FL 33146), Lisa N. Brisson, and Pierre P. Beaujean (Florida Atlantic Univ., Dania Beach, FL 33004)

Detection and classification of underwater UXOs (UneXploded Ordnance) is a task that is receiving considerable attention from the DoD and related agencies that are involved in management of military munitions. Achieving a reasonable accuracy in detection and classification of these objects is extremely difficult mainly due to the harsh cluttered underwater environment. It is now an accepted fact that no single sensing technology can be both accurate and cost-effective. Low visibility in underwater exposes a significant limitation to optical cameras which are usually better suited for identifying the physical shape of objects. While acoustic imaging is a good alternative, the characteristics of imaging and physical system artifacts make the object recognition task non-trivial. Multi-sensory fusion provides an avenue to exploit the strengths of individual sensors and modalities while mitigating their weaknesses. We address the problem of fusion of multiple sensory information for the task of underwater UXO detection via the use of a Dempster–Shafer (DS) theoretic evidence fusion scheme. The DS theoretic foundation of our fusion algorithm also enables it to incorporate domain expert opinions and knowledge from databases to assist in the detection and classification tasks. We illustrate this method via the use of real data obtained at a test site located in the Florida Atlantic University premises.

Contributed Papers

11:15

3aSPb3. Applications of rhythm algorithm for periodic broadband signals. Alexander Ekimov (NCPA, Univ. of Mississippi, 1 Coliseum Dr., Oxford, MS 38677)

Rhythm algorithm, recently published in J. Acoust. Soc. Am. **129**(3), 2011 was developed for any periodic broadband signals. This algorithm was applied in human and animal footsteps studies for signals from a node of orthogonal sensors and for underwater signals of sperm mammals clicks measured with hydrophones. Some new applications of this rhythm algorithm are presented and discussed. The rhythm algorithm was applied for water breaking waves on a lake shore. The signals were from a node of two orthogonal sensors placed on the lake shore. This node had a narrow band ultrasonic microphone and a Doppler sonar unit. The narrow band ultrasonic microphone showed detectable levels of ultrasonic signals for breaking waves while Doppler sonar showed maximum wave velocities. Signals from this node were recorded and processed simultaneously with the rhythm algorithm. Another application of this rhythm algorithm for music periodic sound is presented also. In this case, it was possible to see the music rhythms and variations of these rhythms with time using short term Fourier transform.

11:30

3aSPb4. Acoustically sensing the presence of methane and carbon dioxide in underground coal mine. A. Singh and M. Radhakrishna (Indian Inst. of Info. Tech., CSED, Allahabad, 211012 India, ajitsingh@iiita.ac.in)

A WSN can be used to continuously sense, monitor, and transmit data to a centralized control station in a under ground coal mine. A fact limiting the possibility is the presence of highly humid condition in UG coal mines. Current sensors cannot work continuously over prolonged period in UG coal mine environment. This paper describes a multi-aspect data fusion approach for acoustical sensors, which make it possible to measure the build up of methane and carbon dioxide in UG coal mine environment. Suggested approach takes time of flight, phase, and attenuation of sonic pulses to determine the build up of methane and carbon dioxide. Suggested approach is more power efficient in comparison to existing sensors. A temperature sensor is used to accommodate change in characteristics of sonic pulses.

11:45

3aSPb5. Development of three-dimensional time reversal algorithm to locate tinnitus-related hyperactive neurons. Na Zhu, Sean Wu, and Jinsheng Zhang (Dept. of Otolaryngol., School of Medicine, Wayne State Univ., Detroit, MI 48201)

This paper aims at developing 3-D time reversal (TR) algorithm to locate the positions of tinnitus-related neural activities inside the brain auditory pathways, including the auditory cortex (AC), inferior colliculus (IC), and dorsal cochlear nucleus (DCN), which have been implicated in the etiology of tinnitus. Spontaneous neural network signals will be measured by using multichannel electrode arrays that are implanted on the AC, IC, and DCN of adult Sprague Dawley rats and be taken as input to this TR algorithm. To specify the speeds at which neural signals are traveling through the brain auditory pathways, benchmark tests will be conducted, in which electrical stimulations at pre-determined positions in the AC, IC, and DCN will be used, and the corresponding responses be measured. Once this is done, the precise locations of hyperactivity that is responsible for tinnitus perception can be determined by using TR algorithm to scan through the brain auditory pathways. Note that this 3-D time reversal algorithm will allow for tracing neurons into deeper layers of the AC, IC, and DCN, provided that the electrode arrays are implanted in strategically placed in space. Numerical simulations and preliminary experimental results will be presented.

WEDNESDAY MORNING, 2 NOVEMBER 2011 PACIFIC SALON 3, 7:40 A.M. TO 12:00 NOON

Session 3aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Inversion Methods and Uncertainty Estimation in Ocean Environments

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David P. Knobles, Cochair
Applied Research Lab., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78713

Chair's Introduction—7:40

Invited Papers

7:45

3aUW1. Sequential geoacoustic inversion using frequency coherent processing. Caglar Yardim, Peter Gerstoft, and William S. Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093-0238, gerstoft@ucsd.edu)

Insufficient sampling in frequency domain results in range aliasing that affect geoacoustic inversion. Range aliasing and its effects on source localization and environmental parameter inversion are demonstrated on data collected during the MAPEX2000 experiment. Using a source and geoacoustic particle filter mitigates range aliasing caused by using just a few frequencies. Particle filters are sequential Bayesian techniques that provide dynamic estimation of both the geoacoustic parameters and their evolving uncertainties. This approach allows for side lobe suppression due to prior information supplied by the particle filter and hence improved source localization that results in better geoacoustic inversion.

8:05

3aUW2. Geoacoustic inversion with sequential Bayesian filtering and multipath arrivals. Zoi-Heleni Michalopoulou (Dept. of Math. Sci., New Jersey Inst. of Technol., Newark, NJ 07102, michalop@njit.edu)

When sound propagates in the ocean, multipath arrival receptions at spatially separated hydrophones provide a wealth of information on geometry parameters, including source location and water column depth, and sound speed in medium layers. Additionally, attenuation can be extracted using arrival amplitudes. We extend a method previously developed for arrival time estimation in space using particle filters, which now employs backward filters that smooth arrival time and amplitude estimates. The new approach is applied to synthetic and real data for source localization and geoacoustic inversion. Results are compared with estimates obtained using traditional arrival time estimation methods. The comparison reveals that the new technique provides more accurate results with reduced uncertainty in comparison to other approaches; uncertainty is expressed via posterior probability density functions. It is shown that the smoother plays a significant role in the quality of the time estimates and, subsequently, in geoacoustic inversion. [Work supported by ONR.]

8:25

3aUW3. Examining meso-scale variability: Sequential geoacoustic posterior probability estimation for autonomous underwater vehicle (AUV) towed array data. Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria BC, Canada V8W 3P6), Charles W. Holland (The Penn State Univ. State College, PA), and Stan E. Dosso (Univ. of Victoria, Victoria BC, Canada V8W 3P6)

This paper applies a sequential trans-dimensional Monte Carlo method to data recorded on an autonomous underwater vehicle towed array. The seismo-acoustic recordings are processed in terms of reflection coefficients as a function of frequency and angle, resulting in a sequence of data sets with small seabed footprints. The sequential particle filter applies advanced Markov chain Monte Carlo steps in combination with importance sampling techniques to carry out consecutive geoacoustic inversions along the track. In particular, the typically high information content of reflection-coefficient data is addressed by heuristic interpolating distributions implemented using annealed importance sampling to bridge between adjacent data sets with distant/disjoint high-likelihood regions. The geoacoustic parametrization, including changes in the number of sediment layers, is estimated along the track with trans-dimensional partition modeling. This trans-dimensional approach intrinsically estimates the amount of structure consistent with the data-information content and accounts for the limited knowledge about the parametrization in the geoacoustic posterior probability density estimates. The algorithm is applied to a sequence of data sets collected on the Malta Plateau with small seabed footprints (~10 m), which facilitates examination of meso-scale seabed variability.

Contributed Papers

8:45

3aUW4. Geoacoustic inversion from ambient noise data using a trans-dimensional Bayesian approach. Jorge E. Quijano, Stan E. Dosso, Jan Dettmer (SEOS, Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd., Victoria, BC V8P 5C2, jorgeq@uvic.ca), Martin Siderius, and Lisa M. Zurk (Portland State Univ., Portland, OR 97201)

Geoacoustic inversion of seabed parameters from ambient noise in shallow water is a promising technique, with potential advantages over active survey methods such as low environmental impact, easier deployment procedures, and less restrictive hardware requirements. Bayesian inversion provides a framework to estimate the posterior probability density (PPD) of geoacoustic parameters. Parameter and uncertainty estimates can be obtained from PPD moments, such as the maximum a posteriori model, means, correlations, and marginal distributions. A fundamental step in the inversion is the selection of a model parametrization (i.e., number of seabed layers) consistent with the data information content. Recent developments in Bayesian inversion of seismic and active-source acoustic data have considered a trans-dimensional approach to model selection, where the number of model parameters is treated as unknown. Different models are sampled according to their support by the data, accounting for parametrization uncertainty in the geoacoustic parameter uncertainty estimates. This work applies a trans-dimensional reversible-jump Markov chain Monte Carlo algorithm to ambient noise reflection coefficient data. The approach is demonstrated with data collected on the Malta Plateau using a vertical line array. [Work supported by ONR.]

9:00

3aUW5. Sequential geoacoustic inversion for mobile and compact source–receiver configuration. Olivier Carriére and Jean-Pierre Hermand (Environ. Hydroacoustics lab., Université libre de Bruxelles, Ave. Fr. Roosevelt 50 CP 194/05, 1050 Brussels, Belgium, ocarrier@ulb.ac.be)

The development of light instrumentation for geoacoustic characterization requires higher-frequency transmissions and inversion methods suitable for mobile configurations that efficiently combine any *a priori* knowledge about the environment or the time-varying source–receiver geometry and the acoustic transmission measurements. Sequential filtering methods provide a framework to achieve these objectives. In this paper, repeated short-range acoustic transmissions (between 750 and 1500 m) acquired on a drifting 4-hydrophone array that spans a small part of the water column, are sequentially inverted in nonlinear Kalman filters. The sequential filtering approach is demonstrated on actual data from the MREA/BP07 sea trial, with a space-coherent processing of multitone signals and a phase-coherent processing of linearly frequency modulated signals, in low (250–800 Hz) and medium (800–1600 Hz) frequency bands. The sequential inversion of repeated acoustic transmissions shows a good agreement with hydrographic and geophysical data with more stable estimates than conventional meta-heuristic inversion results and a reduced computational cost. The effect of filter covariance tuning is also examined and monitored with statistical tests. For large propagation modeling uncertainty, extended Kalman filter and ensemble Kalman filter are of comparable accuracy in parameter tracking, but the ensemble method should be preferred to get reliable associated uncertainty estimates.

9:15

3aUW6. Dynamic tomography of a gravity wave at the interface of an ultrasonic fluid waveguide. Lenaic Bonneau, Christian Marandet, Philippe Roux (ISTerre, Maison des Gosciences, 1381 rue de la piscine, 38041 Grenoble, France), Barbara Nicolas, and Jérôme Mars (Gipsa Lab., BP 46, Cedex 9, 38402 Grenoble, France)

Based on single-scattering effects, the diffraction-based sensitivity kernels which make the link between the acoustic perturbations and the medium fluctuations have been extensively studied. This research has been recently extended to perturbations at the air–water interface in an ultrasonic waveguide that scales down with a 1-km-long, 50-m-deep ocean acoustic channel in the kilohertz regime. Using array processing between two source-receiver arrays, the sensitivity kernel for both time and amplitude has been experimentally measured and theoretically calculated for each acoustic echo reverberated in the waveguide in response to a small and local surface elevation. In the present work, an ultrasonic experiment is performed during which the waveguide transfer matrix between the source-receiver arrays is recorded 50 times per second while a low-amplitude gravity wave is generated at the water–air interface. The acoustic inversion performed from the surface sensitivity kernels permits to follow the gravity wave dynamics at any point of the surface between the source and the receiver arrays.

9:30

3aUW7. Estimation of seabed parameters in the presence of highly nonlinear internal waves. Jason D. Sagers and David P. Knobles (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

Nonlinear internal waves (NLIWs) can cause significant scattering of acoustic energy within the modal spectrum. Increased sensitivity to seabed parameters may occur in cases where the primary direction of modal scattering is from lower-angle modes into higher-angle modes, where the mode functions extend deeper into the sediment. In this work, the sensitivity of acoustic propagation to seabed parameters when NLIWs are present is investigated. Comparisons are made between experimental acoustic data from Shallow Water 2006 and modeled acoustic data. The challenge of specifying the temporo-spatial behavior of the water column is addressed so that seabed parameters can be explored.

9:45

3aUW8. Validation of statistical inference using towed source data. Steven A. Stotts, Robert A. Koch, and Jason D. Sagers (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

An idealized concept of inversion modeling assumes for a given model that environmental and geometrical parameter values can be determined to match similar measured parameter values. In practice, data-model mismatch

is inherent due to noise and the inability of the model to provide a one-to-one correspondence with the real environment. Several approaches have been devised to apply statistical inference to the inversion geoacoustic parameter values, but the uncertainties calculated for these values cannot be validated directly with ground truth measurements. However, tones with calibrated source levels in data received from a towed source can provide a direct assessment of whether the estimated source level uncertainties encompass the actual error in the estimated source levels. Examples from the Shallow Water '06 experiment will be discussed to illustrate the procedure.

10:00–10:15 Break

10:15

3aUW9. Inversion of shear properties in ocean sediments from numerical Scholte wave. Zhengliang Cao and Hefeng Dong (Dept. of Electron. and Telecommun., Norwegian Univ. of Sci. and Technol., O.S. Bragstads Plass 2B, 7491 Trondheim, caozhengliang@yahoo.com)

Shear properties in marine environment are important to be measured for the stability of sediments, the tomography of shallow layers, and the conversion of acoustic energy from water-borne into sea bottom. Although surface wave methods are widely used to determine the shear wave velocity profiles at sites, inversion methods of shear attenuation have to be tackled with other difficult problems. So, an inversion method by dispersion curves of surface wave is developed and tested with the numerical Scholte wave data. The method includes three steps: extract of dispersion curves from data, calculation of forward dispersion model, and linear inversion by match of them. To simultaneously estimate dispersion curve of velocity and attenuation, Prony's method is applied for multi-modes of surface wave. Based on the methods of finding complex roots, forward model of dispersion curves is calculated with so accuracy as to get reasonable Jacoby of inversion. Different to inversion of shear velocity, the effect of spread loss is considered and investigated for Scholte wave. The performance of inversion shear wave velocity and attenuation profiles in the upper sediment layers is also examined for multi-mode dispersion curves.

10:30

3aUW10. Broadband geoacoustic inversion on a horizontal line array. A. Tolstoy (1538 Hampton Hill Circle, McLean, VA 22101, atolstoy@ieee.org)

Shallow water bottom properties can be very difficult to estimate and are usually done so by means of vertical line arrays over a range of frequencies. This work will examine the behavior of the minimization broadband processor at low frequencies (25–100 Hz) for a variety of horizontal line arrays at endfire in simulated Shallow Water 2006 environments. Do we ever have enough sensitivity and resolution to estimate any of the bottom properties when using a horizontal array?

10:45

3aUW11. Remote-sensing of hydrodynamics in Tokyo Bay and Bakan Strait. Tokuo Yamamoto (Appl. Marine Phys. Div. RSMAS, Univ. of Miami, 4600 Rickenbacer Cswy, Miami, FL 33149, tyamamoto@bellsouth.net)

In the inertia range, turbulent energy is concentrated in eddies. The eddies interact, dissipate the kinetic energy and reduces the eddy size. The turbulent velocity fields are measured by inverting the acoustic wave attenuation by volume scattering. The eddy viscosity ϵ is the rate of kinetic energy dissipation of the turbulence. Acoustic wave dissipation is roughly proportional to (velocity intensity fluctuations) ϵ, 2/3. The Reynolds stresses are also induced by the velocity intensity fluctuations ϵ, 2/3. The Reynolds stresses take various forms, such as Rip currents, wave set-ups, etc. The acoustic wave volume scattering, transmission loss by volume scattering, turbulent fields, Reynolds stresses, and eddy viscosity fields are inverted from eight point transmission lines in two completely different turbulent flow fields. They are Tokyo Bay ($\epsilon \sim 0.02$) and Bakan Strait ($\epsilon \sim 1.2$). The Reynolds stresses are in Tokyo Bay ($\sim$33 kPa) and in Bakan Strait ($\sim$1200 kPa). The side by side comparisons of the two data sets reviles the quite complicated differences in their responses to turbulent current field which will be presented in this paper.

11:00

3aUW12. Efficient acoustic uncertainty estimation for uncertain source depth. Kevin R. James (2010 Lay Auto Lab., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109) and David R. Dowling (Univ. of Michigan, Ann Arbor, MI 48109)

Uncertainty in source depth is a prevalent source of uncertainty in underwater acoustic measurements and predictions. This uncertainty can arise from limited information about a source, or the movement of a source between measurement intervals, such as the bobbing of a ship's propeller near the ocean surface. This presentation describes the effect of source depth uncertainty on transmission loss predictions at frequencies from a few hundred hertz to a few kilo-hertz in both range-independent and range-dependent environments, as well as the relative contribution of source depth uncertainty compared to other likely sources of environmental uncertainty. Several approaches to estimating the resulting transmission loss uncertainty are considered, including direct simulation with Taylor series approximations, the field-shifting method, polynomial chaos expansions, and a reciprocity-based solution. In particular, it will be shown that a reciprocity-based algorithm provides exact results in range-independent environments with a single field calculation but can require additional calculations to obtain transmission loss versus range information in a range-dependent environment. Strategies for limiting the computational effort and possibilities for implementing the reciprocity approach within the US Navy's existing transmission loss uncertainty estimation technique, the Uncertainty Band algorithm, are presented. [Work sponsored by the Office of Naval Research, Code 322OA.]

11:15

3aUW13. Estimates of the water column sound-speed field in three dimensions using a linearized perturbative technique. Christopher M. Bender, Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029), and Preston S. Wilson (Univ. of Texas at Austin, Austin, TX 78712-0292)

A perturbative inversion scheme [S. D. Rajan *et al*., J. Acoust. Soc. Am. **82**, 998–1017 (1987)] is applied to estimate the water-column sound-speed field in a three-dimensional volume of the shallow ocean. The input data to the inversion are estimates of modal travel time data calculated from measurements from a distributed network of sources and receivers. The differences in the data are used as the basis for updating, or perturbing, a background model through the inversion algorithm to arrive at a solution. According to this technique, the horizontal plane is divided into a grid of range-independent regions and an estimate of the depth-dependent sound-speed profile within each region is obtained. The linearized perturbative technique is particularly well-suited for the three-dimensional inverse problem as the solution typically obtained in less than 20 iterations. During this talk, the method will be validated using synthetic data which are representative of oceanographic range-dependence in shallow water environments. The effect of the density and distribution of sources and receivers on the resolution of the inversion results will be discussed. [Work supported by ARL:UT IRD.]

11:30

3aUW14. A new broadband matched field processor? A. Tolstoy (1538 Hampton Hill Circle, McLean VA 22101, atolstoy@ieee.org)

It is well known that the use of multiple frequencies helps to reduce uncertainty and to improve resolution for matched field processing. This can be extremely important for the estimation of geoacoustic (bottom) parameters and usually means averaging ambiguity surfaces either incoherently or coherently (if the source spectrum is known) for the linear or minimum variance processors. However, a new approach requires that only the minimum processor values be retained over all the frequencies considered. This presentation will show how this method improves resolution and sidelobe reduction (for the simple incoherent method) both for the linear and for the minimum variance processing. Of course, one never gets something for nothing. This improvement comes at the cost of using only those frequencies which show high-matched field values, usually the low frequencies (25–75 Hz), for geoacoustic inversion. If some frequency component is strongly degraded, then the method will not work. The behavior of this new

processor (to be called the "minimization broadband processor") is demonstrated on simulated shallow water data.

11:45

3aUW15. Bottom reflections from rough topography in the long-range ocean acoustic propagation experiment (LOAPEX). Ilya A. Udovydchenkov, Ralph A. Stephen, Timothy F. Duda (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Peter F. Worcester, Matthew A. Dzieciuch (Univ. of California at San Diego, La Jolla, CA 92093), James A. Mercer, Rex K. Andrew (Univ. of Washington, Seattle, WA 98105), and Bruce M. Howe (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

Data collected during the 2004 long-range ocean acoustic propagation experiment form a unique set of measurements containing information about absolute intensities and absolute travel times of acoustic pulses at long ranges. This work primarily focuses on sound reflected from the seafloor at the shortest transmission range of the experiment, approximately 50 km. At this range, bottom-reflected energy interferes with arrivals refracted by the sound channel. Stable bottom-reflected arrivals are seen from broadband transmissions from acoustic source at 800 m depth with 75 Hz center frequency, and from a source at 350 m depth with 68.2 Hz center frequency. Standard acoustic propagation modeling tools such as parabolic equation solvers can be successfully used to predict acoustic travel times and intensities of the observed bottom-reflected arrivals. Inclusion of range-dependent bathymetry is necessary to get an acceptable model-data fit. Although there is no direct ground truth for actual sub-bottom properties in the region, a good fit can be obtained with credible sub-bottom properties. The potential for using data of this type for geoacoustic property inversion in the deep ocean will be discussed. [Work supported by ONR.]

WEDNESDAY MORNING, 2 NOVEMBER 2011 DOVER, 8:30 TO 9:45 A.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

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

G. J. Frye, Vice Chair ASC S3
Frye Electronics, Inc., P.O. Box 23391, Tigard OR 97281

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note–those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, November 1, 2011.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

Session 3pAAa

Architectural Acoustics and Musical Acoustics: Variable Acoustics on Concert Stages II

Bill Dohn, Cochair
Dohn and Associates, Inc., 630 Quintana Rd., Morro Bay, CA 93442

Michelle C. Vigeant, Cochair
Mechanical Engineering, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117-1599

Invited Paper

1:00

3pAAa1. Changes in on-stage and in-house listening conditions with adjustments to over-stage canopy height and tilt: Field investigations in two concert halls. Bill Dohn (Dohn and Assoc., Inc., 630 Quintana Rd., #312, Morro Bay, CA 93442, bill.dohn@dohnandassociates.com)

On-stage and in-house measurements were made in two concert halls with adjustable over-stage canopies to assist with future designs and verify recommended canopy settings for facility users. Results of measurements with varying canopy heights and tilt angles will be shared, along with descriptions of the two (very different) canopy designs and the two (very similar) concert hall designs. User preferences for canopy settings in both halls will also be discussed.

Contributed Papers

1:20

3pAAa2. Working with student musicians on developing standard settings for concert stages with adjustable acoustics. David W. Kahn (Acoust. Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801)

The acoustician's work on concert stages with adjustable acoustics usually includes a so called "tuning" process. This tuning process is unique and different for students attending music school as compared with, for example, a professional orchestra. We describe a tuning process, working with student musicians, at a new concert hall in Texas. Adjustable elements include movable ceiling elements, movable doors on the upper wall surfaces surrounding the concert platform, and retractable curtains that cover both the upper and lower wall surfaces surrounding the platform.

1:35

3pAAa3. Altering stage acoustics using time variant electro-acoustic enhancement. Steve Barbar (30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

Time variant electro-acoustic enhancement systems have been successfully installed in a variety of venues and applications since 1990. Many of these systems have included electro-acoustic support for performers on stage, as well as increased direct sound from virtual reflectors, in both indoor and outdoor venues. Examples of these systems and venues are described, including: The Mormon Tabernacle Choir, Vienna Philharmonic Orchestra, and Arnold Schoenberg Choir, Grant Park Symphony, Adelaide Symphony Orchestra, Midland Symphony Orchestra, and Stonebriar Community Church.

1:50

3pAAa4. Variable acoustics systems and their impact on concert platforms. Todd L. Brooks, Clemeth L. Abercrombie, Nathan A. Pape, Damian J. Doria, and Tateo Nakajima (Artec Consultants Inc., 114 W. 26th St. 10th Fl., New York, NY 10001, tlb@artecconsultants.com)

The modern purpose-built concert hall must accommodate an extremely wide range of program, ranging from unamplified solo and chamber music to large-scale symphony and amplified events. Adjustable systems designed for both acoustical and functional flexibility are essential to adapt the stage platform environment to the needs of these different performance types. The successful modern concert hall may include adjustable overhead canopies, adjustable absorption systems, orchestra riser platforms (possibly mechanized), chorus riser wagons, and platform extension/reduction lifts. We will share our experiences and some lessons learned in a variety of recently completed and in-construction projects. Audible differences resulting from differences in architectural contexts will be discussed. We will link possible correlations between aural variations and objective measures with subjective attributes important to the performers.

2:05

3pAAa5. Choir hearing responses: Rehearsal versus performance configurations. Glenn E. Sweitzer (Sweitzer LLP, 4504 N Hereford Dr. Muncie, IN 47304, glenn.sweitzer@gmail.com)

Choir responses to hearing (parts) in a rehearsal room are compared with those for its associated performance stage. Anonymous responses from each choir member are gathered via a hand-held battery-operated device that enables scaled responses (5). The protocol is administered in the rehearsal room for the choir configured by part versus mixed, in straight rows versus curved rows, standing on a flat floor versus risers, and facing a diffusing wall versus an adjustable absorptive wall. This protocol is repeated on the performance stage, except that the choir faces the audience only. Preliminary results suggest that responses vary widely by configuration and suggest that choir response has been largely ignored in the design of and operation of choir rehearsal and performance facilities.

2:20

3pAAa6. Feasibility of converting a conference hall to a music performance hall. L. Y. Cheung (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong, China, louisa.cheung@polyu.edu.hk) and S. K. Tang (The Hong Kong Polytechnic Univ., Hong Kong, China, besktang@polyu.edu.hk)

A 3-D-simulation for improving the acoustics in a conference hall was done. This aimed at investigating the possibility of converting it for musical performance. A room modeling software, Odeon, was used to simulate the

effects of adding varies reflectors on the stage with its existing demountable acoustic shell. The following cases were put into simulations: a bare stage, stage with demountable acoustic shell, and stage with acoustic shells with various hanging reflectors above the stage. The additional hanging reflectors with the existing acoustical shell reduce the reverberation time in the hall in most of the cases simulated; however, they increase the average energy levels in the hall. The acoustical shell with the reflector gives a larger energy ratio in average across the octave bands that make the hall more suitable for musical performance. [L. Y. Cheung is supported by the Hong Kong Polytechnic University.]

WEDNESDAY AFTERNOON, 2 NOVEMBER 2011 SUNSET, 1:00 TO 3:00 P.M.

Session 3pAAb

Architectural Acoustics: Workshop for American Institute of Architects Continuing Education Units

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square, Ste. 103, Vernon, CT 06066

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

Invited Papers

1:00

3pAAb1. Technical Committee on Architectural Acoustics short course presentation material. K. Anthony Hoover (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects, called "Architectural Acoustics." An architect can earn one continuing education unit by attending this short course, if it is presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, and finish treatments. This paper will cover the course material in order to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this workshop and membership in TCAA are required.

2:00

3pAAb2. American Institute of Architects Continuing Education System Provider Registration and Reporting Requirements. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Sq., Ste. 103, Vernon, CT 06066, bbrooks@brooksacoustics.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects. The TCAA short course is called "Architectural Acoustics" and attendance at this 1-h long course can earn an architect one continuing education unit with HSW Credit (Health Safety and Welfare). This paper will cover the administrative requirements of the AIA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork, so that AIA members may receive credit for the course. Also, the manner in which the course is given is dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course.

Session 3pAB

Animal Bioacoustics: Acoustics for Saving Endangered Species II

Tomonari Akamatsu, Cochair
National Research Inst. of Fisheries Engineering, Hasaki, Kamisu, Ibaraki, 314-0408, Japan

Susan E. Parks, Cochair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Invited Papers

1:15

3pAB1. Exploring the use of acoustics as a tool in male elephant/human conflict mitigation. Caitlin E. O'Connell-Rodwell (Dept. of Otolaryngol., Head & Neck Surgery, Stanford Univ. School of Medicine, 801 Welch Rd., Stanford, CA 94305, ceoconnell@stanford.edu), Rehabeam Erckie (Etosha Natl. Park, Namibia), Werner Kilian (Etosha Ecological Inst., Etosha Natl. Park, Namibia), Jason D. Wood (Univ. of Washington, Seattle, WA, 98195), Colleen Kinzley (Oakland Zoo, Oakland, CA 94605), Timothy C. Rodwell (Univ. of California, San Diego, La Jolla, CA 92093), and Joyce H. Poole (ElephantVoices, Sausalito, CA 94965)

Elephant/human conflict mitigation solutions have been explored with varying degrees of success. We present findings on a potential acoustic tool to reduce negative outcomes of male elephants entering agricultural areas in the region northeast of Etosha National Park, Namibia. We monitored elephant traffic within and outside the park boundary using GPS collars on five male elephants with known hormonal status as well as information on the frequency and location of fence breaks. Male elephants in the hormonal state of musth have an increased range, often extending outside the protected area. We explored the feasibility of attracting musth males away from potential conflict areas noninvasively, using estrus calls. We played back estrus calls to known individual subadult ($n = 9$) and adult musth ($n = 9$) and nonmusth ($n = 6$) male elephants and show that adult musth and subadult nonmusth males were much more likely to respond and approach the source than nonmusth adult males ($p = 0.029$ and $p = 0.009$, respectively) with an equal level of intensity ($p = 0.822$). Our findings suggest that the use of acoustics may serve as an effective tool in noninvasive male elephant/human conflict mitigation, depending on the age and hormonal status.

1:35

3pAB2. Finding baiji and freshwater finless porpoises in the Yangtze River, China. Tomonari Akamatsu (Natl. Res. Inst. of Fisheries Eng., Hasaki, Kamisu, Ibaraki 314-0408, Japan, akamatsu@affrc.go.jp), Lijun Dong, Ding Wang, Kexiong Wang, Songhai Li, Shouyue Dong, Xiujiang Zhao (Chinese Acad. of Sci., Wuhan 430072, P. R.China), and Satoko Kimura (Kyoto Univ., Kyoto 606-8501, Japan)

A stereo passive acoustic event recorder (A-tag) has been applied for range-wide monitoring of baiji and finless porpoises in China. As the pilot study, two research vessels were operated in 1700 km historic habitat of both species from Yichang to Shanghai in 2006. There was no detection of baiji, but 204 and 199 porpoises were counted acoustically by two vessels, respectively. In order to investigate the population trends of cetaceans, periodical survey is necessary. We installed A-tag on the cargo ship, which was operated 1100 km in the river once every month. An average of 6059 clicks and 95 porpoises were acoustically detected in each survey. Detected group sizes of the animals in 120-s time window were not significantly different among the surveys, but the distribution pattern suggested seasonal migration. The animals were detected in most of the survey range except two gap sections with 40 and 60 km lengths, down from Wuhan and Nanjing cities, respectively, where no animals were detected in the first three surveys. Fragmentation of population by anthropological factors was concerned. The cargo ship based passive acoustic survey was effective to monitor the distribution and population trend over time.

1:55

3pAB3. Acoustic monitoring of beluga whales (*Delphinapterus leucas*) in Cook Inlet, Alaska. Manuel Castellote (Natl. Marine Mammal Lab., NOAA Fisheries, Seattle, WA 98115, manuel.castellote@noaa.gov), Robert J. Small (Univ. of Alaska Fairbanks, Juneau, AK), Shannon Atkinson (Univ. of Alaska Fairbanks, Juneau, AK), Marc O. Lammers (Hawaii Inst. of Marine Biology, Kaneohe, HI), Justin jenniges (Univ. of Alaska Fairbanks, Juneau, AK), Anne Rosinski (Hawaii Inst. of Marine Biology, Kaneohe, HI), Chris Garner (U.S. Air Force, Joint Base Elmendorf-Richardson, Anchorage, AK), Sue Moore (NOAA Fisheries, Seattle, WA), and Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI)

Cook Inlet belugas (CIB) form a small geographically and genetically isolated population, endangered under the U.S. Endangered Species Act. They summer in the northern end of the inlet, but their wintertime distribution is essentially unknown. Factors impeding the recovery of this population over the last decade are unknown, yet could include anthropogenic activities that impact their acoustic ecology, including coastal development, oil and gas exploration, and shipping and military activities. Beginning in 2008, a cooperative research project has acquired new information on background noise levels and seasonal presence of CIB using passive acoustic monitoring. Mooring packages containing ecological acoustic recorders and echolocation loggers (C-PODs) have been deployed at 10 sites for

continuous monitoring. It is a challenging environment for acoustic monitoring because of extreme tides and currents, sediment dynamics, debris from rivers, and seasonal ice. Noise from both natural and anthropogenic sources often make beluga call detection challenging. However, the effort to date has met with success and is providing valuable insights into beluga movement patterns and the acoustic environment they face. This methodology also allows monitoring other odontocetes such as killer whales (*Orcinus orca*), detected mostly in the lower inlet, and harbor porpoise (*Phocoena phocoena*) detected throughout the inlet.

Contributed Papers

2:15

3pAB4. Distribution pattern of finless porpoises at the junction of the Yangtze River and Poyang Lake observed by towed passive acoustic device. Satoko Kimura (Graduate School of Informatics, Kyoto Univ., 606-8501 Kyoto, Japan, sk0130@bre.soc.i.kyoto-u.ac.jp), Tomonari Akamatsu (Natl. Res. Inst. of Fisheries Eng., Ibaraki 314-0408, Japan), Songhai Li (Chinese Acad. of Sci., Wuhan 430072, China), Lijun Dong, Kexiong Wang, Ding Wang (Chinese Acad. of Sci., Wuhan 430072, China), and Nobuaki Arai (Kyoto Univ., 606-8501 Kyoto, Japan)

The Yangtze finless porpoise (Neophocaena asiaeorientalis asiaeorientalis) is an endangered freshwater porpoise subspecies unique to the Yangtze River basin. Without immediate conservation measures, it could soon become extinct, just as the Yangtze River dolphin (baiji, Lipotes vexillifer). We report seasonal change in the local distribution of the porpoises living in the conjunction area of the middle reaches of the Yangtze River, side streams, and appended Poyang Lake. A towed stereo acoustic data-logger, A-tag, was used to detect echolocation signals and sound source bearing angles. The independent sound source directions provided the number of animals present, not just the number of sounds. Passive acoustic surveys were performed regularly from May 2007 to August 2010. The water level was highest in August (summer) and lowest in February (winter) and at mid-level in May (spring) and November (autumn). The average number of porpoises detected in 11 surveys conducted in different seasons varied from 0.53 to 1.26 individuals per km. No significant trend of reducing number of porpoises was detected during 3-y monitoring. The distribution of the porpoises was seasonally site-specific. In May and August, the animals were detected more often at river junctions than in the lake, but vice versa from November to February.

2:30

3pAB5. Implications of vocalizations during giant panda breeding. Jennifer L. Keating (Appl. Animal Ecology, Inst. for Conservation Res., San Diego Zoo Global, 15600 San Pasqual Valley Rd., Escondido, CA 92027, jkeating@sandiegozoo.org), Tracey K. Brown, Nancy G. Caine (California State Univ. San Marcos, San Marcos, CA 92096), Ann E. Bowles (Hubbs-SeaWorld Res. Inst., San Diego, CA 92109), Benjamin D. Charlton (Univ. of Vienna, Vienna, Austria), and Ronald R. Swaisgood (Inst. for Conservation Res., San Diego Zoo Global, Escondido, CA 92027)

Silent most of the year, giant pandas (*Ailuropoda melanoleuca*) engage in sustained and diverse vocal behavior during their brief annual reproductive window. These vocalizations suggest a complex communication repertoire and play an important role in facilitation of mating. To increase our understanding of their vocal communication, we analyzed male and female vocalizations during breeding interactions. Digital audio recordings were collected during breeding seasons of 2008, 2009, and 2011 at zoological facilities in San Diego, California, and China, and were processed using SOUNDTRACK PRO 2.0 and RAVEN PRO 1.4 software. Seven types of vocalizations were identified ("bark," "moan," "growl," "squeal," "chirp," "bleat," and "copulation call"). We compared vocalizations from 32 confirmed copulations and 37 non-copulatory breeding sessions of 31 individuals (24 females, 7 males). They revealed differences in vocalizations produced before, during, and after intromission. During intromission, the male produces a copulation call that distinguishes successful breeding encounters from unsuccessful pairings acoustically. This call has not been described previously. The results shed light on the motivational and functional significance of panda vocalizations during mating encounters. They can be used in captive breeding programs to promote reproduction in this endangered species. [Work supported by ZSSD, CSUSM, and CCRCGP.]

2:45

3pAB6. Behavioral audiogram of the giant panda (*Ailuropoda melanoleuca*): Preliminary results. Megan A. Owen, Jennifer L. Keating (Appl. Animal Ecology, Inst. for Conservation Res., San Diego Zoo Global, 15600 San Pasqual Valley Rd., Escondido, CA 92027), Samuel L. Denes (The Penn State Univ.), Kathy Hawk, Juli Boroski, Angela Fiore (Collections Husbandry Sci. San Diego Zoo Global), and Ronald R. Swaisgood (Appl. Animal Ecology, Inst. for Conservation Res., San Diego Zoo Global)

We used behavioral techniques to assess the hearing sensitivity of four, critically endangered, giant pandas at the San Diego Zoo. Study subjects included one adult male (age 19), two adult females (ages 5 and 19), and one sub-adult female (age 3). We used a down-up staircase presentation order and a go/no-go response paradigm and thus far have measured hearing thresholds between 250 Hz and 31.5 kHz. Test stimuli were 500 ms shaped tones, and catch trials represented 30% of presentations. All subjects were trained using positive reinforcement. Preliminary results suggest that giant pandas have good hearing sensitivity between 8 and 14 kHz, and best sensitivity was centered at 12.5 kHz. Low frequency hearing sensitivity declined at 250 Hz for all subjects. All bears retained functional hearing at 31.5 kHz: the younger females could hear tones as low as 15 dB, and the adults could hear to 25 dB. Preliminary results suggest that panda hearing sensitivity is similar to that of other terrestrial carnivores studied to date. Hearing sensitivity data will enhance the understanding of how anthropogenic noise may impact both free-ranging and captive giant pandas.

Session 3pAO

Acoustical Oceanography: Munk Award Lecture

Martin Siderius, Chair
Electrical and Computer Engineering, Portland State Univ., 1900 S.W. 4th Ave., Portland, OR 97201

Chair's Introduction—2:10

Invited Paper

2:25

3pAO1. Underwater acoustics and acoustical oceanography. William A. Kuperman (Scripps Inst. of Oceanogr., Marine Physical Lab., MC0238, 9500 Gilman Dr., La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

Though underwater acoustics (UW) and acoustical oceanography (AO) have different goals, their overlap often makes efforts in these areas indistinguishable. In this talk I emphasize their similarities while also discussing their different approaches. Topics to be briefly covered in this review will include the environmental acoustic aspects of propagation, noise, and inversion with examples from modeling and experiments. Among the interesting outcomes of the differences between these fields is that the emphasis on inversion in AO has increased the accuracy requirements for UW.

WEDNESDAY AFTERNOON, 2 NOVEMBER 2011 ROYAL PALM 5/6, 1:45 TO 3:00 P.M.

Session 3pBA

Biomedical Acoustics: General Biomedical Ultrasound

Robert J. McGough, Chair
Electrical and Computer Engineering, Michigan State Univ., 2120 Engineering Bldg., East Lansing, MI 48824

Contributed Papers

1:45

3pBA1. Transthoracic *in vivo* measurements of porcine myocardial viscoelasticity using Lamb wave dispersion ultrasound vibrometry. I. Z. Nenadic, M. W. Urban, C. Pislaru, and J. F. Greenleaf (Ultrasound Res. Lab., Mayo Clinic College of Medicine, 200 1st Street SW, Rochester, MN 55905)

Diastolic dysfunction is characterized by the stiffening of the left-ventricular myocardium. An imaging technique capable of quantifying viscoelasticity of the left-ventricle would aid clinicians in evaluating diastolic function. Lamb wave dispersion ultrasound vibrometry (LDUV) is an ultrasound-based method for measuring viscoelasticity of plate-like materials by exciting Lamb waves and measuring change of wave velocity as a function of frequency (dispersion). Lamb wave dispersion equation is fit to the dispersion data to estimate mechanical properties. We report *in vivo* transthoracic studies aimed at quantifying elasticity and viscosity of a beating porcine left-ventricle using LDUV. A single ultrasound probe operating at 5 MHz was used for impulse excitation in the myocardium and to track the wave motion. A Fourier space-time analysis of the motion was used to obtain Lamb wave velocity dispersion. ECG-gated transthoracic *in vivo* measurements of group velocity, elasticity, and viscosity throughout a single heart cycle were obtained. Group velocity, elasticity, and viscosity in the frequency range from 50 to 500 Hz increased by at least two-fold from diastole to systole, consistent with contraction and relaxation of the myocardium.

2:00

3pBA2. Evaluation of the power-law impulse response for liver in the time- and frequency-domains. Christopher T. Johnson and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 1213 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

Ultrasound attenuation in biological media often follows a frequency-dependent power-law relationship. Power-law attenuation is accompanied by dispersion in which phase velocity also changes as a function of frequency. The power-law wave equation has exact time- and frequency-domain Green's function solutions that are used in numerical evaluations of the Rayleigh–Sommerfeld diffraction integral. These Green's functions contain stable probability density functions that are evaluated in the time-domain using the STABLE toolbox and in the frequency-domain by evaluating the characteristic function of a stable distribution. The impulse response for a circular piston is evaluated on-axis in the time- and frequency-domain using values for human liver. The results show the accuracy of the time-domain impulse response calculation is dependent upon

3p WED. PM

adequate spatial sampling of the piston face. The frequency-domain impulse response exhibits aliasing caused by wrap around of the heavy tail of the stable distribution. Frequency-domain accuracy is dependent upon the spatial sampling of the piston face and the density of the frequency samples. Numerical evaluations show that the frequency- and time-domain calculations converge to the same result. [This work was supported in part by NIH Grant R01 EB012079.]

2:15

3pBA3. Frequency domain calculations of pulse-echo ultrasound signals with the fast nearfield method. Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 2120 Eng., East Lansing, MI 48824, mcgough@egr.msu.edu)

Simulations of pulse-echo signals transmitted and received by an ultrasound phased array are evaluated in the frequency domain using FOCUS, and the results are compared with the output of Field II. The frequency domain simulation approach in FOCUS calculates the fast Fourier transform of the input signal, evaluates the individual frequency components of the transmit and receive pressure transfer functions using the fast nearfield method, and then computes the inverse fast Fourier transform of the product to obtain a numerical representation of the received time signal. With this approach, synthetic aperture signals are simulated in MATLAB for each pair of transmit and receive elements for three transmit elements and sixteen receive elements. Each element is excited with a 3 cycle Hanning-weighted input pulse with a 3 MHz center frequency. The temporal sampling of the computed signal in FOCUS is 12 MHz. To achieve approximately the same accuracy, Field II requires a temporal sampling of 4 GHz. Examples of simulated radio-frequency signals, the envelopes of these signals, and the associated numerical errors will be demonstrated for each method, and formulas for converting the pulse-echo outputs generated in FOCUS and Field II into equivalent quantities will be shown.

2:30

3pBA4. Estimation of two-dimensional strain rate based on high frame rate ultrasound imaging method. Hong Chen and Jian-yu Lu (Dept. of Bioengineering, Univ. of Toledo, 2801 W. Bancroft St., Toledo, OH 43606)

Strain rate (SR) measurement of heart tissue based on ultrasound images provides useful information of tissue hardness for diagnosing heart diseases. However, current SR estimation methods utilizing speckle tracking technique are based on conventional delay and sum (DS) imaging method, which causes skewed heart image resulting in inaccurate SR. To overcome the problem, a method to combine high frame rate (HFR) imaging method with speckle tracking technique was proposed. Using only one or a few transmissions for each image; compared with 91 for DS, new method can get a snapshot of moving targets, avoiding the skewing problem in DS. Two studies, with simulated and experimental echo data, respectively, were performed to verify the method. Both plane wave and limited diffraction beam (LDB) were studied for HFR. Both SR estimations in lateral and axial directions were calculated for the new and conventional ultrasound imaging methods. Results show that the new method has comparable or lower velocity errors than DS and more accurate SR, especially in lateral direction. Moreover, it can measure high velocity for other applications such as blood flow measurement. With the full view of heart image, SR of interest can be localized and then accurately estimated.

2:45

3pBA5. Transducer designs and simulations for high frequency scanning acoustic microscopy for applications in exploring contrast mechanism and the mechanical properties of biological cells. Yada Juntarapaso, Richard L. Tutwiler (Graduate Program in Acoust., The Penn State Univ., Univ. Park, PA 16802), and Pavlos Anastasiadis (Univ. of Hawaii, Honululu, HI 96822)

Scanning acoustic microscopy (SAM) has been extensively accepted and utilized for acoustical cellular and tissue imaging including measurements of the mechanical and elastic properties of biological specimen. SAM provides superb advantages: it is a noninvasive method; it can measure mechanical properties of biological cells or tissues; and fixation/chemical staining is not necessary. The first objective of this research is to develop a program for simulating the images and contrast mechanism obtained by high-frequency SAM. Computer simulation algorithms based on MATLAB® were built for simulating the images and contrast mechanism. The mechanical properties of HeLa and MCF-7 cells were computed from the V(z) measurement data. Algorithms for simulating V(z) responses involved calculation of the reflectance function and were created based on ray theory and wave theory. The second objective is to design transducer arrays for SAM. Theoretical simulations based on Field II programs of the high frequency ultrasound array designs were performed to enhance image resolution and volumetric imaging capabilities. The new transducer array design will be state-of-the-art in improving the performance of SAM by electronic scanning and potentially providing a four-dimensional image of the specimen. Phased array beam forming and dynamic apodization and focusing were employed in the simulations.

Session 3pEA

Engineering Acoustics and Biomedical Acoustics: Micromachined Acoustic Transducers II

Neal Hall, Cochair
Electrical and Computer Engineering Dept., Univ. of Texas at Austin, 1 University Station, Austin, TX 78712

Won-Kyu Moon, Cochair
Dept. of Mechanical Engineering, Pohang Univ. of Science and Tech., Pohang, 790-784, Korea

Contributed Papers

1:00

3pEA1. A high-dispersion regime of the zero-order symmetric Lamb wave mode for ultrasonic nondestructive evaluation. Li-Feng Ge (Anhui Univ., 3 Feixi Rd., Bldg. 166, Rm. 304, Hefei, Anhui 230039, China, lfge@ahu.edu.cn)

The zero-order symmetric Lamb wave mode (S0 mode) has been less studied than antisymmetric one (A0 mode). The S0 mode excited by micromachined electrostatic airborne ultrasonic transducers can be used for nondestructive evaluation of multilayer or composite materials, which deserves special attention. The dispersion curve of the S0 mode is investigated in depth and determined accurately by a three dimensional Plot Method presented previously [Ge, J. Acoust. Soc. Am. **126**(4), 2281 (2009); Proc. Meet. Acoust. **8**, 065003 (2011)]. It is revealed that as the product of the transverse wave number and thickness increases from a low limit to infinitive the phase velocity of the S0 mode decreases from transverse wave velocity to surface acoustic wave velocity. The low limit determined is 3.46 for Poisson ratio 0.34, and will be 3.31 and 3.05 for 0.28 (steel) and 0.17 (silicon), respectively. Also, it is seen that at a small regime over the limit the S0 mode is highly dispersive. Further, an approximate formula to determine the dispersion relation of the regime is derived analytically for the convenience of practical applications. Since the zero-order mode carries more energy than higher-order modes, the high-dispersion regime is significant particularly for ultrasonic nondestructive evaluation. [Work supported by NSFC (60774053).]

1:15

3pEA2. Split domain antiresonance in micromachined lithium niobate. Igor Ostrovskii and Lucien Cremaldi (Dept. of Phys. and NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677, iostrov@phy.olemiss.edu)

The micromachined periodically poled LiNbO3 (PPLN) demonstrates acoustic superlattice properties. Different acoustic modes in PPLN wafer have their stop-bands with low FL and upper FU cutoff frequencies. The micromachined transducers based on such structures operate at frequencies near FL and FU; they have a strong domain resonance-antiresonance when acoustic half-wavelength is close to domain length. In this work, we investigate structures with 300- and 450-micron long domains micromachined in a 0.5-mm-thick Z-cut lithium niobate wafer. The metal electrodes are deposited onto multi-domain structures having up to 90 total inversely poled domains. Experimentally, two modes of operation are investigated: (1) acousto-electric current excitation of domain vibrations when rf current passes through the whole multidomain structure and (2) plate-waves excitation and detection at two opposite ends of domain pattern when acoustic vibration travels through the multidomain pattern. In both cases, the split antiresonances are observed, and they are characterized by two main minima coinciding with the FL and FU frequencies. The difference (FL − FU) constitutes the stop-band for propagating zero antisymmetric mode, at which PPLN transducer operates. The experimental data are in agreement with theoretical calculations. Possible application of the split antiresonance effect for micromachined ultrasonic transducers is discussed. [This work is made possible in part due to the research grant "Nonlinear vibrations of piezoelectric resonators," UM, 2011.]

1:30

3pEA3. Micromachined ultrasonic transducers and nonclassical nonlinear effects due to domains. Igor Ostrovskii and Andriy Nadtochiy (Dept. of Phys. and NCPA, Univ. of Mississippi, University, MS 38677, iostrov@phy.olemiss.edu)

Two-dimensional ultrasonic transducers and delay lines fabricated on micromachined $LiTaO_3$ and $LiNbO_3$ plates are demonstrated. The 0.5-mm-thick commercial wafers are inversely poled by applying an external electric field to get periodic and aperiodic multidomain structures. The metal electrodes are deposited on the structures to excite ultrasonic vibrations by applying radio frequency voltage. The domain lengths are in a broad range from 16 μm to 1 mm with corresponding frequencies from 347 to 3 MHz. The finite element method is used for computer simulation of the micromachined transducers. Experimental results are in agreement with theoretical calculations. All transducers have a so called domain resonance and operate at frequencies near this resonance. The highest vibration amplitude is usually observed when acoustic wavelength is two times greater than a single domain length. The multidomain structures have their own nonclassical nonlinear properties, which may be explained by domain kinetics and their collective behavior. For instance, an efficiency of transduction may depend on such physical effects like domain reverberations and so called stop-bands in an acoustic superlattice, which in turn are specific functions of frequency and other conditions of excitation. Possible applications of the piezoelectric micromachined transducers and vibrators are discussed. [This work is made possible in part due to the research grants "Nonlinear vibrations of piezoelectric resonators," UM, 2011; "Multidomain plate acoustic wave devices," UM, 2007.]

1:45

3pEA4. Micromachining of acoustic transducers and arrays. Zhihong Wang, Xixiang Zhang, and Xianbin Wang (Adv. Nanofabrication Core Lab., King Abdullah Univ. of Sci. and Technol., Thuwal 23955-6900, Saudi Arabia, zhihong.wang@kaust.edu.sa)

Microelectromechanical systems (MEMS) technology has been used to fabricate miniature acoustic transducers for a variety of applications in different media and frequency ranges. The consistency in performance of individual transducer afforded by batch-fabrication also makes it feasible to employ MEMS transducer arrays in beam forming or beam steering technique. This paper will first introduce our own research work on micromachining of piezoelectric acoustic transducers, including sol–gel thin film piezoelectric microphone for audio application; composite thick film piezoelectric micromachined ultrasonic transducer array for biomedical imaging; and transducer arrays fabricated by Si-PZT bonding technology for

hydrophone and audio beam applications. The preliminary results on receiving and transmitting performance of the fabricated transducers and arrays will also be reported. In addition, a scanning acousto-optic microscope, which uses a single micromachined acousto-optic sensor and interferometry, will also be introduced. With this technique, it is possible to map a spatial sound field distribution at micrometer resolution.

2:00

3pEA5. A micro-machined microphone based on field-effect-transistor and electrets. Kumjae Shin (Dept. of Mech. Eng., POSTECH, PIRO 416 Hyoja-Dong, Nam-Gu, Gyungbuk, 790-784 Pohang, South Korea, forhim13@postech.ac.kr), Yub Je (PIRO 416 Hyoja-Dong, Gyungbuk, 790-784 Pohang, South Korea), Haksue Lee (Jinhae, Gyungbuk, South Korea), James Edward West (Johns Hopkins Univ.), and Wonkyu Moon (PIRO 412 Hyoja-Dong, Gyungbuk, 790-784 Pohang, South Korea)

Micro-machined microphones are attracting attention of industry because of their benefit of size over conventional ones. Since most of micro-machined microphones are capacitive sensors, the sizes of their electrodes determine the low frequency noise level that increases with inverse of frequency (1/f). Therefore, the size of microphone itself becomes larger than one that can be fabricated. Here, we introduce a micro-machined microphone that can overcome the limit of capacitive microphones. The proposed microphone is composed of a field-effect-transistor (FET) and an electret. The difference between the conventional electret capacitive microphones and the proposed microphone may be the transduction mechanism: The change in the position of an electret causes the change in electric field on the gate of FET. Compared with capacitive transduction, the resistive channel of FET can be designed to have low sensor impedance, and subsequently have low impedance at low frequency. To make experimental specimen, FET onto membrane and electret was fabricated with conventional metal-oxide-semiconductor fabrication process and micromachining process, respectively. The FET membrane chip and the electret chip were assembled. Simple current to voltage converter was applied as a pre-amplifier. Its feasibility to apply low frequency acoustic sensor will be proved by simulation and experimental results.

2:15

3pEA6. The piezoelectric micromachined ultrasonic transducer arrays as a high-intensity sound generator: An experimental analysis. Yub Je (Dept. of Mech. Eng., POSTECH, San 31, Hyoja-dong, Nam-gu, Pohang, Gyungbuk, Republic of Korea, effortjy@postech.ac.kr), Haksue Lee (Agency for Defense Development, Jinhae, Gyungnam 645-016, Republic of Korea), and Wonkyu Moon (POSTECH, Pohang, Gyungbuk, Republic of Korea)

High-intensity sound generation in air with high efficiency is usually difficult, since the acoustic impedance of air is generally much smaller than that of the transducers. In an earlier study by [H. Lee *et al*., J. Acoust. Soc. Am. **125**, 1879-1893 (2009)], the potential for high-efficiency sound generation of a piezoelectric micromachined ultrasonic transducer (pMUT) was examined theoretically and experimentally. A thin-film transducer such as pMUT can generate sound with improved mechano-acoustic efficiency since the internal mechanical impedance of the thin-film structure can be reduced. The highly efficient characteristics of a single pMUT, however, cannot easily be used to yield high-intensity sound from pMUT array, due to variations between units in membrane size, driving voltage, and acoustic loading. In this work, a pMUT array consisting of efficient PZT uni-morph elements was designed and fabricated. The unit-to-unit variations of the pMUT array are analyzed by measuring velocity amplitude and phase of each unit transducer. The result shows that the radiated sound from pMUT array is greatly reduced due to the unit-to-unit variations of the pMUT. To eliminate the problems, uniformity control of fabrication process, optimal design of electrode pattern, and alignment method of an array pattern are considered for high-intensity sound generation.

2:30

3pEA7. Digital beam-steering of parametric array sound beams using piezoelectric micromachined ultrasonic transducers. Yub Je (Dept. of Mech. Eng., San 31, Hyoja-dong, Nam-gu, Pohang, Gyungbuk, 790-784, Republic of Korea, effortjy@postech.ac.kr), Haksue Lee (Agency for Defense Development, Jinhae, Gyungnam 645-016, Republic of Korea), and Wonkyu Moon (Gyungbuk 790-784, Republic of Korea)

In a previous study [H. Lee *et al*., J. Acoust. Soc. Am. **125**, 1879–1893 (2009)], a parametric array (PA) transducer in air using micromachined ultrasonic transducers (MUT) was examined as an ultrasonic ranging sensor. It was shown that the piezoelectric micromachined ultrasonic transducer (pMUT) may be effective for PA ultrasonic ranging sensors. Since the pMUT are usually used in their array, PA source using pMUT can steer its directional sound beam by manipulating phases of radiator in the array. The beam steering capability of the PA source can provide very useful function for parametric loudspeakers as well as directional ultrasonic range sensors. In this work, 16-channel pMUT array for digital beam-steering of parametric array was designed and fabricated. Each channel of the pMUT array consisted of 9 PZT unimorph elements. The two resonance frequency-type pMUT units are designed for generating two primary beams. Beam-steering of each primary beam was controlled separately by applying complex weighting to each channel. An acoustic experiment verified the PA beam-steering of the pMUT. The result shows that the proposed pMUT is suitable for digital beam-steering of parametric array and that it is also useful for the parametric loudspeaker and the PA ultrasonic range sensor.

2:45

3pEA8. The performance enhancement of the hydrophone based on the piezoelectric gate of field effect transistor for low frequency application. Min Sung, Kumjae Shin (PIRO 416, Pohang Univ. of Sci. and Technol., San31, HyoJa-dong, Nam-gu, Pohang-city, KyungBuk, South Korea, smmath2@postech.ac.kr), and Wonkyu Moon (PIRO 405, Pohang Univ. of Sci. and Technol., Pohang-city, KyungBuk, South Korea)

A hydrophone based on the transduction using the electric field of piezoelectric ceramic directly applied on the gate of field effect transistor was proposed, and the feasibility was shown in the year of 2010. Although the feasibility was proved with proper working condition, the fabricated device had some problems on the stability, noise, and sensitivity. The electric field generated from the piezoelectric ceramic in thickness mode, which receives the acoustic pressure from the pressure amplifying head mass structure, modulates the channel current of field effect transistor. The parametric analysis for the transduction was done for the performance enhancement. To realize stable and highly sensitive modulation, the field effect transistor was designed and fabricated with improved gate insulation layer. The passivation for the field effect transistor was also performed to protect the device from external factors and reduce the noise. The sensitivity evaluation setup using the free field calibration method was modified for more accurate measurement. The evaluation results for the improved device are to be presented. [Research supported by MRCnd.]

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Carl J. Rosenberg, Chair
Acentech, 33 Moulton St., Cambridge, MA 02138

Chair's Introduction—2:10

Invited Paper

2:15

3pED1. Teaching architectural acoustics to architecture, architectural engineering, and music students. Robert C. Coffeen (School of Architecture, Design & Planning, Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045, coffeen@ku.edu)

A rewarding career in architectural acoustics has provided the opportunity to teach as a faculty member of the School of Architecture, Design and Planning at the University of Kansas. This presentation will discuss methods and techniques for teaching the basics of architectural acoustics to architecture students so that they might apply these basics to their building designs, for teaching acoustic basics to music students so that they will appreciate and properly use venues for music performance and rehearsal, and for teaching architecture and architectural engineering students interested in building acoustics and who desire to have a career in professional acoustical consulting. These methods and techniques are built on current acoustic knowledge, measurements, and analysis-measurement computer software; on 35 yr of acoustical consulting experience; on opportunities for assisting with architecture studios, for teaching established courses and portions of courses, and for creating new courses with such opportunities provided by the architecture and the architectural engineering faculty and administration; by the Acoustical Society of America with teaching opportunities such as provided by the annual student design competitions; and by many excellent architecture and architectural engineering students. Specific methods, techniques, presentations, course syllabi, etc., will be discussed.

Session 3pIDa

Interdisciplinary: Hot Topics in Acoustics

Peter Gerstoft, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Invited Papers

1:00

3pIDa1. Hands-on acoustics for public school students. Uwe J. Hansen (Dept. of Chemistry and Phys., Indiana State Univ. Terre Haute, IN 47809) and Wendy Adams (Univ. of Northern Colorado, Greely, CO 80639)

Since the appointment of an Education Coordinator, the hands-on equipment purchased by ASA, as well as material donated by various Science Education companies, has been sorted and prepared for safe and easy shipping. It is ready for continuing use at the ASA semiannual meetings and within limits will also be accessible for use by ASA members in connection with various events at locations throughout the country. The general hands-on philosophy and a number of the experiments included in the material will be discussed and demonstrated. A detailed description of the individual experiments, as presented in a student hand-out, will also be distributed. The "Acoustics Activity Kit for Teachers" will also be demonstrated and discussed.

1:20

3pIDa2. Hot topics in physical acoustics. Veerle Keppens (Univ. of Tennessee, Materials Sci. and Eng., Knoxville, TN 37996)

Understanding the propagation and interaction of acoustic waves in "unconventional" systems is one of the main focus areas of physical acoustics, where unconventional may indicate that the system is far from thermodynamic equilibrium, poorly understood, newly

discovered, or a combination thereof. In this talk, I will attempt to give a few specific examples of recent contributions from the field of physical acoustics to a better understanding of "hot" materials such as Pu and pnictide superconductors. Additional examples from current efforts in physical acoustics will be used to illustrate the breadth of the field and show how physical acoustics is closely tied to many other technical areas within the Society.

1:40

3pIDa3. Hot topics in noise: Wind turbines. Kenneth Kaliski (Resource Systems Group, Inc., 55 Railroad Row, White River Junction, VT 05001, kkaliski@rsginc.com)

Since 2000, the installed capacity of wind energy in the U.S. increased seventeen-fold, with the greatest increases in recent years. Today, we commonly find terrestrial wind farms with individual towers of 80–100 m and rotor diameters up to 112 m. The number of wind farms is projected to increase, as many states are adopting renewable energy portfolio standards in an effort to help combat climate change. As wind farms grow, there are inevitably more potential conflicts related to the impacts of the noise generated by the turbines. Noise has taken center stage in both proceedings for proposed wind farms and complaints from some operating wind farms. This talk will review the major issues regarding wind turbine noise, including sound emissions, sound propagation and modeling, sound monitoring, human perception, and regulation. Areas of controversy will be explored, including infrasound and low frequency sound, amplitude modulation, and health impacts.

WEDNESDAY AFTERNOON, 2 NOVEMBER 2011 ROYAL PALM 1/2, 1:00 TO 3:15 P.M.

Session 3pIDb

Interdisciplinary: Demystifying Standards

Paul D. Schomer, Cochair
Schomer and Associates, Inc., 2117 Robert Dr., Champaign, IL 61821

Susan B. Blaeser, Cochair
Acoustical Society of America Standards Secretariat, 35 Pinelawn Rd., Ste. 114E, Melville, NY 11747

Invited Papers

1:00

3pIDb1. Standards 101. Paul Schomer (Schomer and Assoc. Inc., 2117 Robert Dr. Champaign, IL 61821, schomer@SchomerAndAssoc..com)

The Acoustical Society of America (ASA) was founded in 1929 and the standards program of the Acoustical Society began in 1930. It followed the establishment of meetings and the journal. Currently, only meetings attract greater participation than does the standards program. Although standards have been an integral part of the ASA for 81 yr, it still remains a mystery to many members. This paper introduces the special session on standards and describes the basic organization and operation of the standards program. It is followed by case studies highlighting the standards process and the reasons organizations of different types and sizes find it worthwhile to participate. While this session primarily has a U.S. perspective, it also includes a paper presenting a non-U.S. perspective.

1:15

3pIDb2. Modern tools for improving the development of acoustical standards. Christopher J. Struck (CJS Labs., 57 States St., San Francisco, CA 94114, cjs@cjs-labs.com)

The development of acoustical standards has in the past largely followed on the trailing edge of technology. Given the progressively more rapid development of new technology, it is critical that new standards be developed when needed in the timeliest manner possible. Furthermore, participants in working groups are volunteers with limited time available for this important work. Budgetary constraints may also limit travel to in-person meetings for many persons otherwise interested in participating and whose practical experience is essential to the process. On-line meeting and collaboration tools enable shorter, more effective, and more frequent meetings to move draft standards more quickly to a ballot-ready document. Documents can be edited collaboratively in real time using standard mark-up tools for immediate feedback from participants. This also enables participation across time zones. The use of a password protected "cloud network" ftp site for working group documents (e.g., drafts, reference documents, meeting minutes, etc.) eliminates unnecessary e-mail traffic with large attachments and enables participants to access documents at any convenient time. These tools will be described and demonstrated to show the contemporary standards development process in action.

1:30

3pIDb3. An equipment manufacturer's perspective of acoustical standards. Stephen J. Lind (Ingersoll Rand, Trane Bldg. 12-1, 3600 Pammel Creek Rd., La Crosse, WI 54601)

As an HVAC manufacturer, it is our goal to create safe, comfortable, and efficient environments (i.e., classrooms, health care facilities, offices, etc.). Standards are useful in gaining agreement regarding the appropriate sound levels for these spaces and establishing how to verify that these levels are met. Regular interaction and participation in writing the standards helps to ensure that all are aware of the requirements and it also allows participants to provide input on what is practical. Participating in the process also helps to prevent adversarial relations between the parties involved in the design and use of the equipment. Standards also provide the technical basis for which manufacturers and customers can make a valid comparison of different products. The information provided is useful in the design process. Sound power of the equipment is a commonly used metric. Standardizing sound power measurement methods reduces the customer's risk by making sure the published sound levels are representative of the equipment as it is intended and gives a defined expectation of repeatability. Thus, participating in the standards process, we help to ensure that the measurement methods are practical for the intended equipment.

1:45

3pIDb4. The standards process from the perspective of a government scientist. Arnold G. Konheim (U.S. Dept. of Transportation, 1200 New Jersey Ave., SE, Washington, DC 20590)

Through its operating administrations, the U.S. Department of Transportation issues regulations covering the safety, security, health, and environmental protection of all modes of transportation. Many of these regulations incorporate consensus standards as required under the National Technology Transfer and Advancement Act of 1995, which directs federal agencies to generally use standards developed by voluntary consensus bodies. Particular examples of the role of consensus standards in DOT regulations are discussed in this presentation.

2:00

3pIDb5. Taking ASA/ANSI Standards to ISO. Michael Bahtiarian (Noise Control Eng., Inc., 799 Middlesex Turnpike Billerica, MA 01821, mikeb@noise-control.com)

The author served as the chairman of the working group that completed the first voluntary consensus standard for measurement of underwater noise from ships (ANSI/ASA S12.64-2009/Part 1, Quantities and Procedures for Description and Measurement of Underwater Sound from Ships, Part 1; General Requirements). Within the last year, an effort to bring this standard to the international arena has started. The S12.64 standard is an entirely new document without any prior written standards to use as a template, however, the methodology was previously used by the U.S. Navy and NOAA. This session will address the process of organizing and developing a new ASA/ANSI standard and then taking such a standard to International Standards Organization (ISO). It will address organizational challenges of working with experts from industry, government, and academia, both from the United States. and overseas. It will cover issues related to planning and executing physical and web enabled meetings. Lastly, it will address the similarities and differences of the standards development process under ISO.

3p WED. PM

2:15

3pIDb6. Tower of Babel, or why bother about international standards? Osten Axelsson (Passvaegen 30, SE-14753 Tumba, Sweden)

While a vast nation like the USA has the capacity to be self-sufficient, many countries lack this privilege. Take Sweden as an example, a small country in northern Europe with a population of 9 million. Swedes are proud to be international. And they should, because how could a nation, which since the days of the Vikings has depended on international trade, sustain itself without a global economy. International standards support the development within this global economy, just like English as business language facilitates global collaboration. Imagine humanity without these common frames of reference.

2:30

3pIDb7. International standards development and the U.S. Technical Advisory Group process. Susan B. Blaeser (Acoust. Society of America Standards Secretariat, 35 Pinelawn Rd., Ste. 114E, Melville, NY 11747, sblaeser@aip.org)

As industry becomes increasingly global the importance of international standards increases, too. How can U.S. companies, government agencies, and other organizations ensure that their voices are heard and their interests are protected? The U.S. Technical Advisory Group (U.S. TAG) provides the only avenue for U.S. stakeholders to provide input to technical committees in the International Organization for Standardization and the International Electrotechnical Commission. The Acoustic Society of America currently administers nine of these U.S. TAGs. This paper discusses TAG membership and participation in international standards development.

2:45–3:15 Panel Discussion

Session 3pNS

Noise, Psychological and Physiological Acoustics, and Speech Communication: New Advances in Bone and/or Tissue Conduction of Noise

Richard L. McKinley, Chair
Air Force Research Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433-7901

Chair's Introduction—1:00

Invited Papers

1:05

3pNS1. Investigation of the relationship between the vibration patterns of the skull and bone-conducted sound. Margaret Wismer (Dept. of Phys., Bloomsburg Univ., Bloomsburg, PA 17815, mwismer@bloomu.edu) and William O'Brien (Univ. of Illinois Urbana-Champaign)

Analytic and numerical models are used to evaluate acoustic energy converted into bone conducted sound that reaches the middle and inner ear. Sound impinging upon the skull, from an outside source, is channeled into surface waves on the outside of the skull. These waves have components in both the solid bone and the background medium of air, travel at much slower speeds than longitudinal sound speed for bone and have wavelengths smaller than those of an acoustic signal existing solely in bone or in the cranial cavity. Frequencies, at which an integer number plus one half of surface wave wavelengths fit around the circumference of the sphere, cause resonant vibrations. The added one half is due to phase matching surface waves at the poles. The relatively slow sound and short wavelengths mean resonance frequencies for the skull are lower than if the acoustic energy were confined to standing waves within the cranial cavity wherein wavelengths are longer. At acoustic frequencies skull vibrations are analogous to lowest order flexural modes of a fluid-loaded elastic sphere. Numerical models suggest bone's conducted sounds are confined to the skull surface with wave speeds much lower than speed of sound for boney or soft tissue.

1:30

3pNS2. Bone and/or tissue conducted noise: Implications for advanced hearing protection. Hilary L. Gallagher (USAF/AFRL 711 HPW/RHCB, 2610 Seventh St. Bldg. 441, WPAFB, OH 45433), Melissa A. Theis (Oak Ridge Inst. for Sci. and Education, TN 37831), and Richard L McKinley (USAF/AFRL 711 HPW/RHCB, OH 45433)

Military personnel working in various high noise environments may be exposed to continuous noise levels reaching 150 dB. At those levels, bone and tissue conduction pathways become the dominate pathway for sound transmission to the cochlea. Accordingly, hearing protection devices designed to attenuate noise transmitted via air conduction pathways may not be sufficient for meeting hearing conservation requirements. The goal of adequately protecting personnel in these types of environments requires a better understanding of the bone and/or tissue conduction flanking pathways, the susceptibility of the cochlea to bone and tissue conducted energy, and the accompanying mitigation strategies. This experiment investigated the linearity of air conducted noise to the transmission of bone, tissue, and bone/tissue conducted noise. Specially designed bone and tissue conduction drivers, which primarily isolate and excite the respective desired pathways, were used to conduct a loudness matching study. Preliminary findings from this study will be discussed, as well as the implications for development of more effective hearing protection and potential needs for creating new safety requirements based on bone and tissue conducted stimulation. [Work sponsored by AFOSR.]

1:55

3pNS3. Protecting beyond the bone-conduction limit. Anthony J. Dietz, William E. Audette, Jed C. Wilbur, and Christian H. Passow (Creare Inc., P.O. Box 71, Hanover, NH 03755, ajd@creare.com)

Workers operating in extreme noise may need levels of hearing protection that are beyond that possible with double hearing protection comprising earplugs and circumaural hearing protectors. In such noise fields, sound conducted along bone-conduction transmission paths that bypass the ear canal can be sufficient to cause hearing damage. To provide sufficient protection beyond this bone-conduction limit, hearing protectors must attenuate sound that is transmitted to the cochlea via bypass mechanisms. The design of a passive hearing protection helmet that provides protection beyond the bone-conduction limit is described here. The helmet was developed for Navy aircraft carrier deck crews who are exposed to extreme sound levels of 150 dB during aircraft launch operations. The helmet design was based on extensive measurements with human subjects and with a human head simulator built to measure bone-conducted sound. These measurements demonstrated that a helmet shell fitted with an edge seal that created an acoustic seal between the shell and the head was effective in attenuating bone-conducted sound. Measured attenuation data are presented in addition to fit and performance data from tests in the laboratory and the field. Lessons learned from this development effort are also discussed. [Work sponsored by the U.S. Navy].

2:20

3pNS4. Ambiguity in phoneme recognition when using a bone conduction microphone. Karl Buck and Vronique Zimpfer (APC, French German Res. Inst., 5 r Gal. Cassgnou, BP 70034, FR68301 St. Louis, France, karl.buck@isl.eu)

Bone conduction microphones are often promoted as good means for recoding voice signals in very noisy and/or harsh environments. However, it is reported in the literature that the speech intelligibility when using contact microphones is, in all noise conditions, worse than when using a boom microphone. Therefore, long term spectra of speech signals simultaneously recorded with bone and air microphones have been analyzed. This analysis shows, for all measured subjects, that the module of the transfer function between the signal recorded at the air-microphone and, because of this, at the contact microphone increases toward higher frequencies. The results also show large interindividual differences. In order to get more perception related information, a corpus of French vowels has been recorded simultaneously with air and contact microphone. This corpus has been used for a listening test aiming to show confusion between different vowels. The main confusions are for vowels having the same frequency of the first formant (e.g., [i][y][u]) which then are perceived as the central vowel. These confusions could explain the systematically lower intelligibility reported for speech when recorded with contact microphones. Planned experiments using plosive and fricative consonants should give more information about transfer mechanisms of bone conducted voice.

Contributed Papers

2:45

3pNS5. A simplified axi-symmetric finite element model of the human outer ear to determine the earplug induced auditory occlusion effect. Martin K. Brummund (Dept. of Mech. Eng., ETS, 1100 Rue Notre Dame O., Montreal, PQ H3C1K3, Canada, martin.brummund.1@ens.etsmtl.ca), Franck Sgard (IRSST, Montreal, PQ, H3A3C2, Canada), Yvan Petit, and Frédéric Laville (ETS, Montreal, PQ, H3C1K3, Canada)

Earplugs are a frequently used short-term solution for hearing conservation in the workplace environment. Due to limited auditory comfort, however, workers often only wear them for short periods of time and become prone to hearing loss. An important source of discomfort is the auditory occlusion effect, which expresses itself through the distortion of the wearer's voice and the amplification of physiological noises upon earplug insertion. Simplified numerical modeling can help to better assess and design earplugs, because it requires few system resources and is simpler in terms of numerical and experimental implementation than an equivalent complex model. This work describes a novel coupled linear elasto-acoustic two dimensional finite element (FE) model of the human outer ear. The model comprises the auditory canal as well as the bony, cartilaginous, and skin tissues whose material parameters were approximated using literature findings. The outlined model can compute the transfer functions between the sound pressure levels at the eardrum and a structure-borne excitation for both an unoccluded ear and an ear occluded by a molded earplug. Simulated occlusion effects are examined as a function of excitation, earplug, and insertion depth. Predicted model results are compared to literature findings and to findings obtained from an equivalent three dimensional FE-model.

3:00

3pNS6. Integrating speech enhancement with subband active noise control to improve communication in hearing protectors. Eric R. Bernstein, Anthony J. Brammer, Gongqiang Yu, Martin G. Cherniack, and Donald R. Peterson (Ergonomics Technol. Ctr., Univ. of Connecticut Health Ctr., 263 Farmington Ave., Farmington, CT 06030)

Many workers refuse to wear hearing protection devices (HPDs) because they would rather accept the health risks than sacrifice the ability to communicate with coworkers. Integrating active noise reduction (ANR) techniques with speech enhancement algorithms could solve these limitations of modern electronic HPDs. An adaptive delay feedforward subband structure has been implemented by forming parallel signal filtering and filter update paths for each frequency band. Subband ANR provides additional attenuation of environmental noise beyond that of passive HPDs within the lower frequencies associated with speech communication. The subband structure can also provide valuable information regarding the spectral content of the environmental noise that can be exploited to determine where additional power is needed in a communication channel to improve intelligibility. The system, initially developed in simulation, maintains a specified signal-to-noise ratio in each subband while simultaneously limiting the power added by the communication channel to below that associated with hearing damage. Speech Transmission Index models have been used to validate the improvements in speech intelligibility over typical passive and ANR communication headset designs. Preliminary results for a circumaural HPD constructed to implement the concept will be presented for comparison with the simulation. [Work supported by NIOSH (R01 OH008669).]

Session 3pSC

Speech Communication: Speech Rhythm in Production, Perception, and Acquisition I

Amalia Arvaniti, Chair
Dept. of Linguistics, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108

Chair's Introduction—1:00

Invited Papers

1:05

3pSC1. Speech rhythm analysis with empirical mode decomposition. Sam Tilsen (Dept. of Linguist., Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853)

This paper presents a new approach to characterizing speech rhythm, based upon empirical mode decomposition (EMD) [Huang *et al*., Proc. R. Soc. London Ser. A **454**, 903–995 (1998)]. Intrinsic mode functions (IMFs) are obtained from EMD of a vocalic energy envelope of speech, which is a smoothly varying signal, representing primary vocalic resonance energy. EMD uses an iterative sifting process to decompose the signal into IMFs with zero mean and zero-crossings between extrema. Investigations of IMFs revealed that the last two IMFs appear to capture foot- and syllable-timescale oscillations in the envelope, respectively. EMD was applied to a seven-language corpus of speech previously used to compare languages using interval-based rhythm metrics [Ramus *et al*., Cognition **73**, 265–292 (1999)]. The ratio of signal power in the foot- and syllable-associated IMFs was used as metric of the relative influence of foot-based timing on speech. The performance of the IMF power ratio metric in distinguishing languages is comparable to analyses based on interval metrics. The method is argued to have the potential for broad applicability in speech research.

1:25

3pSC2. Comparing envelope- and interval-based rhythm metrics. Amalia Arvaniti (Dept. of Linguist., UC San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, aarvaniti@ucsd.edu) and Sam Tilsen (Dept. of Linguist., Cornell Univ., 203 Morrill Hall, 159 Central Ave., Ithaca, NY 14853)

Numerous metrics have been developed in the attempt to characterize cross-linguistic differences in speech rhythm, particularly with regard to the rhythm class hypothesis, which holds that languages differ in whether they privilege regularity in timing of stress, syllables, or moras. This paper tests for consistency in the performance of different types of rhythm metrics, using speech corpora of English, German, Greek, Italian, Korean, and Spanish obtained from eight speakers of each language with three elicitation methods: read sentences, read running text, and spontaneous speech. Rhythm metrics tested were: interval-based metrics derived from durations of consonantal and vocalic intervals, low-frequency spectral analysis of the vocalic energy envelope, and a recently developed metric based upon the power ratio of foot and syllable intrinsic mode functions obtained from empirical mode decomposition of the vocalic envelope. For all languages, the metrics indicate that spontaneous speech exhibits more stress-timing like characteristics than read speech, having higher interval variability and more dominant stress-timescale periodicity in the envelope. Cross-linguistic differences emerged in some cases, but these were not entirely consistent across metrics and were affected by the elicitation method. Overall the data suggest that the elicitation effects (i.e., read versus spontaneous speech) are larger than differences between languages.

1:45

3pSC3. Speech cycling in Korean. Younah Chung (Dept. of Linguist., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0108)

This paper reports the results of a cycling experiment in Korean, a paradigm in which speakers produce short phrases in time with a metronome. It was hypothesized that in Korean the onsets of accentual phrases would act as beats in this task, playing the same part as stress plays in English; therefore, the accentual phase onsets would remain in the same phase within a cycle independently of their composition. Speakers read Korean sentences with three accentual phrases that had the same number of syllables or differed so one of the accentual phrases had twice as many syllables as the others; the composition of the syllables also varied between CV and CVC. The results so far suggest that speakers keep the accentual phrases in phase although the variation of syllable count and composition also affected phasing. This provides evidence that cycling is a viable task in Korean and supports our hypothesis about the role of the accentual phrase in Korean rhythm. Finally, the relative importance of the syllable cycle supports a view of rhythm that does not rest on the timing of one prosodic constituent, such as the accentual phrase, but on the relative salience of different levels of prosodic structure.

2:05

3pSC4. Speech rhythm in Akan/Twi: A preliminary experiment. Jonathan C. Anderson (Dept. of Linguist., Indiana Univ., Memorial Hall 322, 1021 E. Third St., Bloomington, IN 47405, andersjc@indiana.edu)

This study explores the rhythmic timing patterns of Akan/Twi, a West African tone language thought to be syllable-timed, using the Speech Cycling Task (Cummins & Port, 1998; Port, 2003, 2007; Tajima & Port, 2003). How rhythm appears in tone languages without

stress/accent is poorly understood. In the Speech Cycling Task, speakers were expected to place prominent elements within rhythmic modes, where specific syllables are pronounced more frequently in certain temporal regions. To uncover which syllables are considered rhythmically prominent, a previous tapping experiment (Darwin & Donovan, 1980; Donovan & Darwin, 1979; Purvis, 2009) was used in which subjects displayed entrainment between tapped beats and specific syllables (akin to stress-timing), rather than beat entrainment with all syllables. Prominent elements should also be resistant to temporal displacement when syllables are inserted between them and temporal compensation should occur when syllables can be deleted. The data included 20 phrases ranging from 4 to 8 syllables in length and four tone melodies (H, L, HL, and LH) repeated at varying rates. Results suggest subjects prefer certain rhythmic modes and that displacement and compensation occur such that rhythmic patterns change. Implications for the stress-timing/syllable-timing dichotomy and how tonal melodies affect rhythmic patterns are discussed.

2:25

3pSC5. Tone-based macro-rhythm from the perspective of prosodic typology. Sun-Ah Jun (Dept. of Linguist., UCLA, 405 Hilgard Ave. Los Angeles, CA 90095-1543, jun@humnet.ucla.edu)

In the autosegmental-metrical model of intonational phonology [Pierrehumbert, 1980, Ladd, 1996/2008], prosody is defined in terms of the prosodic structure of an utterance and the prominence relations within the structure. Jun (2005) proposed a model of prosodic typology based on the types of prominence-marking and rhythmic/prosodic units. Languages were categorized as Head-prominence when the head of a prosodic unit such as stress is marked prominent (e.g., English, Spanish, Greek), or Edge-prominence when the edge of a prosodic unit such as an accentual phrase (AP) is marked prominent (e.g., Korean), or Head/Edge-prominence when both the head and the edge are marked prominent (e.g., French, Bengali). The rhythmic/prosodic units covered both micro-rhythm (regularity due to the traditional rhythm category, e.g., stress-timed) and macro-rhythm (regularity due to a tonally defined prosodic unit, e.g., AP). Macro-rhythm was proposed to describe the rhythmic nature of a language where stress is not easily perceptible. In this talk, I will show that macro-rhythm is also crucial in describing sub-groups of Head-prominence languages as well as capturing the relationship between the complexity of tonal category and the types of prominence-marking across languages. Combining the prominence types and the f0-based macro-rhythm provides a better way to establish prosodic typology.

2:45–3:00 Panel-Discussion

Plenary Session and Awards Ceremony

Mardi C. Hastings, President
Acoustical Society of America

Presentation of Certificates to New Fellows

Keith A. Gillis
Mark Hasegawa-Johnson
Veerle M. Keppens
Masao Kimura
Michael J. Owren
Elizabeth A. Strickland
Alexander Sutin
Zhaoyan Zhang

Presentation of Science Writing Awards

Science Writing Award for Professionals in Acoustics
Diana Deutsch, for "Speaking in Tone" published in *Scientific American Mind*, July 2010

Science Writing Award in Acoustics for Journalists
Christopher Bauer, Lindsay Kelliher, Amy Miller, Linda Peckham,
Paul Rogers for their Internet segment
"QUEST Lab: Speed of Sound" aired April 6, 2010

Announcement of the 2011 Munk Award granted jointly by
The Oceanography Society, the Office of Naval Research,
and the Office of the Oceanographer of the Navy

Presentation of Acoustical Society Awards

Rossing Prize in Acoustics Education to Robert C. Coffeen

Distinguished Service Citation to Uwe J. Hansen

Distinguished Service Citation to Richard Stern

Silver Medal in Signal Processing in Acoustics to Theodore G. Birdsall

Trent-Crede Medal to Peter R. Stepanishen

Wallace Clement Sabine Award to J. Christopher Jaffe

Session 3eED

Education in Acoustics and Committee on Women in Acoustics: Listen Up and Get Involved

Marcia J. Isakson, Cochair
Applied Research Lab., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78713

Tracianne B. Neilsen, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N283 Eyring Science Center, Provo, UT 84602

Invited Papers

3eED1. Who is an Acoustician? Wendy K. Adams (Dept. of Phys., Univ. of Northern Colorado, CB 127, Greeley, CO 80639)

The "traveling road show" originally put together by ASAs own Uwe Hansen includes a range of hands-on demonstrations of physical phenomena such as standing waves & resonance, energy carried by sound, and spectrum analysis just to name a few. This show is presented to Girl Scouts with the goal of exciting young ladies about science and letting them interact with female scientists. To further this goal, I propose creating a station that describes careers of acousticians in the various technical areas to educate the young ladies about what different acousticians do in their day to day activities. Career profiles that were previously created for the exploresound.org website will be used along with information presented in a fun, straightforward, graphical, easy to read poster presentation.

3eED2. The sound of resonance. Katherine Hart and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602)

A basic understanding of the production of sound waves can be obtained by studying standing waves and resonance frequencies. A vibrating string will settle into a steady, standing wave pattern when it is driven at one of its resonance frequencies. The resonance frequencies depend on the string's length and the tension applied to the string. Similarly the resonance frequencies of tubes are determined by the tube's length and whether the ends are open or closed. In a series of hands-on demos, we will explore the factors that influence the creation of standing waves by exciting the resonances of strings, tubes, rods, a metal plate, slinky, and a wine glass. These simple models provide insight into how musical instruments produce sound.

3eED3. Do you hear what I hear? Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602)

Have you ever wondered if the person next to you hears the same thing you hear? It depends. The response of our ears varies significantly with age because a person's cumulative exposure to noise lessens their ears' ability to respond to sounds. This common form of hearing loss is called presbycusis and is characterized by a significant degradation in the high frequency content of the perceived sound. A series of auditory demonstrations distributed by the Acoustical Testing Center at NASA Glenn Research Center lets you hear what a person with progressively worse presbycusis hears.

3eED4. The influence of noise on speech intelligibility. Hillary Jones and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602)

When you are listening to speech, a lecture, or a performance, the sound waves travel through the room to reach you. Along the way, the speech sounds interfere with the other sounds in the room such that the sound reaching your ears is often corrupted by noise. The addition of noise decreases the intelligibility of the speech. A series of auditory demonstrations distributed by the Acoustical Testing Center at NASA Glenn Research Center let you hear how the speech intelligibility decreases as the amount of external and internal noise in a lecture hall, classroom, or workplace increases.

3eED5. Scripps classroom connection NSF Graduate STEM Fellows in K-12 education program. Brianne Moskovitz (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., Bldg. 4, San Diego, CA 92106, bmoskovi@ucsd.edu), Hubert Staudigel, and Cheryl Peach (Scripps Inst. of Oceanogr., San Diego, CA 92037)

The scripps classroom connection (SCC) graduate STEM fellows in K-12 Education is a 5-yr program funded through NSF's Division of Graduate Education. The goal of this program is to prepare graduate students for professional and scientific careers for the 21st century. In addition to improving communication and presentation skills for the graduate fellows, the program brings authentic science into middle and high school classrooms of the San Diego Unified School District and creates an enduring set of lesson plans and activities based on the interdisciplinary science conducted at Scripps. One such lesson involves the introduction to ocean acoustics. An activity to accompany this lesson shows the difference in comparison of how sound travels in air and how sound travels in water. Students are able to understand that acoustic waves travel differently in different media.

3eED6. Hands-on demonstrations for Project Listen Up: Education outreach part III. Amie E. Nardini, Jacqueline Blackburn, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MA 21402)

Midshipmen will be getting involved in an ASA education outreach effort by presenting or recording a number of acoustical demonstrations geared to promote a hands-on learning experience for middle- and high-school age girl scouts. This is an extension of the demonstration effort and outreach presented at the ASA Meeting held in Baltimore in 2009 and Seattle in 2011. The demos are designed to visualize certain wave effects that will be explained by the Midshipmen "live" or by computer video. The participants will be free to explore, control the apparatus, and make their own scientific discoveries. The hands-on demonstrations will include (1) a ripple tank with two separated periodic point sources or a plane wave source driver for wave studies, (2) an ultrasonic motion sensor for measuring and displaying displacement, particle velocity and acceleration of an oscillating object or a person's motion, (3) tuning forks matched to Helmholtz resonators, and (4) a driven clamped circular plate supporting a column of dry sifted masonry sand (or uncooked rice or grass seed) for studying the resonant frequency versus. granular mass loading.

WEDNESDAY EVENING, 2 NOVEMBER 2011 — SUNRISE, 7:00 TO 9:00 P.M.

An Evening to Remember with the Hutchins Consort

Thomas D. Rossing, Master of Ceremonies
Stanford University, Stanford, CA 94305

The Hutchins Consort plays on the eight-scaled violins of the violin octet designed and built by famed luthier Carleen Hutchins. The instruments are the first successful attempt to create an acoustically balanced set of instruments that can sound truly like violins across the entire range of written music. The Hutchins Consort plays music of the Middle Ages and Renaissance to the music of the modern masters. With original compositions and transcriptions commissioned by the Catgut Acoustical Society for the octet of violins, and new transcriptions by members of the Consort, The Hutchins Consort displays a breadth and depth that few traditional groups match, and a sound that is truly unique.

WEDNESDAY EVENING, 2 NOVEMBER 2011 — 7:30 TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings beginning at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Royal Palm 5/6
Signal Processing in Acoustics	Royal Palm 1/2

Session 4aAA

Architectural Acoustics, Noise, and Committee on Standards: Networking in Soundscapes—Establishing a Worldwide Collaboration I

Gary W. Siebein, Cochair
Dept. of Architecture, Univ. of Florida, 231 Arch, P.O. Box 115702, Gainesville, FL 32611

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square, Ste. 103, Vernon, CT 06066

Chair's Introduction—7:45

Invited Papers

7:50

4aAA1. Progress in soundscape research requires a common agenda. Osten Axelsson (Passvaegen 30, SE-147 53 Tumba, Sweden)

It is commonly believed that progress and success in any field requires competition. This is probably true, but this belief implies that all competitors have a common view on the objectives. There would not be much competition if all parties ran off in opposite directions, striving to achieve different goals. Nor would it lead to much progress. The present session calls for networking and international collaboration in soundscape research. For such collaboration to be successful, it is critical to agree on a common agenda; a mission; an objective. Recent development in soundscape research makes evident that the objective must be practical and applicable. Our minds must be set to implementing soundscape research in practice to avoid exhausting academic debates, which tend to be ends in themselves and do not contribute to progress. Two excellent, recent examples of international collaboration in soundscape research, contributing to progress, are ISO/TC 43/SC 1/WG 54 and the European COST Action TD0804 "Soundscape of European Cities and Landscapes." Both illustrate the need for international and interdisciplinary collaboration among acousticians, architects, and urban planners to accelerate progress in soundscape research. The present paper presents possible topics for a common agenda in soundscape research.

8:10

4aAA2. Languages and conceptualization of soundscapes: A cross-linguistic analysis. Caroline Cance (INCAS3 and CIRMMT, Dr. Nassaulaan 9, Assen 9401HJ, The Netherlands, ccance@gmail.com), Catherine Guastavino (McGill Univ. and CIRMMT Montreal, QC H3A1X1, Canada), and Danièle Dubois (CNRS Univ. Paris 6, Paris, France and INCAS3, Assen, 9401HJ, The Netherlands)

In the past decade, soundscape research has emerged accounting for acoustic phenomena as they are perceived and assessed by humans. In this view, concepts and methodologies from social sciences and humanities are needed to identify the diversity of conceptualizations across time, space, and languages. Specifically, our approach relies on linguistics and psychology in analyzing how people describe their sensory experience (what is being said and how it is being said), in order to identify different conceptualizations conveyed in their discourse. We first investigate the linguistic resources available in different languages with a cross-linguistic survey of free-format verbal descriptions of acoustic phenomena in European languages (e.g., French, English, Dutch, Spanish), extending the pilot investigation by Dubois and Guastavino (2008). Then, coupling this linguistic analysis with cognitive theories on categorization, we can infer a diversity of conceptualizations for the same acoustic phenomena. This approach further allows us to overcome some limitations of current survey design: the use of closed-ended questions confining responses to categories pre-defined by the experimenter, and basic translations not taking into consideration semantic languages specificities. Our results provide a theoretical grounding and methodological guidelines for designing questionnaires for cross-cultural evaluation of soundscapes.

8:30

4aAA3. Impacts on soundscape appreciation by focussing on sources. Andre Fiebig (HEAD acoustcis GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

Based on COST, the European Cooperation in Science and Technology as an instrument supporting cooperation among scientists and researchers across Europe, collaborations among soundscape researchers was and is still encouraged (COST action TD 0804). In this context, researchers from 18 COST countries and 7 partners outside Europe come together and work on terminology, collection and documentation of soundscapes, harmonization of methodologies and indicators, and design of soundscapes. One scientific issue deals with sound source constellations in soundscapes and evoked attention-attracting processes. Humans can easily focus on a certain source suppressing the noise of other sources. This process obviously influences the general appreciation of the whole soundscape. In different tests this phenomenon was investigated and its potential analyzed with respect to soundscape design. The process of international and interdisciplinary cooperation within the COST framework will be shortly described and benefits and limitations discussed. Moreover, different case studies will be presented to show the effect of source attention and its impact on soundscape evaluation.

8:50

4aAA4. Understanding soundscape as a specific environmental experience: Highlighting the importance of context relevance. Itziar Aspuru (Environ. Unit, Tecnalia Res. & Innovation, Parque Tecnológico Bizkaia, Derio, 48106, Spain, itziar.aspuru@tecnalia.com)

A proposed general conceptual model about environmental experience is presented to guide the soundscape studies. This proposal is the output of a review of the literature on soundscape and our experience regarding psychosocial studies in Tecnalia focuses on the relationship between environment and persons or communities. The general model has been structured around five main elements: person (community), place, activity, previous interaction between person and place, and environmental experience. The environmental experience is a holistic experience within that soundscape and is linked to other perceptions, such as landscape, odour, etc. The relevant factors and variables of each main element have been identified to help us to explain human and social holistic experience in relation to place, in general, and soundscape (perception) in particular. The main objective of this paper is to present this model and to discuss it with interested colleagues.

9:10

4aAA5. Defining the scope of soundscape studies and design methods. Gary W. Siebein (School of Architecture Univ. of Florida Gainesville, FL 32611)

One important goal of soundscape studies is to develop methods that architects, urban designers, planners, landscape architects, and others involved with the design of the indoor and outdoor physical environment can be implement as part of their work. As such the role of design is an inherent aspect of soundscape studies. Design of the physical environment, especially its sonic qualities, is similar to the composing and orchestrating of a piece of music. This paper explores the translation of soundscape theories into physical design concepts through a review of the seminal literature of Shafer and Truax and the systems ecology models of Odum and Brown. The use of physical design elements to create sonic niches for specific acoustic events to occur within that are defined in location and time is used as a technique to develop local sonic attributes within soundscapes. Advanced modeling and simulation techniques; long and short term acoustical measurements that reflect the way sounds are heard by people; interviews, focus group discussions, and questionnaires given to participants in the soundscape to explore the ways people understand sounds in their context; and transformations of the data into physical acoustical interventions are important parts of the process.

9:30

4aAA6. Architectures of sound: A new material in socio-spatial formation. Lorenzo Beretta (BIAD Res., Birmingham City Univ., Gosta Green, Corp. St, Birmingham, UK B4 7DX, Lorenzo.Beretta@bcu.ac.uk)

In architecture, sound is only haphazardly used as an accessory to pre-defined, pre-constituted architectural blocks. The paper examines those sonic properties that are able to alter cognition, behavior, and human interaction leading toward the definition of sound as a material. Over the centuries, buildings have changed their constituent materials ranging from ceramic bricks to concrete, from wood to glass, from light to vapor. Sound is not just a floating, ephemeral presence that exists over time but a solid material that defines and changes the space around us. As a result, spaces are not just lived through their solid elements but through their transposition in the mind of each user. It is for this that by acting on the properties of sound, it is possible to redefine the way people navigate and inhabit spaces. Sound should no longer be used as an accessory or content of architectural spaces but, instead, as a material able to give form, volume, and shape to what is generally referred to as architecture—an inhabitable entity of social relevance in the urban context.

9:50

4aAA7. The use of musical composition and spatial analysis to document and design soundscapes. Joshua C Fisher (6879 Pentland Way #63, Fort Myers, FL 33966, jfisher@jocofi.com)

A series of studies were conducted that used augmented environmental noise analysis techniques. Impulse response analysis and digital processing techniques were applied to both previously recorded sound samples and binaural recordings found on a target site. To provide a representation of the character of a given space, recordings of individual sounds on the site were used to construct digitally composed auralizations of the objects and spaces that were representative of the target space over time. By treating the spatial environment and the individual sounds as separate elements, this technique allowed existing soundscape studies to be used as base models for new soundscapes. This technique was implemented in a number of studies in various natural and manmade locations, all of which demonstrated that sound could be used as part of an integrated design process that allows the designer to treat sound as a material rather than as an unwanted substance (noise).

10:10–10:20 Break

10:20

4aAA8. Soundscape study methods of band rehearsal room. Lucky Tsaih (School of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32601, akustx@yahoo.com)

Soundscape study methods were applied to a band rehearsal room to understand the acoustical issues involved in this situation. A taxonomy of sound sources, direct observation to identify the specific sonic events and the multiple source and receiver paths involved in the complex listening and performing tasks that occur during band rehearsals were identified. It was found that students spend almost 1/2 of their rehearsal time involved with verbal instruction and discussion. Interviews with conductors and questionnaires administered to music students in three different band rooms were used to determine what musicians are listening for during rehearsals. Source and

receiver combinations for physical acoustical measurements were located to study the multiple listening tasks identified in the questionnaires. Musicians were constantly trying to “hear each other” for intonation, rhythm, dynamics, articulation, and tone quality during rehearsals. Statistical models linking the qualitative results of the questionnaires with the acoustical measurements and architectural features of the rooms show that the ceiling height, room volume, area of sound diffusing surfaces, low frequency sound level, early reflected sound energy, and reverberation were related to the ability of musicians to hear each other and the detailed attributes of music.

10:40

4aAA9. Application of soundscape method for a worship space. Sang Bum Park (6029 SW 85th St., Gainesville, FL 32608, sbpark04@gmail.com)

Soundscape methods have been used to investigate the sound quality of natural and urban environments and the effects of sonic events on people. Soundscape methods can also be used to study the dynamic listening, speaking, and musical acoustical environment of a worship space. Observations of services in three worship spaces led to identification of a taxonomy of sounds that occur in the rooms as well as the multiple participants involved in producing the sounds and listening to them, i.e., the acoustic community. Five groups including ministers, the choir, music director, the congregation, and sound engineers each with their own physical location in the room, communication paths, and values were identified. Sound sources and receivers for physical acoustical measurements and questionnaires were located for each of the groups. Sound systems were used in each space for various reasons. Analysis of physical acoustical measurements and survey results revealed that different groups had different experiences in the worship spaces when the sound system was used and when it was not used.

11:00

4aAA10. A proposed method of data collection and analysis for soundscape-based noise evaluation. Adam D. Bettcher (Univ. of Florida, Architecture Bldg., P.O. Box 115702, Gainesville, FL 32611)

As an information-rich means of describing and accessing a particular aural environment, soundscape analysis requires additional types of information beyond those used for energy-average and statistical methods of noise evaluation. Managing the flow of data from different types of measurements is crucial in providing a meaningful description of the sonic landscape of a site. The proposed method of data collection and processing produced a means of describing the aural environment with quantitative descriptors from analysis of simultaneously recorded sound level measurements and sound recordings. Sound recordings were analyzed to determine the makeup of a soundscape comprised of individual sounds. The sounds were organized into a taxonomy; and analysis of frequency of occurrence, sound levels, and spectra of each sound, duration of each sound, and the rate of change of each sound were conducted to quantitatively describe the elements of the sonic landscape. This method of data analysis was applied to a soundscape study and produced information that exceeded the abilities of sound-energy measurements alone in providing helpful quantitative descriptors for qualitative soundscape analysis.

11:20

4aAA11. Pursuing cooperative solutions to noise management issues arising in U. S. National Parks. Kurt M. Fristrup (Natural Sound and Night Sky Div. Natl. Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

An emerging paradigm in wildlife conservation holds that ecological knowledge is but one of several dimensions that must be addressed to realize successful outcomes. Human factors—history, culture, economics, and mechanisms for decision and implementation—must be taken into account to devise effective solutions. Addressing these factors demands systematic identification and pursuit of partnerships to synthesize a social tool for conservation. In soundscape management, there has been substantial discussion of metrics for measuring noise exposure and how these measures of noise exposure translate into functional consequences for humans. These challenges, which are substantial, may be the least formidable obstacles to constructive change. The recent history of noise management efforts in the National Park Service illustrate the necessity of cultivating partnerships in many pursuits: education and outreach, research and evaluation, informing policy decisions, and implementation of management plans.

11:40

4aAA12. Soundscape collaboration for science, management, and public outreach at a national historic site. Robert C. Maher (Elec. and Comput. Engr., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717-3780, rob.maher@montana.edu) and Christine Ford (Grant-Kohrs Ranch Natl. Historic Site, MT 59722)

Scientists and engineers involved in soundscape research at national parks and historic sites have the opportunity to collaborate with park management and public outreach professionals. The technical and signal processing considerations of the acoustician can complement the management and regulatory considerations of the park supervisors and the public awareness and outreach efforts of the professional staff and interpretive rangers. The triad of science, policy, and outreach involves all of the key stakeholders in soundscape assessment and evaluation. Examples of collaborative activity at a U.S. National Historic Site are presented.

Session 4aAB

Animal Bioacoustics: Long-Term Acoustic Monitoring of Animals I

Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720

Simone Baumann-Pickering, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Chair's Introduction—8:00

Invited Papers

8:05

4aAB1. Acoustic monitoring of fish movements across multiple spatial and temporal scales. Yannis P. Papastamatiou (Florida Museum of Natural History, Univ. of Florida, 280 Dickinson Hall, Gainesville, FL 32611, ypapastamatiou@gmail.com)

Understanding the movements of fishes is important for understanding ecological interactions, conservation, and fisheries management. One of the key techniques used to quantify fish movements is acoustic telemetry. Fish can either be actively tracked (where the fish is continuously followed) or be remotely monitored using an array of underwater listening stations (passive telemetry). Ultimately, an understanding of fish movements is required over multiple spatial and temporal scales, which requires the use of several tools. The specific tools used will also vary based on the species being studied, which can range from small reef fishes to pelagic sharks. The use of telemetry in fish movement studies will be reviewed and examples of what sort of questions may be answered using these tools will be given. Future advances in both the tools utilized and the analytical techniques used to interpret data will also be discussed.

8:25

4aAB2. Comparing data collection and processing options for terrestrial acoustical monitoring addressing long durations and large spatial scales. Kurt M. Fristrup (Natural Sound and Night Sky Div. Natl. Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

For generations, skilled naturalists have listened attentively to expand the scope of their field searches and surveys. The rapid proliferation of digital audio recorders with large storage capacity and low power consumption offers numerous options to pursue standardized acoustical surveys of several weeks duration across large areas. This presentation will assess a representative sample of available equipment in relation to the parameters determining their suitability for these applications: price, capacity, fidelity, and ancillary features. Enormous data collection capacity is not useful unless there are adequate tools for processing the data to render the necessary results. The features and performance of several software packages will be compared in the context of different classes of potential application. In particular, the merits of highly selective detectors will be discussed in relation to the alternative of more permissive screening for potentially interesting sounds followed by a measurement and clustering or classification process.

Contributed Papers

8:45

4aAB3. Long-term mapping of red grouper sound production on the West Florida Shelf. Carrie C. Wall (College of Marine Sci. Univ. of South Florida, 140 7th Ave. S., St., Petersburg, FL 33701, cwalll@mail.usf.edu), Michael Lindemuth (Ctr. for Ocean Tech., Univ. of South Florida, 830 1st St. S., St. Petersburg, FL, 33701), Peter Simard, and David A. Mann (College of Marine Sci. Univ. of South Florida, 140 7th Ave S., St. Petersburg, FL 33701)

While it is widely known that numerous fish species produce sound, discerning when and where is more challenging. Through the use of autonomous passive acoustic technology, the spatial and temporal patterns of fish sound production, namely red grouper Epinephelus morio, in the eastern Gulf of Mexico were documented. Two methods have been employed off west-central Florida: moored passive acoustic arrays deployed in 2008 and 2009 covering over 16 600 km^2 from the coast to 100 m deep, and autonomous gliders with integrated hydrophones deployed cross-shelf for up to 4 weeks. Over four million acoustic files generated from these methods were analyzed using DSGLab, an open-source database and data analysis system implemented using MATLAB and MYSQL. An automatic detection algorithm was created and implemented in DSGLab to determine the presence of red grouper calls. False detections were removed manually and the results were analyzed to determine diel and seasonal variability of red grouper sound production in addition to identifying the range of red grouper in the eastern Gulf of Mexico. Support was provided by the University of South Florida, Center for Ocean Technology glider staff, and the captains and crew of the R/Vs Weatherbird II, FishHawk, Eugenie Clark, and Allicat, and the M/V Narcosis. This research was funded by NOPP (OCE-0741705) awarded to DM and the USF/USGS Graduate Assistantship awarded to CW.

9:00

4aAB4. Passive acoustic fish location with a 3-D fixed array. Rodney Rountree, Yegor Sinelnikov, and Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

There is a strong need for improved sound source localization software for use by scientist interested in conducting passive acoustic surveys of marine and aquatic habitats where most biological sounds are currently unidentified. Fish sounds are typically low frequency (50–1200 Hz), narrow band, knock trains, short duration grunts, or tones, and can be repeating or irregular. These characteristics together with typically fuzzy signal onset, noisy environmental conditions, and shallow depths often make source localization challenging. We used small 3-D arrays of six hydrophones placed in a fixed-frame square diamond configuration to collect known and unidentified fish sounds for testing from a variety of shallow marine and freshwater habitats. The various methods of sound source localization based on measurements of Differences in Time of Arrival (DTOA) at various hydrophones were investigated, including cross-correlation, first pulse arrival, and phase measurement of tonal components in the fish signal. The most accurate DTOA measurements were obtained using cross-correlation with the phase transform (PHAT) methods. The various algorithms of source localization based on DTOA were considered and MATLAB program for these algorithms were developed. Our goal is to develop publically available software realizing the optimal method of DTOA measurements and source localization.

9:15

4aAB5. Ocean conditions and occurrence of cetacean species near Pt. Sur, California, in 2008–2010. Tetyana Margolina, Alison K. Stimpert, John E. Joseph, Curtis A. Collins, and Christopher W. Miller (Dept. of Oceanogr., Naval Postgrad. School, 1 University Cir., Monterey, CA 93943)

The central California region of the California Current system is characterized by rich marine life and highly variable physical and biological ocean conditions. Here, patterns of marine mammal occurrence near Point Sur are analyzed and linked to seasonal and larger scale variability of the California Current system. A summary of marine mammal vocalizations was created by scanning passive acoustic recordings in the 10–100 kHz frequency band acquired with High-frequency Acoustic Recording Packages (HARPs) deployed at depth of approximately 840 m from August 2008–November 2010. Calls of detected baleen and toothed whales were presented as occurrence time diagrams. To characterize the ocean conditions of the California Current system during this time period, various datasets were used including current meter data collected at Sur Ridge and satellite-derived information on chlorophyll-a concentration (MODIS), temperature (AVHRR), and sea surface height anomalies (AVISO SSH products). Data series of chlorophyll concentrations were used to examine possible changes in the ocean physical and biological state off central California in response to El Niño/Southern Oscillation processes. Correlations between patterns of cetacean occurrence and variability of oceanographic conditions are analyzed and discussed. [Research supported by US Navy CNO(N45).]

9:30

4aAB6. Marine mammal vocalizations and shipping patterns off the California coast near Sur Ridge. John E. Joseph, Alison K. Stimpert, Tetyana Margolina, and Christopher W. Miller (Dept. of Oceanogr., Naval Postgrad. School, 1 Univ. Circle, Monterey, CA 93943, jejoseph@nps.edu)

Studying the effects of anthropogenic noise on marine life is the focus of many ongoing research efforts, and regional studies can provide useful insight into the broader issues. Waters off the central California coast are well known for the rich occurrence of a variety of marine mammal species. The region also contains important ship routes used by vessels transiting between major US west coast ports. Here, we examine the relationship between marine mammal vocalizations collected near Sur Ridge and local shipping patterns determined from automatic identification system (AIS) reports broadcast by ships passing through the region. Passive acoustic recordings of vocalizations were acquired with a moored high-frequency acoustic recording package (HARP) over the frequency bandwidth 10 Hz–100 kHz. Number of vessels within several different radii from the HARP mooring is correlated with presence/absence data of several baleen whale, beaked whale, and dolphin species over 5-min intervals between January 2009 and November 2010. We also examine whether diel patterns of marine mammal distribution are influenced by diel patterns in ship traffic. These data will be useful for establishing mammal-vessel interaction rates in the Monterey Bay National Marine Sanctuary. [Research supported by US Navy CNO(N45)].

Invited Papers

9:45

4aAB7. Acoustic and thermal monitoring of temperate anuran populations. Rafael Marquez, Diego Llusia (Fonoteca Zoológica, Dep Biodiv y Biol Evol, Museo Nal Ciencias Naturales (CSIC). José Gutiérrez Abascal, 2, 28006, Madrid, Spain rmarquez@mncn.csic.es), and Juan Francisco Beltrán (Univ de Sevilla. Av Reina Mercedes, s/n, 41012, Sevilla, Spain)

The results of multi-year acoustic monitoring using automated recording systems (ARS) of the calling activity of two populations per species of five species of anurans from the Iberian peninsula are reported: two species of tree frogs (Hyla) and three species of midwife toads (Alytes), with populations of each species being located at the thermal extremes (coldest vs. hottest) of their distribution. We also report one instance where ARS systems contributed to the recording of a previously undescribed species call, the Moroccan midwife toad. We report methodological procedures such as calculation of the effective area of the recording station, temperature-specific phenological information, and a comparative analysis of environmental predictors of calling activity. Implications for amphibian monitoring and conservation are discussed.

10:05–10:25 Break

10:25

4aAB8. Long-term fish monitoring in the Southern California Bight. Ana Širović (Scripps Inst Oceanogr., UCSD, 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), David A. Demer (Southwest Fisheries Sci. Ctr., NOAA Fisheries, La Jolla, CA 92037), Sean M. Wiggins, and John A. Hildebrand (UCSD, La Jolla, CA 92093-0205)

While over 100 fish families produce sounds during behaviors like spawning, aggression, and feeding, passive-acoustic sampling is not used commonly for long-term fish population monitoring. Fish sounds consist mostly of low-frequency pulses of variable duration, number, and repetition rate, but it is often difficult to identify their sources to species. For example, underwater sounds from marine life have been studied in the Southern California Bight (SCB) for over 60 years, but because the sound producing fishes are difficult to locate and identify visually, their sound production remains poorly understood. The spatial and temporal distributions of the likely fish sounds recorded in SCB were analyzed, but the species producing those sounds are generally unknown. Where the species are known, more information is needed on the seasonal and interannual variations of their sound production if the passive-acoustic records are to be used to estimate their abundances and distributions. We show that sound characteristics and diel sound production patterns for some species, like bocaccio (*Sebastes paucispinis*), have not changed for over four decades. More directed studies are needed on the behavioral context of fish sound production in SCB to facilitate the use of passive-acoustic monitoring for long-term studies of fish population dynamics.

10:45

4aAB9. Monitoring white seabass spawning sounds using a long-term acoustic recording system. Scott A. Aalbers and Chugey A. Sepulveda (Pfleger Inst. of Environ. Res. (PIER), 315 N. Clementine St., Oceanside, CA 92054, Scott@pier.org)

The white seabass (*Atractoscion nobilis*) is an economically important member of the family Sciaenidae that generates a series of distinct low-frequency sounds during spawning. Long-term acoustic recorders (LARS; Loggerhead Instruments) were moored to the seafloor at three sites along the southern California coastline to monitor white seabass sound production. LARS were programmed to record ambient underwater sounds at a sampling rate of 8820 Hz during periods of peak spawning activity (+/− 1 h sunset) from March through July of 2007–2011. White seabass spawning signals were detected at all three sites and verified through the concurrent collection of gravid individuals. Heightened white seabass sound production was documented during May and June, in conjunction with increasing water temperatures and photoperiod. Detection rates were highly variable between adjacent sites and over consecutive seasons, suggesting that spawning activity and site fidelity is influenced by oceanographic conditions. Although additional work is necessary to determine optimal spawning habitats and environmental conditions, this study confirms the utility of a bioacoustic approach to non-invasively identify white seabass spawning periods and locations.

11:00

4aAB10. Nighttime foraging by deep diving echolocating odtonocetes in the Hawaiian Islands. Whitlow W. L. Au, Giacomo Giorli, Michael Richlen, and Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744)

Ecological acoustic recorders (EARs) were deployed in deep waters at five locations around the island of Kauai and one in waters off Ni'ihau in the main Hawaiian island chain. The EARs were moored to the bottom at depths between 400 and 800 m. The data acquisition sampling rate was 80 kHz and acoustic signals were recorded for 30 s every 5 min to conserve battery power and disk space. The acoustic data were analyzed using a suite of software including the M3R (marine mammal monitoring on navy ranges) algorithm, an energy-ratio-mapping algorithm developed at Oregon State University, TRITON software developed at Scripps Institute of Oceanography, and custom MATLAB programs. A variety of deep diving odontocetes, including pilot whales, Risso's dolphins, sperm whales, spinner and pan-tropical spotted dolphins, and beaked whales were detected at all sites. Foraging activity typically began to increase after dusk, peaked in the middle of the night, and began to decrease toward dawn. Between 75 and 87% of biosonar clicks were detected at night. At present, it is not clear why some of the known deep diving species, such as sperm whales and beaked whales, concentrate their foraging efforts at night.

11:15

4aAB11. Long term remote monitoring of cetacean calls using a solar powered autonomous detector. Douglas Gillespie (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St Andrews, KY16 8LB, Scotland), Andrew Maginnis, and Gordon Hastie (SMRU Ltd. New Technol. Ctr., North Haugh, St. Andrews, KY16 9SR, Scotland)

The use of autonomous underwater recording devices is now well established as a method for long term population monitoring. However, the life time of autonomous recorders is still restricted by battery lifetimes and, particularly when monitoring at high frequencies, by available storage space. We present a system for long term monitoring of cetacean populations using a solar powered system in which real time detection algorithms for multiple species can run concurrently on an embedded processing platform. Power consumption is typically below 3 W, with sample rates of up to 500 kHz; the system is therefore suitable for the detection of all known cetacean calls. Data volumes of detected calls are typically below 1 MB a day, meaning that data can be transmitted ashore in near real time using cell or satellite phone networks. Furthermore, communications are bi-directional allowing sampling and detection parameters to be updated remotely. The combination of solar power, real time processing, and data transmission denotes that deployment lifetimes are limited only by the mechanics of the mooring and the need to remove bio-fouling from hydrophones.

11:30

4aAB12. Automatic detection of vocalizations of the frog *Diasporus hylaeformis* in audio recordings. Arturo Camacho (Esc. de CC. de la Comp. e Inf., Univ. de Costa Rica, P.O. Box 2060, San José, Costa Rica, arturo.camacho@ecci.ucr.ac.cr), Adrián García-Rodríguez, and Federico Bolaños (Esc. de Biología, Univ. de Costa Rica)

A method for the automatic detection of calls of the frog *Diasporus hylaeformis* (Eleutherodactylidae) in audio recordings is proposed. The method uses the loudness, timber, and pitch of the vocalizations to identify the calls of the most prevalent individual in a recording. The first step consists in calculating the loudness of the signal to recognize the sections where the focal individual's vocalizations are. The second step consists in using the timber of the signal to recognize vocalizations. Finally, we use two principles we observed in the sounds produced by this species to discriminate between the calls of the most prevalent individual and other calls: individuals tend to vocalize using an almost constant pitch and different individuals use different pitches. Results show that the method is resistant to background noise (including calls of individuals of the same species), microphone-manipulation-induced noise, and human voice, and also that it adapts well to variations in the microphone level produced during the recording.

Session 4aBA

Biomedical Acoustics: Thrombolysis and Microbubble-Mediated Therapies

Azzdine Y. Ammi, Chair
Cardiovascular Medicine, Oregon Health and Science Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239

Chair's Introduction—7:45

Invited Papers

7:50

4aBA1. Low frequency therapeutic ultrasound causes vasodilation and enhanced tissue perfusion. Robert J. Siegel (Heart Inst., Cedars Sinai Medical Ctr., 8700 Beverly Blvd, Los Angeles, CA 90048)

It has been found that ultrasound has a unique effect on arterial and venous dilation as well as tissue perfusion. Our group found that low frequency ultrasound (20 kHz, 0.1 w/cm^2) results in coronary arterial and coronary venous dilation in dogs. The magnitude of vasodilation is similar to that seen with nitroglycerine (NTG) administration but unlike NTG, USD does not lower blood pressure. Our human studies show that USD induces brachial arterial dilation after 1 min with the vasodilatory effect lasting 20 min. In animals we ligated coronary arteries, stopping epicardial coronary flow, resulting in a drop in myocardial tissue perfusion to 70% of normal; myocardial tissue *p*H fell from 7.43 to 7.05. After 60 min of USD, tissue perfusion improved by 20% and *p*H normalized in spite of persistent coronary artery occlusion. The enhanced perfusion effect of ultrasound was eliminated if an inhibitor of nitric oxide synthase was given prior to ultrasound exposure. Conclusions: low frequency ultrasound causes vasodilation and improves tissue perfusion. These effects appear to be mediated at least in part by the USD enhancing tissue release of nitric oxide.

8:10

4aBA2. Bifrequency excitation for extracorporeal ultrasound thrombolysis. Bruno Gilles, Izella Saletes, Mamdouh Dhahbi, Maher Ben Chiekh, Jean-Christophe Béra (INSERM - U1032, Univ. Claude Bernard Lyon 1, 151 Cours A. Thomas, Lyon, 69003, France, bruno.gilles@inserm.fr), and Rares Salomir (Univ. Hospital of Geneva, Geneva, Switzerland)

A bifrequency excitation consisting of two neighboring frequency components can reduce intensities needed to achieve strong inertial cavitation activities. We present *in-vitro* experimental results aiming at testing such a bifrequency excitation for extracorporeal ultrasound thrombolysis. In a first set of experiments, human blood clots were inserted in small tubes filled with saline and placed at the focus of a piezoelectric transducer. The efficiencies of mono- (550 kHZ) and bifrequency (535 and 565 kHz) excitations were compared for (spta) intensities ranging from 50 to 160 W/cm^2, and a passive recording of the cavitation activity was performed during treatment. A modified setup enabled to measure the size distribution of the debris resulting from thrombolysis experiments realized under flow. A comparison of the spatial temperature distribution for each type of excitation was performed in another set of experiments using MR temperature imaging. Under the conditions of the experiments, 80% of thrombolysis was achieved with a monofrequency intensity of 150 W/cm^2, while 80 W/cm^2 were sufficient with a bifrequency excitation. Mean debris size was reduced by the use of a bifrequency excitation, and MR temperature imaging showed that, for a given intensity, the spatial temperature distributions are the same for both types of excitation.

8:30

4aBA3. Assessing thrombolytic efficacy *in vitro*: Clot mass loss versus fibrinogen protein fragment concentration. Stephen R. Perrin Jr. (Biomedical Eng. Program, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3940, Cincinnati, OH 45267, perrinsr@mail.uc.edu), Gail J. Pyne-Geithman (Univ. of Cincinnati, Cincinnati, OH 45267), Nikolas M. Ivancevich (Siemens Medical Solutions, Issaquah, WA 98029), Shauna L. Beiler, Kenneth R. Wagner (Univ. of Cincinnati, Cincinnati, OH 45267), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267)

Ultrasound (US) acts synergistically with recombinant tissue plasminogen activator (rt-PA) to accelerate thrombolysis. A correlation between clot mass loss and production of the fibrin degradation product D-dimer was investigated using an *in vitro* clot thrombolysis model. Fully retracted clots formed from 1.5-ml human whole blood were suspended in plasma at 37°C and treated with rt-PA, or rt-PA, Definity®, and pulsed 120-kHz US for 30 min. Clots in plasma alone served as controls. Thrombolytic efficacy was assessed as percent clot mass loss. Samples from plasma surrounding the clot and from macerated clots were analyzed for D-dimer using an enzyme linked immunosorbent serologic assay. A statistically significant enhancement in clot mass loss was observed for clots exposed to rt-PA, Definity®, and US compared to rt-PA alone. Clots treated with rt-PA exhibited a higher concentration of D-dimer both in the clots and plasma. However, clots treated with rt-PA, Definity®, and US did not exhibit an enhanced level of D-dimer. In future studies, we plan to elucidate the role of erythrocyte liberation in US-enhanced clot mass loss. [This work was supported by NIH RO1 NS047603.]

8:50

4aBA4. Ultrasound-enhanced thrombolysis in porcine clots in a flow system. Azzdine Y. Ammi, Yan Zhao, Aris Xie, Jonathan Lindner (Div. of Cardiovascular Medicine, OHSU, 3181 SW Sam Jackson Park Rd., Portland, OR 97239), Thomas R. Porter (Univ. of Nebraska Medical Ctr., Omaha, NE 68198), and Sanjiv Kaul M.D (Div. of Cardiovascular Medicine, OHSU, Portland, OR 97239)

Ultrasound and ultrasound contrast agent microbubbles (UCAM) are able to mechanically induce clot reduction. The aim of the study was to demonstrate the efficacy of various ultrasound conditions to enhanced thrombolysis in porcine clots inside a flow system. Clots were formed by infusing 4.1 ml of blood in transfer pipets. The pipets contained Dacron grafts to anchor the clot and initiated its formation. A 14-gage needle was placed at the center of the pipet and removed after clot formation to allow flow inside the clot. The clots were treated for 20 min with ultrasound and homemade UCAM at a concentration of 107 microbubbles/ml (flow rate 0.9 ml/min). Thrombolysis was monitored using an ultrasound scanner in pulse inversion mode. Results show that the radiation force causes the microbubbles to be pushed against the inner clot wall and cavitation induced lysis. The portion of the clot closest to the transducer was not affected by the therapy as the microbubbles were pushed in the direction of propagation. Mechanical clot reduction was observed in real-time in a flow system at various acoustic settings.

Contributed Papers

9:10

4aBA5. Quantified lysis of cell-like lipid membranes due to nanoparticle-facilitated cavitation. Michael J. Benchimol, Stuart D. Ibsen (Jacobs School of Eng., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA, 92093, mbenchim@ucsd.edu), Dmitri Simberg (Moores UCSD Cancer Ctr., La Jolla, CA 92093), Zhe Wu, Robert F. Mattrey (Univ. of California, San Diego, La Jolla, CA 92093), and Sadik C. Esener (Univ. of California, San Diego, La Jolla, CA 92093)

Certain nanoparticles act as nucleation sites for acoustic cavitation. Their surface roughness and hydrophobic regions can decrease the pressure required to induce the cavitation event. This concept has been examined for potential to provide a sensitizer for high-intensity focused ultrasound (HIFU), where cavitation can be beneficial. While passive cavitation detection can accurately determine the pressure threshold, variations in acoustic impedance/backscatter from different nanoparticles can make it difficult to normalize the amount of cavitation and determine the precise magnitude of the physical consequences. Here we demonstrate a method to determine the extent of lysis of lipid membranes using liposomes loaded with a self-quenching fluorophore. Lysed liposomes released the fluorophore, causing an increase in the fluorescence signal. Liposomes were mixed with the nanoparticles and insonated with a HIFU transducer at physiological temperature. The pressure threshold for dye release was measured for a panel of nanoparticles. Nearly complete release of the dye was achievable in all cases, but it required higher pressures in the absence of nanoparticles. In addition, we measured the effect of a viscous medium, which is more representative of certain physiological states. Furthermore, the encapsulation of the nanoparticle within a liposome can create a platform delivery vehicle for anti-cancer therapeutics.

9:25

4aBA6. Enhanced viral activity in tumors using focused ultrasound and microbubbles—A long term study. James J. Choi (Inst. of Biomed. Eng., Old Rd. Campus Res. Bldg., Univ. of Oxford, Oxford OX3 7DQ United Kingdom. james.choi@eng.ox.ac.uk), Robert C. Carlisle (Univ. of Oxford, Oxford OX3 7DQ United Kingdom.), and Constantin C. Coussios (Univ. of Oxford, Oxford OX3 7DQ United Kingdom.)

Oncolytic viruses target and kill cancer cells and self-amplify through replication. However, viral therapy *in vivo* is limited by insufficient systemic delivery. Here, focused ultrasound (FUS) was used in conjunction with microbubbles to produce cavitation and enhance tumor viral delivery. Human breast cancer cells (ZR75.1) were injected subcutaneously in mice (n = 8), which grew to a tumor volume of at least 30 mm^3. The tumors were then exposed to FUS (frequency: 0.5 MHz, pulse duration: 50 000 cycles, pulse repetition frequency: 0.5 Hz) for 4 min following injection of 100 μl of SonoVue microbubbles (SVM) with polymer-coated adenovirus encoding luciferase (pc-Ad-Luc). Experiments were repeated for three controls: pc-Ad-luc with neither FUS nor SVMs, FUS and SVM without pc-Ad-Luc, and buffer alone. Acoustic emissions were recorded with a passive detector, which validated the presence of inertial cavitation and ensured good SVM reperfusion kinetics. Tumor viral expression was then imaged using IVIS following luciferin injection. At 1, 2, 3, and 7 days post-injection, 3, 20.3, 30.2, and 22.9-fold increases in photons/s/cm^2 were observed when pc-Ad-Luc was used with FUS and SVM compared to pc-Ad-Luc alone. In conclusion, FUS and SVM enhanced oncolytic virus delivery resulting in amplified viral expression over time.

9:40

4aBA7. Ultrasound-induced temperature elevation for in-vitro controlled release of temperature-sensitive liposomes. Christophoros Mannaris, Eleni Efthymiou (Dept. of Mech. and Manufacturing Eng., Univ. of Cyprus, 75 Kallipoleos St. 1678 Nicosia, Cyprus), Jean-Michel Escoffre, Ayache Bouakaz (UMRS INSERM U930, CNRS ERL3106, Universite Francois Rebelais, Tours, France), Marie-Edith Meyre, Matthieu Germain (NANOBIOTIX, 60, rue de Wattignies bat. B, 75012 Paris), and Michalakis A. Averkiou (Univ. of Cyprus, 1678 Nicosia, Cyprus)

Drug loaded temperature-sensitive liposomes (TSLs) release their payload with mild hyperthermia near their phase transition temperature ($T_m = 43–45\,^\circ C$). Such a release may improve therapeutic efficacy and reduce toxic side effects in cancer treatment. In the present work, two different approaches are considered where focused ultrasound is used to induce the required temperature elevation for the release of doxorubicin from TSLs: (a) primary heating due to thermo-viscous absorption of ultrasound in absorptive media (oil, glycerol) and (b) secondary heating in non-absorptive media (blood, cell medium) due to heat transfer from the surroundings. Fine-wire thermocouple readings where in close agreement with theoretical predictions of temperature elevation with the Bioheat equation. Pulsing schemes to elevate and maintain the temperature at the desired value were designed with the Bioheat equation and validated with experiments. Fluorescence spectroscopy was used to assess the release of free doxorubicin that exhibits higher fluorescence intensity than the liposomal formulation. Significant drug release was achieved with both approaches.

9:55

4aBA8. Dynamical-systems measures of ultrasound contrast agent proximity to target walls. Fatimah Dzaharudin (Dept. Mech. Eng., Univ. of Melbourne, Melbourne, VIC 3010, Australia, f.dzaharudin@student.unimelb.edu), Sergey A. Suslov (Swinburne Univ. of Technol., Hawthorn, VIC 3122, Melbourne, Australia), Andrew Ooi (Univ. of Melbourne, Melbourne, VIC 3010, Australia), and Richard Manasseh (Swinburne Univ. of Technol., Hawthorn, VIC 3122, Melbourne, Australia)

Targeted ultrasound contrast agents are microbubbles that strongly scatter ultrasound, providing contrast on a scan, and have also been coated in molecules that adhere to target pathologies. The ultimate aim is to identify diseased tissue in clinical ultrasound practice. One issue is to discriminate in real time between microbubbles that have adhered to their target pathologies on blood-vessel walls, from those that are freely flowing in the bloodstream. It is known that linear theory predicts a shift in resonant frequency owing to the presence of a wall. Weakly nonlinear theoretical results are presented on alterations to the dynamical-systems behavior of one or more microbubbles on and near to walls. In particular, the bifurcation diagram is altered as

microbubbles approach a wall. Near a wall, period-doubling and period-quadrupling bifurcations and transitions to broadband chaos occur at altered values of the incident pressure amplitude. Alterations in the bifurcation diagram increase as multiple bubbles are held fixed close to each other and to the wall. This suggests that filtering of the returning echoes around selected solution branches could provide a further real-time indicator of locations where targeted ultrasound contrast agents have adhered.

10:10–10:25 Break

10:25

4aBA9. Spatio-temporal mapping and characterization of acoustic cavitation seeded by microbubbles and solid microparticles during focused ultrasound exposure. James J. Choi and Constantin C. Coussios (Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Univ. of Oxford, OX3 7DQ, United Kingdom, james.choi@eng.ox.ac.uk)

Cavitation nuclei are often used to seed and promote acoustic cavitation in therapeutic applications. However, the effects of fluid velocity and ultrasound exposure parameters peak rarefactional pressure (Pr), pulse duration (PD), and pulse repetition frequency (PRF) on the spatial distribution, type, magnitude, and number of acoustic cavitation events for each nuclei remains poorly understood. In this study, a 1.6-mm diameter tunnel phantom (3% agar) was perfused (fluid velocity: 10–40 mm/s) with either microbubbles (SonoVue) or hydrophobic solid microparticles (TALC) and exposed to 74 different acoustic parameter combinations (frequency: 0.5 MHz, Pr: 150–1500 kPa, PD: 1–100,000 cycles, PRF: 1–50 Hz, number of pulses: 10–250). Spatial mapping of passively acquired acoustic cavitation emissions was performed with a 64-element array coaxial to the focused ultrasound transducer. At pressures above the cavitation threshold, cavitation activity generated from microbubbles was significantly reduced and spatially biased upstream after the first pulse at high PRFs relative to the fluid velocity. On the other hand, solid microparticles had no spatial bias and no significant reduction in the energy of acoustic emissions after the first pulse. Whereas microbubbles may be destroyed, and therefore, cease to act as cavitation nuclei, solid microparticles do not suffer from depletion of energy with high PRFs.

10:40

4aBA10. *In vitro* acoustic characterization of a novel poly-lactic acid polymer shelled contrast agents. Shirshendu Paul, Daniel Russakow, Tyler Rodger, Kausik Sarkar (Dept. Mech. Eng., Univ. of Delaware, Newark, DE 19701), and Margaret Wheatley (Drexel Univ., Philadelphia, PA 19104)

Micron sized encapsulated gas bubbles have been extensively studied as contrast enhancing agents for ultrasound imaging. This study will report on *in vitro* acoustic characterization of a novel poly-lactic acid (PLA) shelled contrast agent. PLA is a bio-degradable polymer approved by the FDA to be used in drug delivery applications. Thus, PLA shelled contrast bubbles have the potential of being developed as the next generation contrast agents. Both attenuation and scattering measurements will be reported. Attenuation measurements are obtained using three different transducers (central frequencies 2.25, 3.5, and 5 MHz). Pressure dependent scattered response is obtained for two different excitation frequencies of 2.25 and 3.5 MHz. Results indicate excellent scattering properties of the PLA shelled bubbles. The strongly non-linear nature of scattered response makes PLA bubbles a potential choice for harmonic and sub-harmonic contrast imaging applications.

10:55

4aBA11. Experimental characterization of dye-loaded echogenic liposomes. Shirshendu Paul, Daniel Russakow, Tyler Rodger, Kausik Sarkar (Mech. Eng., Univ. of Delaware, Newark, DE 19716), Rahul Nahire, and Sanku Mallik (Pharmaceutical Sci., North Dakota State Univ., Fargo, ND 58108)

Liposomes are submicron sized vesicles with a lipid bilayer encapsulating an aqueous phase inside. Due to their favorable properties like longer circulation time, lesser toxicity, and greater uptake, they are a prime candidate for drug delivery. Recently, they are being specially prepared so as to encapsulate air, making them good scatterers of ultrasound wave. These echogenic liposomes, therefore, can be used both for ultrasound contrast imaging and drug delivery. We will report *in vitro* attenuation and scattering measurement from echogenic liposomes loaded with carboxyfluorescein (used as a surrogate for small molecular weight drugs). The results will be compared with non-dye-loaded ones. Effects of dye loading and presence of bovine serum albumin (BSA) on size distribution, echogenicity, and release characteristics will also be discussed.

11:10

4aBA12. Localized activation and cellular effects of ultrasound triggered drug delivery vehicles with encapsulated microbubbles. Stuart Ibsen (Dept. of Bioengineering, Moores Cancer Ctr., Univ. of California San Diego, 3855 Health Sci. Dr. # 0815, La Jolla, CA 92093-0815), Michael Benchimol, Dmitri Simberg (Univ. of California San Diego, La Jolla, CA 92093), Carolyn Schutt (Univ. of California San Diego, La Jolla, CA 92093-0815), Jason Steiner (Univ. of California San Diego, La Jolla, CA 92093), and Sadik Esener (Univ. of California at San Diego, La Jolla, CA 92093)

The harmful side effects of chemotherapy originate from indiscriminate exposure of healthy tissue to the drugs. The goal of targeted drug delivery is to reduce these side effects by encapsulating concentrated drug in a vehicle which releases it only in the tumor region. Low intensity focused ultrasound can be used as a trigger to specifically activate these vehicles by highlighting only tumor tissue, creating a stark differentiation with healthy tissue. A new injectable drug delivery vehicle has been developed with a stabilized nested lipid shell geometry that encapsulates a high capacity chemotherapy payload, and a stabilized microbubble into one structure. Ultrasound affects the microbubble only in the small focal volume, creating a localized shockwave which ruptures the vehicle's outer membrane triggering pinpoint release in tissue phantoms. These shockwaves, and their interactions with the delivery vehicle membranes and live cells, have been documented for the first time using a custom system which combines high-speed videography and fluorescent microscopy with focused ultrasound. Vehicles which do not pass through the tumor will be excreted through normal processes. This externally-activated scheme could lead to truly tumor-specific drug delivery. [NCI Grant No. 5U54CA119335-05, and UCSD Cancer Center Specialized Support Grant No. P30 CA23100 supported this work.]

11:25

4aBA13. Role of effective surface tension on the frequency dependent subharmonic threshold for contrast microbubbles. Amit Katiyar and Kausik Sarkar (Mech. Eng., Univ. of Delaware, Newark, DE 19716)

We numerically investigate the predictions from several contrast microbubble models to determine the excitation threshold for subharmonic generation. In contrast to the classical perturbative result, the minimum threshold for subharmonic generation is not always obtained near twice the resonance frequency; instead it can occur over a range of frequency from resonance to twice the resonance frequency. The quantitative variation of the threshold with frequency depends on the model, bubble radius, and encapsulation properties. All models are transformed into a common interfacial rheological form, where encapsulation is represented by two radius dependent surface properties—effective surface tension and surface dilatational viscosity. Variation of the effective surface tension with radius, specifically having an upper limit (resulting from strain softening or rupture of the encapsulation during expansion), plays a critical role. It destroys a sharp minimum at twice the resonance frequency. Without the upper limit on effective interfacial tension, the threshold is extremely large especially near the resonance frequency.

11:40

4aBA14. Three-dimensional dynamical equations of interacting bubbles. Eru Kurihara (Dept. of Eng., Oita Univ., Dannoharu, Oita City 870-1192, Japan, kurihara@oita-u.ac.jp)

It is known that bubble cavitation plays important role in kidney stone fragmentation in the shock wave lithotripsy and other medical applications of shock wave. The behavior of such bubbles, however, considerably complicated because of its nonlinearity and mutual interactions among the bubbles. For weakly nonlinear oscillations, dynamics of interacting bubble can be approximately expressed by the method of multi-pole expansion with

spherical harmonics. In the previous study, the author derived a set of dynamical equations of two interacting aspherical bubbles with Lagrangian mechanics. The axisymmetrical system of two interacting bubbles can be described in two-dimensional coordinate system, and then shape oscillation of the bubbles is expressed with Legendre polynomials. The bubble behavior described by the derived equations qualitatively agreed with experimental results by high-speed photographs. In the dynamics of three or more bubbles, however, the behavior of bubbles is essentially three dimensional, and thus the system of these bubbles should be represented by three-dimensional spherical harmonics (associated Legendre functions). In this study dynamical equations for three interacting aspherical bubbles are derived by multi-pole expansion in the framework of Lagrangian formalism. [Work supported by Grants-in-Aid for Scientific Research 23760142 and a research grant from The Mazda Foundation.]

THURSDAY MORNING, 3 NOVEMBER 2011 PACIFIC SALON 6/7, 8:30 TO 9:45 A.M.

Session 4aEAa

Engineering Acoustics: Energy Harvesting

Stephen C. Thompson, Chair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Contributed Papers

8:30

4aEAa1. Energy harvesting of tonal sound excited by heat addition and vortex shedding. Sungmin Jung, Rafael Hernandez, and Konstantin I. Matveev (MME School, Washington State Univ., Pullman, WA 99164, matveev@wsu.edu)

Tonal sound may appear inside resonating duct systems due to heat addition or vortex shedding in the presence of mean flow. Amplitudes of this sound can reach significant levels. The sound power can be captured and converted to electricity using electroacoustic transducers. Two pipe setups were constructed to demonstrate energy harvesting of tonal sound using piezoelements. In the first system, the sound was excited by vortex shedding and impingement on baffles in the presence of mean flow. The second system represented closed and open types of a standing-wave thermoacoustic engine. Electric power in excess of 0.5 mW was captured and released on passive electric loads from the tonal sound. This power level is sufficient for some low-power sensors. Further system optimization can significantly increase the amount of harvested energy. [Work supported by the NSF Grant 0853171.]

8:45

4aEAa2. Electromechanical transduction system design for optimal energy harvesting from ocean waves. Amadou G. Thiam and Allan D. Pierce (Dept. Mech. Eng., Boston Univ., Boston, MA 02215, thiam@bu.edu)

While details of the currently most highly publicized devices for conversion of ocean wave energy to electrical energy are generally not disclosed in the open literature, the authors believe that, for devices not on the coastline, the common transduction mechanism involves electromagnetic induction with conducting wires moving relative to permanent magnets. A general discussion is given of how such a mechanism can be used in this application. The overall analysis of the mechanical system with lumped or distributed masses and elastic elements driven by buoyancy forces associated with incident ocean waves is facilitated, if the transduction system is modeled as linear mechanical dashpots, and the procedures for deriving effective dashpot constants are described. The mechanical analysis suggests that, for waves in a general frequency range, there is an optimal choice for the parameters of the mechanical system, so that the maximum electrical power can be harvested. The optimal energy extracted per wave cycle is invariably much less than the total mechanical energy of the oscillating components of the system. A distinction is made between freely floating systems and systems anchored to the ocean bottom and between systems which are driven near a resonant frequency and those driven substantially below resonance.

9:00

4aEAa3. Energy conversion through thermoacoustics and piezoelectricity. Robert M. Keolian (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804-0030, keolian@psu.edu) and Scott Backhaus (Condensed Matter and Magnet Sci. Group, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Waste or prime heat can be converted into electricity with thermoacoustic-Stirling engines coupled to piezoelectric alternators. An inline arrangement of engines and alternators allows a vibration balanced, multiphase power generator that is compact, light weight and low cost. The engines convert heat into high amplitude $\approx$400 Hz oscillations in pressurized helium gas. These pressure oscillations cause a thin steel diaphragm to flex like a drumhead. The diaphragm is supported at its perimeter by a ring of piezoelectric elements. As the diaphragm flexes in either direction, it pulls inward on the piezoelectric elements causing a large amplified $\approx$800 Hz fluctuating compressive stress in the elements which then convert the stress into electricity with high efficiency. The flexible-diaphragm piezoelectric alternator overcomes the large acoustic impedance mismatch between the helium and piezoelectric elements without exceeding the limited fatigue strength of available materials. So far, a prototype generator has produced 37 W, and is being modified to produce 600 W. Also, a project is underway to recover 7 kW peak electrical power from the exhaust of an over-the-road heavy-duty diesel truck. The generator appears scalable up to megawatt power levels. [Work supported by DOE, ONR, Clean Power Resources, Innovation Works, and Applied Research Laboratory.]

9:15

4aEAa4. Modifying a balanced armature speaker for energy harvesting applications. Nikolas T. Vitt and Stephen C. Thompson (Appl. Research Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

Balanced armature transducers are produced in large quantity for use as miniature speakers in hearing aids and in-ear headsets. These devices are reciprocal and might be used as the generator in vibration energy harvesting. However, previous work has shown that, as manufactured, the speakers do not have sufficient vibration sensitivity to be directly used in this way. This

paper explores a set of design modifications to increase the sensitivity to a useful level. Preference is given to modifications that require minimum investment in new manufacturing tooling.

9:30

4aEAa5. Stepped-plate transducer as an energy transmitter. Yonghwan Hwang, Yub Je (Dept. of Mech. Eng., POSTECH, Pohang, South Korea, serenius@postech.ac.kr), SungQ Lee, Gunn Hwang (ETRI, Daejeon, South Korea, Hermann@etri.re.kr), and Wonkyu Moon (POSTECH, Pohang, South Korea)

Power transmission through acoustic energy may be useful in such cases as supplying power to wireless small sensors. In the case, the radiation and reception power efficiency is important. Even in ultrasonic frequency bands, the power efficiency of most acoustic radiators in air is not high enough. The stepped-plate ultrasonic transducers, introduced by Gallego-Juarez *et al.* [Ultrasonics **16**, 267–271(1978)], may be a good candidate for the radiator for this purpose because it can effectively generate highly directive, large-amplitude, ultrasonic sounds in air. The transducer consists of langevin transducer that causes wave generation, mechanical amplifier, and stepped radiation plate. Although it is reported to achieve 80% of power efficiency, it is not reported how to achieve maximum power efficiency. For design of large-amplitude, high-efficiency stepped-plate transducer, the design of not only individual parts but also system integration of entire transducer is important. In this research, we developed an analytical model for the whole transducer by combining continuum models of each parts and found proper design parameters for the radiation power and the power transmitting efficiencies. Then, we seek the optimal design for maximizing power efficiency through parametric analyses, and the results are confirmed through finite element method analysis. [Work supported by ETRI (South Korea).]

THURSDAY MORNING, 3 NOVEMBER 2011 PACIFIC SALON 6/7, 10:00 A.M. TO 12:00 NOON

Session 4aEAb

Engineering Acoustics and Underwater Acoustics: Vector Sensors, Projectors, and Receivers I: Projectors and Reversible Transducers

Stephen C. Butler, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Roger T. Richards, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Invited Papers

10:00

4aEAb1. The octoid modal vector projector. Alexander L. Butler and John L. Butler (Image Acoust., Inc., 97 Elm St., Cohasst, MA 02025, abutler@imageacoustics.com)

The eight piston octoid transducer is a descendent of the three piston trioid and four piston astroid transducers. These transducers were developed for low frequency underwater sound applications where wide bandwidth and small volume is desirable. Magnified piston motion is achieved through attached leveraged shell action driven by radial piezoelectric stacks. This condition yields magnified piston displacement and lowered Qm through magnified loading on the piezoelectric stacks. The new octoid design concept was implemented on an eight stack power wheel modal transducer. And this new design has been used to significantly reduce the outer diameter for the same response or, alternatively, yield lower frequency response for the same diameter. Finite element model and measured results are presented for the power wheel with and without the octoid leveraging and the octoid transducer is also compared to the previous trioid and astroid transducers.

4a THU. AM

10:20

4aEAb2. A compact Terfenol-D vector projector. Julie C. Slaughter (Etrema Products, Inc., 2500 North Loop Dr. Ames, IA 50010, julie.slaughter@etrema.com)

In active sonar applications, it is desirable to have a directional, steerable sound source to accelerate the target localization process. A narrow cardioid beam pattern with approximately 78° beam width and rear side lobes 25 dB down can be generated by combining monopole, dipole, and quadrupole beam patterns with appropriate amplitudes and phases. A cylindrical array composed of eight Terfenol-D Tonpilz transducers arranged in two rings with each transducer pointing radially outward from a common center mass has been developed to generate narrow cardioid beam patterns. Focus during the design process was on minimizing size, maximizing the bandwidth, and maximizing the duty cycle. The diameter of the source is 0.19 wavelengths at the lowest operating frequency and 0.76 wavelengths at the highest operating frequency. Bandwidth of the source is greater than two octaves. Lumped parameter modeling and finite element modeling are used to demonstrate the monopole, dipole, quadrupole, and narrow cardioid beam patterns. The effects of extraneous vibration modes on the beam patterns and sound output are discussed. Maximum continuous wave sound output and duty cycle at maximum output are estimated from thermal models and test data from a single Tonpilz element. [Work supported by the Office of Naval Research.]

10:40

4aEAb3. High power, broad bandwidth, compact, single crystal vector projector. P. David Baird, James M. Powers, and Ivan A. Rodriquez (Progeny Systems, 2401 South 1070 West, St. 16A, West Valley City, UT 84119)

An approach to producing a high power, broad bandwidth, compact transducer with steerable super-directionality will be described. The design, utilizing high coupling coefficient PMN-PT single crystal material, will be compared to conventional transducers utilizing PZT ceramic. Performance predictions for transmit response, impedance, efficiency, power, volt-amperes, source level, and bandwidth will be provided.

Contributed Papers

11:00

4aEAb4. Unidirectional multimode piezoelectric spherical transducers. David A. Brown, Boris Aronov, and Corey Bachand (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St., Falls River, MA 02723)

Analytical and experimental results are presented for unidirectional broadband multimode piezoelectric acoustic transducers utilizing axisymmetric vibrations of spherical transducers. The analysis covers the acoustic radiation and reception by including the acoustic impedance's and diffraction coefficients for transducers with conformal baffles. The energy method is used to obtain equivalent parameters for a multi-contour electromechanical circuit representation of the transducer and to calculate performance. Experimental data obtained are in good agreement with analytical results.

11:15

4aEAb5. Compact cylindrical transducer arrays for directional communications and navigation. David Brown and Boris Aronov (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St., Falls River, MA 02723)

Designs and experimental results are presented for compact self-baffled cylindrical arrays comprising piezoelectric transducers suitable for telemetry stations and small vehicle directional communications or navigation. Designs with cylindrical transducers, rod/bar transducers, and tonpilz elements are compared with single transducers using turnable baffles to achieve unidirectional beams that may be steered in azimuthal plane.

11:30

4aEAb6. Improved spiral-wavefront transducer for underwater acoustic navigation. David Brown, Boris Aronov, and Corey Bachand (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St., Falls River, MA 02723)

Spiral-wavefront transducers have received recent attention as enabling elements in phase-based underwater navigation systems, whereby the signaling transducer launches a diverging wave that is omnidirectional by magnitude but linearly phase-biased with azimuthal angle. The detected signal is compared with a control signal having constant phase to allow the unique determination of bearing angle. Range can be determined by time-of-flight. While such systems are used in air traffic control navigation, the transition to underwater acoustics had only emerged with the advent of increased unmanned underwater vehicle traffic (Dzikowicz and Hefner). Analytical and experimental test results, including beam patterns, TVR, and power factor, are presented for resonance operations at 25 kHz (c-band). [Work supported by BTech Acoustics LLC.]

11:45

4aEAb7. Design of a wideband multimode tonpilz transducer with a nonuniform piezoelectric layer stack. Yongrae Roh and Saosometh Chhith (School of Mech. Eng., Kyungpook Natl. Univ., 1370 Sankyukdong, Bukgu, Daegu 702-701, Korea, yryong@knu.ac.kr)

It has been well-known that a multimode transducer could provide a wider frequency bandwidth than a single mode one, and conventionally a multimode transducer can be achieved by designing the geometry of its head mass, increasing the head mass radius and reducing the head mass thickness. However, a very large head mass can cause some drawbacks to transducer performance when used in an array, i.e., low source level and big crosstalk with neighboring transducer elements. In this work, a new and very simple design method has been developed to widen the bandwidth of a Tonpilz transducer, which is replacing the uniform PZT layer stack by a nonuniform one. A piezoelectric stack composed of nonuniform PZT layers can generate higher mechanical energy than that composed of uniform layers for the same input electrical energy, which means a higher coupling coefficient thus a wider bandwidth. The effects of the nonuniformity of PZT layer thicknesses on a multimode Tonpilz transducer performance were investigated through finite element analyses. Then, the functional forms of the performance were derived in relation to the nonuniform PZT thicknesses and were inserted into a genetic algorithm to achieve the widest possible bandwidth of the Tonpilz transducer.

Session 4aMUa

Musical Acoustics: Physical Models For Sound Synthesis II

Edgar J. Berdahl, Chair
Dept. of Music, Stanford Univ., Stanford, CA 94305

Contributed Papers

8:00

4aMUa1. Estimating the clarinet mouthpiece reflection from measurement and the instrument's produced sound. Tamara Smyth (School of Computing Sci. Simon Fraser Univ., 50-13450 102nd Ave. Surrey, B.C., Canada V3T 0A3, tamaras@cs.sfu.ca) and Jonathan Abel (CCRMA, Dept. of Music, Stanford Univ., Stanford, CA, abel@ccrma.stanford.edu)

In this work, a method is presented for estimating the reflection off the clarinet mouthpiece, using a priori measurement of the bell, and post processing of the instrument's produced sound. A previously introduced measurement technique is used to obtain measurement of clarinet bell and transmission filters. In addition to these elements, however, the round-trip propagation loss in the clarinet bore and bell also includes wall loss and mouthpiece reflection. Though the former is accurately modeled theoretically, assuming the clarinet bell is close to cylindrical, the mouthpiece is more difficult to measure, both because of a supposed oscillating reed, and because the required placement of a measurement device would obstruct the mouthpiece's characteristic reflection. The lumped round-trip loss filter in the bore is estimated from the clarinet signal by first considering the signal's periodic structure. After taking the signal's autocorrelation, which preserves its periodicity and naturally provides the beginning of the period, the round-trip filter is iteratively estimated by constructing an optimization function from the first and second phases of the autocorrelation sequence. Once the round-trip loss is estimated, the mouthpiece reflection may be extracted by removing the effect of the other known comprising filter elements.

8:15

4aMUa2. Inverse problem in sound synthesis and musical creation using mass-interaction networks. Jérôme Villeneuve and Claude Cadoz (Laboratoire ICA, 46 Ave. Felix Viallet, 38000, GRENOBLE, FRANCE, jerome.villeneuve@imag.fr)

Sound synthesis with mass-interaction physical modeling networks is known as a general paradigm capable of being the central part of complete software environments for both sound synthesis and musical creation. GENESIS 3, resting on the CORDIS-ANIMA formalism and developed by ACROE/ICA Laboratory, is the first environment of this kind. Using it, the artist may be facing an inherent problematic of every creation process: how to use a given tool in order to obtain an expected result. In our context, the question would be: "Considering a sound, which physical model could produce it?" Our work aims at formalizing this inverse problem and therefore at helping the user through his creative process. Therefore, we will consider how he could describe a sound (entry of the inverse problem), how to define a generator model based on mass-interaction physical networks and each one of its subcomponents (formal solution of the inverse problem), and, obviously, how to compute this solution considering an entry (resolution of the inverse problem). In this paper, we will develop each one of those three points and present the first algorithmic resolutions already implemented and used within GENESIS 3.

8:30

4aMUa3. Real-time finite-difference string-bow interaction floating point gate array (FPGA) model coupled to a violin body. Pfeifle Florian and Bader Rolf (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, 20354 Hamburg, Germany, Florian.Pfeifle@haw-hamburg.de)

A string/bow interaction model as proposed by Bader, R.: Whole geometry Finite-Difference modelling of the violin. In: Proceedings of the Forum Acusticum 2005, 629-634, 2005 and a 3D violin geometry with top plate, back plate, and enclosed air was implemented in real-time using an FPGA hardware implementation. The bow/string interaction uses conditions for gluing and tear-off the string to/from the bow, where gluing happens with relative string/bow velocity below a threshold and tear-off with either the string curvature or the string tension at the bow point too high modelling the cases appearing with normal Helmholtz motion or the occurrence of subharmonics, respectively. The model can be played changing bow pressure and velocity, bow point on the string, and string length. The resulting sounds are highly realistic showing gradual timbre changes from normal Helmholtz motion, double slit motion, weakening of the fundamental, or noise with very low pressures. It appears that these interaction conditions can realistically be used to model the bow/string interaction.

8:45

4aMUa4. Measurement and physical modelling of sound hole radiations of lutes. Florian Pfeifle (Musicological Inst., Univ. of Hamburg, Neue Rabenstrasse 13, 20354 Hamburg, Germany, Florian.Pfeifle@haw-hamburg.de)

A structural feature that can be found in many string instruments is a hollow body with one or several sound holes. The sound radiated from these holes interacts with the sound radiation from the rest of the body and perceivably influences the timbre and the loudness of the instrument. In this work three non-European lutes with sound holes are measured: the Mauretanian ginbri, the West-African gunubri, and the Chinese ruan. All of these instruments have distinct cavity air modes and a measureable Helmholtz frequency. Each instrument is measured with a 11×11 microphone array and analyzed with a focus on the radiated spectrum and sound intensity of the hole(s). In a further step, the findings of the measurements are compared to a Finite Element model and incorporated into a real-time finite differences physical model of the ruan.

9:00

4aMUa5. Synthesizing classic recording microphones characteristics using a spherical microphone array. Nils Peters and Andrew W. Schmeder (Ctr. for New Music and Audio Technologies, UC Berkeley, 1750 Arch St., Berkeley, CA 94720, nils@icsi.berkeley.edu)

Using spherical microphone arrays to form directed beams is becoming an important technology in sound field analysis, teleconferencing, and surveillance systems. Moreover, in scenarios for capturing musical content, the

recording and post-production process could be simplified through flexible beamforming technology. Often, audio engineers favor the use of conventional recording microphones over spherical microphone arrays which might be due to the engineer's preference for distinct spatial and timbral characteristics of different microphone types and brands. We present an approach to create beamforming pattern using a 144 channel spherical microphone array, which aims to match the distinct spatial and timbral characteristics of classic microphones. For this, we first measured the spatial and timbral characteristics of several classic microphones types as well as the characteristics of our spherical microphone array in an anechoic chamber. Using a regularized least-square approach, these data were then used for computing the filters for the spherical microphone array that forms the desired beams. We show the results of several microphone-beam simulations and compare them with the impulse responses of the original classic microphones. Advantages and limitations of our approach will be discussed.

9:15

4aMUa6. Numerical simulation of the sound of organ pipes. Michael Steppat (Beuth Univ. of Appl. Sci., Luxemburger Str. 10, 13353 Berlin, Germany, steppat@beuth-hochschule.de)

Numerical simulations of musical instrument sounds are useful in studies, when questions about the influence of physical parameters stand in first place. Differential equations can be used to solve the given problem. The results given as a discrete sequence can be stored in audiofiles and made audible. In the current project, the influence of the attack of organ pipes with respect to the reverb time of the acoustical environment in the church building is studied by different numerical calculation methods such as finite element method and computational fluid dynamics. The resulting sound pressure and the velocity in a vortex street of the jet emerging from a pipe can be calculated at different listening positions. The computation allows also visualization of the jet in a three dimensional view. In the tests, the same parameters used by organ builders to voice organs are used for the simulation. Main point is the strength of the resulting starting transient and its influence to the acoustical environment. The calculation process which includes the synthesizing and adding a reverb and the results with different voicing parameters will be presented.

9:30

4aMUa7. FireFader: A single degree-of-freedom force-feedback device for multimodal interaction with physical models. Edgar J. Berdahl (Audiokommunikation, Tech. Univ. of Berlin, Sekr. EN-8, Einsteinufer 17c, 10587 Berlin, Germany, eberdahl@mail.tu-berlin.de)

The design of a relatively inexpensive force-feedback device known as the FireFader is presented. It is controlled using physical models to provide multimodal force, auditory, and visual feedback in real time, and it is based upon a linear potentiometer fader coupled to a DC motor, also known as a "motorized fader". Lamps are connected electrically in parallel with the motor in order to visually communicate the strength of the force. The device is linked by a serial USB interface to a general-purpose computer, which employs a physical model to calculate the motor force as a function of the fader position. The USB interface causes delay of the control signal, but it facilitates easier programming and less expensive control. Furthermore, additional sensed parameters can help provide the illusion of more than a single degree-of-freedom (DOF) feedback, via modulation of the physical model parameters. For estimation of the downward force applied by the performer on the fader, a pair of force sensors can be sandwiched in between the motorized fader and the housing. In conclusion, we hypothesize that by providing multimodal feedback in real time, the FireFader may help promote the expressivity of new media interactions.

THURSDAY MORNING, 3 NOVEMBER 2011 TOWNE, 10:00 TO 11:30 A.M.

Session 4aMUb

Musical Acoustics: Analysis of Instrument Sounds

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

Contributed Papers

10:00

4aMUb1. Tuning parameters of the Nigerian slit drum. Ashley Cannaday and Thomas Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789)

The slit log drum, sometimes referred to as a "talking drum," is an idiophone indigenous to many African and South Pacific cultures. It is made from a log that has been hollowed out through two square openings, which are separated by a solid piece of the wood. The solid piece is cut down the middle to produce two tabs that when struck produce pitches that are usually separated by a musical fifth. We report on an investigation of the tuning parameters of slit log drums from Nigeria using both numerical and analytical models. We conclude that the most efficient and effective method of tuning the drum is to carve the interior walls near the tabs so that they have different thicknesses, which is indeed is how the Nigerian artisans produce the two distinct pitches.

10:15

4aMUb2. Calculation of Helmholtz frequency of a Renaissance vihuela string instrument with five tone holes. Rolf Bader RB (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, 20354 Hamburg, Germany, R_Bader@t-online.de)

A replica of the historical *Guadalupe* vihuela, a Renaissance string instrument was investigated. As it has five sound holes a formula was developed to calculate its Helmholtz frequency radiated from these five holes. Therefore, using a 128-microphone array, the radiation pattern at the Helmholtz frequency of 138 Hz was measured showing strong radiation at the holes near the center of the body and lower radiation at the holes near the upper top plate boundary. Calculating a mean radius from the single radii and weighting them with their radiation strength, the measured Helmholtz frequency is calculated correctly. This is not the case when including the ornamentation of the sound holes into the calculation which indeed cover

65% of their area. Additionally, the overall radiation from the different top plate parts of this vihuela was compared to that of another much smaller vihuela and those of a classical guitar showing the *Guadalupe* replica to have a very large frequency range of strong sound hole radiation up to 500 Hz, where the classical guitar is stronger in the bass but its sound hole radiation part is restricted to lower frequencies. This makes the vihuela a mixture between a guitar and a lute.

10:30

4aMUb3. Video analysis and modeling of the kalimba tine. Daniel O. Ludwigsen (Phys. Dept., Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu)

The kalimba, like the more traditional mbira, uses plucked metal tines mounted to a wood resonator. The mounting is similar to a three point flexural test and provides an initial strain to part of the tine. The plucked end of the tine is free once released, but modeling the other end is less obvious. The motion of the longest tine of the treble kalimba (B3, 247 Hz) was captured via high speed video (1200 f.p.s.). Analysis of tine displacement informed the boundary conditions of an Euler–Bernoulli model for this thin beam vibration, which in turn predicted mode shapes and frequencies. The two lowest mode frequencies can be compared to prominent features in the spectrogram of recorded tones. Rapidly decaying harmonic content observed in the spectrogram is not suggested by this simple model; a more sophisticated approach is required to fully understand the behavior of the kalimba tine.

10:45

4aMUb4. Validation of a descriptor, the "sum function", related to "quality" and derived from the input impedance of wind instruments. R. E. Causse, P. Eveno, B. Kieffer (IRCAM (UMR CNRS 9912), 1 place Igor Stravinsky, 75004 Paris, France), J. Gilbert, J. P. Dalmont (LAUM (UMR CNRS 6613), Le Mans, 72085 France), and J. F. Petiot (IRCCYN (UMR CNRS 6597) Ecole Centrale de Nantes, 44321 Nantes, France)

The quality of a musical instrument embraces many aspects such as tuning, ease of play, tone, etc. This study aims to validate the use of the sum function (SF) proposed by Wogram from the measurement of input impedance as a descriptor of quality. This work is part of a wider project, PAFI (Aid Platform for Instrumental Making), supported by the French National Agency of Research. To validate the choice of the SF, we created a family of trumpets made from a basic instrument for which the leadpipe will be slightly modified for each model. The SF was calculated for a range of selected frequencies from the measurement of the input impedance of these different trumpets. The next step was to ask musicians experts to play these instruments, to measure the playing frequencies, and to note their feedback about quality. These tests were supplemented by comparative tests carried out this time with the help of a robotic artificial mouth. The final step involved was to try to identify correlations between the SF and the results of various tests and propose correction factors to be made to the formula of the SF, related to the nuance or range for example.

11:00

4aMUb5. Modal response and sound radiation from a hammered dulcimer. Benjamin Y. Christensen, Kent L. Gee, Brian E. Anderson, and Alan T. Wall (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, ukeben@gmail.com)

The sound radiation of the hammered dulcimer has been investigated. The dulcimer studied is a 16/15 fixed-top instrument of typical size with Baltic birch (laminated plywood) soundboard and back. To determine the instrument body resonances, the dulcimer was driven at the treble and bass bridges with a shaker. Accelerometers were used to obtain the resonance frequencies of the soundboard and back, and a microphone was placed inside the instrument to obtain the cavity resonances. The individual resonance peaks found were further investigated using scanning laser Doppler vibrometry and near-field acoustical holography. Preliminary results show that there is little modal response of the instrument at the fundamental frequencies of the lowest notes of the dulcimer. In addition, the vibration coupling to the back plate through the internal bracing causes it to serve as a second soundboard. Lastly, the holography results indicate significant radiation from the sound holes at some frequencies, which may contradict the commonly held notion that the dulcimer sound holes are largely decorative.

11:15

4aMUb6. Equivalent circuit modeling and vibrometry analysis of the Udu Utar Nigerian drum. C. Beau Hilton (Dept. of Humanities, Classics, and Comp. Lit., Brigham Young Univ., 4110 JFSB, Provo, UT 84602, wearoscarwilde@myway.com), Brian E. Anderson, and Hillary Jones (Brigham Young Univ., N283 ESC, Provo, UT 84602)

The udu drum is both an aerophone and an idiophone played with both of the musician's hands. It originates from Nigeria where it began as a functional water pot made out of fired clay. At some point, a side hole was cut into the pot and it became a musical instrument, traditionally played by women, which had an important role in religious ceremonies. The udu is capable of producing deep tones that result from acoustic resonances similar to those of Helmholtz resonators, though with a second hole. It also may produce higher pitch sounds that result from the musician tapping the surface of the udu. This paper will discuss one-dimensional equivalent circuit modeling of the acoustic resonances of the udu. A comparison of the resonance frequencies in the equivalent circuit modeling to measured resonance frequencies will be given. Additionally, an analysis of the structural modes of the udu as measured by a scanning laser vibrometer will be given, along with some insights into the sound produced by striking the drum at different locations. This information may be used by udu designers to better tune these instruments.

Session 4aNS

Noise and Physical Acoustics: Launch Vehicle Noise I

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

R. Jeremy Kenny, Cochair
Marshall Space Flight Center, Bldg. 4203, Huntsville, AL 35812

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. Further development of a launch pad noise prediction model. Kenneth J. Plotkin (Wyle Labs., 241 18th St. South, Ste. 701, Arlington, VA 22202) and Bruce T. Vu (NASA Kennedy Space Ctr., Mail Code NE-M1, Kennedy Space Ctr., FL 32899)

A model, PAD, has been developed for prediction of noise in the vicinity of launch vehicles, with specific application to the mobile launcher and tower for the Ares I launch vehicle. It follows the basic principles of a traditional NASA model (NASA SP-8072, 1971), with updated source components, including impingement and water suppression. The recent 5% scale Ares scale model acoustic test (ASMAT) exhibited sources not properly represented in PAD. These sources are noise increase associated with flat plate deflection of a supersonic plume, and generation of noise from impingement of the plume on an edge such as a launch mount or the edge of the exhaust hole in the launcher deck. New sources, based on ASMAT measurements, have been added to PAD to account for these effects. Treatment of the launch deck has also been generalized to permit full three dimensional launcher configurations, rather than a simple two dimensional arrangement with the tower over the flame trench. The prediction domain has also been expanded to include exhaust plume noise levels on the underside of the deck. (Work supported by the National Aeronautics and Space Administration.)

8:25

4aNS2. Effects of diffraction for acoustic loads on building structures. Louis C. Sutherland (5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275)

Acoustic loads on building structures from external noise are influenced by acoustic diffraction of the incident wave front. An experimental study by Wyle Labs of diffraction consisted of measurements on a solid ground-mounted cubical obstacle insonified by approximately plane waves. The diffraction effect is equal to the sound level at various positions on the cube relative to the free field sound level. The former was measured with a microphone mounted flush with the surface of the cube. As expected, the diffraction effect varied systematically with the ratio of the cube side to wavelength and is portrayed by contours of equal diffraction correction. The maximum correction was equal to or more than the expected 6 dB near the center of the side facing the source but also exceeded 6 dB near the center of the back (shadow) side of the cube when the ratio of cube side to wavelength is close to 1. This positive diffraction correction may not normally be considered in assessment of noise on the shadow side of buildings. The experimental data are shown to be consistent with diffraction theory for a single cube. Theoretical predictions of diffraction effects for spherical and cylindrical obstacles are also shown.

8:45

4aNS3. Effects of ground impedance on large rocket motor noise measurements. Debbie Pilkey (ATK Aerosp. Systems, UT40-LF3, P.O. Box 707, Brigham City, UT 84302, deborah.pilkey@atk.com), R. Jeremy Kenny (Acoust. and Stability Team, ER42 , NASA Marshall Space Flight Ctr., Huntsville, AL 35812), and Jared Haynes (Qualis Corp. / ESTS Group, NASA Marshall Space Flight Ctr., Huntsville, AL 35812)

Free field acoustic measurements have been collected on large solid rocket motors at the ATK test facility in Promontory, Utah. Ground effects have been measured to understand the impact on the overall data collection effort. Ground effects were measured at two test stands with vastly different terrain that hold the reusable solid rocket motor (RSRM) and RSRMV (five-segment RSRM) static test motors. Techniques for measuring and understanding ground effects are investigated, and examples presented from two different methods.

9:05

4aNS4. A review of large solid rocket motor free field acoustics. Debbie Pilkey (ATK Aerosp. Systems, UT40-LF3, P.O. Box 707, Brigham City, UT 84302-0707, deborah.pilkey@atk.com) and R. Jeremy Kenny (Acoust. and Stability Team, ER42, NASA Marshall Spaceflight Ctr., Huntsville, AL 35812)

Approximately twice a year, ATK Aerospace Systems has static fired an assembled reusable solid rocket motor (RSRM) in a horizontal configuration at its test facility in Utah. The firings took place on elevated terrain with the nozzle exit plume mostly undeflected and the landscape allowing placement of microphones within direct line of sight to the exhaust plume. NASA and ATK underwent a significant effort over several years to collect acoustic data in the free field and on the motor itself during RSRM static test firings. These data were used to characterize the acoustic field and to update the prediction methodologies in the monograph NASA SP-8072 "Acoustic Loads Generated by the Propulsion System." This work represents a review of the free field acoustics generated during large solid rocket motor firings and may be the only repeatable free field acoustic experiment on motors like these.

9:25

4aNS5. Mechanism of acoustic radiation from Supersonic Jets Impinging to Inclined Flat Plates. Seiji Tsutsumi, Taku Nonomura, Kozo Fujii (JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa, 252-5210, Japan), Yuta Nakanishi, Koji Okamoto (Univ. of Tokyo, Kashiwa, Chiba, 277-8561, Japan), and Susumu Teramoto (Univ. of Tokyo, Bunkyo-ku, Tokyo, 113-8656, Japan)

Acoustic wave radiated from supersonic cold jets impinging to inclined flat-plates is investigated numerically with the help of the experimental work. This type of acoustic generation is important for estimating and minimizing the acoustic loading of launch vehicle at lift-off. Through the study on the 45-degree-inclined flat plate located 5 D downstream from the nozzle exit, two noise sources are found to be generated due to the jet impingement: (1) interaction between the vortex of the jet shear-layer and the shock waves appearing at the jet impingement region and at the downstream region, (2) the Mach wave radiated from the large-scale vortex structure of the flow downstream of the plate. The former is similar to the broadband shock-associated noise. Those features are clearly confirmed by applying the proper orthogonal decomposition analysis to the numerical result. Prediction accuracy of 5 dB in far-field OASPL is obtained in the current numerical technique.

9:45

4aNS6. Large-Eddy-Simulations of over-expanded supersonic jet noise for launcher applications. Jean-Baptiste Dargaud, Julien Troyes, and François Vuillot (Onera - The French Aerosp. Lab, F-92322 Châtillon, France)

During the lift-off phase of a space launcher, powerful rocket motors generate harsh acoustic environment on the launch pad. Following the blast waves created at ignition, jet noise is a major contributor to the acoustic loads received by the launcher and its payload. This paper describes recent simulations performed at ONERA to compute the noise emitted by solid rocket motors at lift-off conditions. Far-field noise prediction is achieved by associating a LES solution of the jet flow with an acoustics surface integral method. The computations are carried out with in-house codes CEDRE for the LES solution and KIM for Ffowcs Williams and Hawkings porous surface integration method. This work has been conducted in the framework of the cooperation on launcher acoustics between CNES (French National Space Agency) and JAXA (Japan Aerospace Exploration Agency) involving the French AEID research group. The test case is that of a reduced scale solid rocket motor, fired vertically and has been provided by JAXA. Computations were run for varied numerical conditions, and the final paper will detail results and discuss comparisons with experimental acoustic measurements.

10:05–10:20 Break

10:20

4aNS7. On the use of tailored functional bases for space launcher noise sources localization and reconstruction. Damiano Casalino, Samuele Santini (CIRA, Italian Aerosp. Res. Ctr., Capua, I-81043, Italy, d.casalino@cira.it), Mariano Genito (ELV S.p.A., Colleferro, I-00034, Italy), and Valerio Ferrara (AVIO S.p.A., Colleferro, I-00034, Italy)

A source localization and inverse reconstruction methodology is applied to analyze wall pressure signals measured on the surface of a space launcher mock-up. The methodology is based on the use of elementary acoustic solutions tailored to the space launcher geometry as functional basis. Two classes of functional bases are considered: plane waves impinging on the surface of an infinite cylinder computed analytically through a literature formula, and plane waves impinging on the real space launcher computed numerically through a FEM computational acoustic technique. In both cases, the 3D acoustic field resulting from the impingement of an arbitrarily oriented plane wave is obtained by a truncated series summation of elementary axial-symmetric solutions. These two classes of functional bases are proven to provide consistent results in the addressed frequency range. However, although the analytical basis enables a faster localization of the noise sources that take place during the rocket firing, the numerical basis is expected to enable, at an acceptable computational cost, a more reliable reconstruction of the acoustic loads on the space launcher surface in a higher frequency range. The proposed FEM-based beam-forming and source reconstruction technique is therefore a useful tool for the vibro-acoustic design of future launch vehicles.

10:40

4aNS8. Energy-based acoustic measurement system for rocket noise. Michael M. James (Blue Ridge Res. and Consulting, LLC, 15 W. Walnut St., Ste. C, Asheville, NC 28801, Michael.James@BlueRidgeRes.com) and Kent L. Gee (Brigham Young Univ., Provo, UT 84602)

Accurate estimates of the vibroacoustic loading placed on space vehicles and payloads during launch require knowledge of rocket noise source properties and near-field acoustic energy flow characteristics. Without these data, structures may not be designed to handle the correct vibroacoustic loads, which can result in either an over-built, excessively massive structure or an under-designed vibration mitigation system that could result in damage to payloads. These measurements are difficult to perform because of the extreme nature of the acoustic and temperature environments near the rocket plume as well as the large physical size of the rocket noise source. With these design constraints in mind, a field-deployable data acquisition system and energy-based measurement probe have been developed to measure the magnitude, directivity, and spectral content of the rocket source. Initial measurements with various prototypes were conducted during a static test fire at ATK Space Systems Test Services in Promontory, Utah with limited results presented here. [Work sponsored by NASA John C. Stennis Space Center.]

Contributed Papers

11:00

4aNS9. Low-frequency calibration of a multidimensional acoustic intensity probe for application to rocket noise. Jarom H. Giraud, Kent L. Gee, Scott D. Sommerfeldt, R. Troy Taylor (Dept. of Phys. and Astronomy, Brigham Young Univ., Eyring Sci. Ctr., Provo, UT 84602, kentgee@byu.edu), and Jonathan D. Blotter (Brigham Young Univ., Provo, UT, 84602)

Measurements of acoustic vector quantities in the near field of a solid-fuel rocket motor are useful for enhancing prediction models related to rocket noise. However, the probes used to measure these quantities traditionally have low-frequency bandwidth limitations (e.g., below 45 Hz) thereby excluding the lowest, and in some cases, the loudest frequencies generated by large rocket motors. At these low frequencies, the phase and magnitude mismatch between microphones become greater and the acoustic phase separation between any two microphones becomes smaller, resulting in more error in estimating the pressure gradient between microphones. To investigate the low-frequency response of an acoustic intensity probe, a turntable is used to rotate a four-microphone probe with variable microphone spacing in a low-frequency noise field and an experimental assessment of the bandwidth is given for both magnitude and directional response. Also discussed is the effectiveness of a microphone interchange calibration technique to remove amplitude and phase mismatch and increase the usable bandwidth of the probe.

11:15

4aNS10. On the use of prepolarized microphones in rocket noise measurements. R. Troy Taylor, Kent L. Gee, Jarom H. Giraud, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., Eyring Sci. Ctr. 84602, kentgee@physics.byu.edu), Jonathan D. Blotter, and Curtis P. Wiederhold (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

The acoustic field near large-scale solid rocket motors represents a harsh, high-amplitude noise environment rich with high-bandwidth acoustic shocks. Type-1 prepolarized microphones may be used in these environments with the benefit of reduced cost and measurement because they require only a constant-current supply available in many data acquisition systems. However, there are two potential issues related to microphone response that should be considered. The first is a well-known RC-lowpass filter effect that is associated with using insufficient current to drive long cables with relatively high capacitance. The second has to do with temporary failure of the constant-current supply due to an insufficiently fast response time in representing rapid voltage changes at shocks, which results in spurious, capacitive-like effects in the waveform data that are also manifest as a low-frequency roll-up in the spectrum noise floor. An experiment was conducted to identify under what circumstances these waveform effects arise. Data were measured from a solid rocket motor using several combinations of transducer, cable type, cable length and constant current supply. Results and mitigation methods found from the experiment are discussed. These include increasing the supply current, using low-impedance cables, and selecting microphones with low sensitivities.

11:30

4aNS11. Statistical analysis of noise from solid rocket motors. Stuart A. Harper, Kent L. Gee, Jarom H. Giraud, and Michael B. Muhlestein (BYU Dept. of Phys. and Astronomy, N283 ESC, Provo, UT 84602)

In an effort to better understand the properties of rocket noise, the statistical properties of noise data from various-sized solid propellant rocket motors. Time waveform data sampled at 204.8 kHz using 6.35 and 3.18 mm microphones were collected near motors with nozzle exit diameters ranging from 0.13 to 1.22 m. Non-Gaussian features of the data have been explored by calculating estimates of the probability density functions of the data and various statistical moments, including skewness and kurtosis. This has been carried out for both the pressure waveform and its first order time difference to better understand the formation of acoustic shocks within the noise. The analysis shows greater similarity between the statistics for the pressure than for the time derivative estimates.

11:45

4aNS12. On the computation of farfield cross-spectra and coherences from reduced parameter models of high speed jet noise. Havard Vold, Parthiv Shah, and Mike Yang (ATA Eng., Inc., 688 Deepwood Dr. Charleston, SC 29412)

Reduced parameter models of jet noise mechanisms serve as computational vehicles to estimate sound pressure and directivies at arbitrary locations in the farfield. The customary formulations have been successful at predicting autospectra and directivities, but the calculation of crossspectra and coherences has not been attempted. The authors will present a procedure for calculating crossspectra and coherences from the simple source model, i.e., an equivalent monopole density over a volume enclosing the jet noise sources. It will be shown that realistic coherences and crossspectra are only well defined when several mutually incoherent noise sources are being considered, and both convergence and spatial aliasing phenomena will be defined and investigated.

Session 4aPA

Physical Acoustics: Measurements, Applications, and More

Martin D. Verweij, Chair
Lab. of Electromagnetics Research, Delft Univ. of Tech., Mekelweg 4, Delft, 2628 CD, The Netherlands

Contributed Papers

7:30

4aPA1. Photoacoustic detection of thin layers of explosives. Logan Marcus, Richard Raspet, and Slava Aranchuk (Univ. of Mississippi, NCPA, 1 Coliseum Dr. University, MS 38677, LSMarcus@olemiss.edu)

Explosives may be identified by their infrared absorption spectra. Photoacoustic measurements, which measure the acoustic output generated by the explosive sample absorbing radiation from a modulated laser beam, are efficient at detecting and identifying explosives. We are investigating the stand-off photoacoustic detection of surface traces of explosives on a variety of substrates using a laser Doppler vibrometer. This talk describes the detailed modeling of the measurement. A modulated infrared Gaussian profile laser source is absorbed in the layer of explosive and in the substrate. The heat absorbed raises the temperature of the explosive and the substrate leading to expansion and surface displacement. The heat absorbed by the gas adjacent to the solid generates an outward propagating acoustic wave. A laser Doppler vibrometer measures the phase shift due to the surface displacement and the compression and refraction of the acoustic wave. A cylindrically symmetric calculation of each physical step will be described. Calculations demonstrating how substrate parameters such as coefficient of thermal expansion, IR absorptivity, and Poisson's ratio effect the detection threshold of the measurements will be presented.

7:45

4aPA2. Sliding dynamic studies by use of elastography. Soumaya Latour, Stefan Catheline, Michel Campillo, Francois Renard, Christophe Voisin, Eric Larose, and Thomas Gallot (ISTERRE, Grenoble Univ., 38000 Grenoble, soumaya.latour@obs.ujf-grenoble.fr)

To get an insight into the processes underlying dynamic friction that plays an important role in seismic sources, we developed a sliding dynamic experiment coupled to elastography imaging. This experimental setup permits to observe simultaneously the frictional interface and the waves emitted in the bulk during slipping. We use soft solids made of hydro-organic gel of PVA, in contact with either glass or sandpaper. The huge interest of such soft solids is that elastography allows to observe in real time the rupture nucleation and propagation, as well as shear waves themselves inside the medium. We investigate the friction in two different cases. In the case of friction on sand paper, links are formed between the gel and the sand paper by local pinning. The breaking of these links emits a characteristic wave pattern, and their occurrence is related to the local sliding velocity. In a very different way, when the gel slide on a glass surface, with an interlayer of sand grains, the slip occurs as successive rupture events, with a rupture front crossing the whole surface. We can study then the rupture velocity, and in the cases of ruptures faster than the shear wave velocity, we observe a Mach cone of shear waves.

8:00

4aPA3. Two techniques for measurement of flow resistance. Eric C. Mitchell, Anand Swaminathan (Grad. Program in Acoust., Pennsylvania State, P.O. Box 30, State College, PA 16804), Steven L. Garrett, and Matthew E. Poese (Pennsylvania State Univ., State College, PA 16804)

The accurate measurement of flow resistance has many applications in acoustics. In our laboratory, it is particularly important for the characterization of materials used as regenerators in thermoacoustic refrigerators and for the quantification of leakage paths in complex assemblies. This presentation will describe two techniques for flow resistance measurements made at sufficiently low Reynolds numbers that the resistances measured for unidirectional flow are relevant to acoustic flows. One technique uses a "constant current generator" configuration to characterize stacked stainless steel screens and the other uses an Airpot® graphite piston in a glass cylinder to produce a constant pressure difference and accurate flow rate. Both techniques use air at atmospheric pressure as the test fluid. The constant current technique produces results that are consistent to $\pm 3\%$ for stacks of stainless steel screens that vary in thickness from 10 to 50 screens. The Airpot® technique can produce similar accuracy for flow resistances as large as 10^{10} Pa-sec/m^3. [Work supported by the Applied Research Laboratory and the U.S. Department of Energy.]

8:15

4aPA4. Statistical analysis of a characteristic shock formation distance for high-amplitude noise. Michael B. Muhlestein and Kent L. Gee (BYU Dept. of Phys. and Astronomy, N283 ESC, Provo, UT 84602, mimuhle@gmail.com)

Previous research involved investigating a characteristic shock formation distance for Gaussian, finite-amplitude noise propagating in a cylindrical plane wave tube [Muhlestein and Gee, POMA 12, (in press)]. In particular, the evolution of the probability density function of the pressure and the first-order time derivative of pressure along with the skewness of the pressure derivative were experimentally studied. It was concluded that a constant-factor modification to the nonlinear distortion length defined by Gurbatov and Rudenko may yield a suitable characteristic shock formation distance in a statistical sense. Additional Gaussian noise data with a broader frequency range have now been taken, and the effects of boundary layer dispersion considered. Furthermore, noise with other statistical distributions and pressure statistics mimicking high velocity jet noise have been examined. These data are analyzed statistically as before using probability density function estimates and the skewness of the pressure derivative. An additional figure of merit, the characteristic number of shocks per zero crossing, is also examined.

8:30

4aPA5. An investigation into the interaction of a grazing angle broadband spherical audio signal with the dynamically rough air–water interface of shallow flows in rivers and channels. Andrew Nichols, Kirill V. Horoshenkov, Simon J. Tait (School of Eng., Univ. of Bradford, Bradford, West Yorkshire, BD71DP, United Kingdom), and Keith Attenborough (The Open Univ., Milton Keynes, MK7 6AA, United Kingdom)

Laboratory measurements were made of a broadband audio signal transmitted and received at grazing angles over a range of shallow water flow regimes. Synchronous measurements of local surface fluctuation were taken using a thin-wire wave gauge positioned at the point of specular reflection. The first and second statistical moments of the acoustic intensity at the receiver are shown to be directly related to the second statistical moment of the water surface fluctuations. It was hypothesized that this relationship was due to the path-length of the dominant signal fluctuating in direct relation to the interface fluctuations at the specular reflection point. This hypothesis is

corroborated by analysis of the second statistical moment of the "time-of-flight" of the surface-reflected signal, and by direct analysis of the instantaneous water surface position.

8:45

4aPA6. Waveguide sound propagation in a turbulent atmosphere above a statistically rough surface of the ground. Vladimir E. Ostashev (Coop. Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 325 Broadway, Boulder, CO 80305, vladimir.ostashev@colorado.edu), D. Keith Wilson, and Sergey N. Vecherin (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755)

Waveguide sound propagation in a refractive, turbulent atmosphere above a statistically rough surface of the ground is considered. The waveguide is formed between the surface of the ground and turning points of sound waves in the downwind direction or in a nocturnal boundary layer of the atmosphere when the temperature increases with height. Using a modal decomposition of the sound field and the Chernov method, closed-form equations for the coherence function of the sound field are derived. The solution is expressed in terms of the effective turbulence spectrum, which is a linear combination of the 1-D spectra of temperature and wind velocity fluctuations, and a spectrum of the surface roughness. The coherence function can be calculated with equations obtained or by using the effective spectrum in the already existing code for the coherence function of the sound field propagating in a refractive, turbulent atmosphere above a flat ground [D. K. Wilson, V. E. Ostashev, and M. S. Lewis, Waves Rand. Comp. Media, V. **19**, 369–391 (2009)]. The derived effective spectrum is also used to study the relative contributions of atmospheric turbulence and surface roughness to the coherence loss of propagating sound.

9:00

4aPA7. Performance of thermoacoustic device and its thermal contact to a source. Ivan Rodriguez, Orest G. Symko, and Myra Flitcroft (Dept. of Phys. Astronomy, Univ. of Utah, 201 James Fletcher Bldg., 115 South 1400 East, Salt Lake City, UT 84112-0830)

An important application of a thermoacoustic prime mover is in the conversion of heat to electricity when coupled to a sound to electricity converter. In order to achieve maximum sound output, the thermal coupling of the heat source is critical. This was studied here by using (i) a constant heat flow heat source and (ii) a constant temperature difference heat source. As in an electric system where the higher efficiency of power delivery is for a constant voltage source, maximum heat is delivered to the acoustic device from a constant temperature difference heat source. This was investigated on a 1.94 kHz thermoacoustic engine coupled to an acoustic cavity. A shutter located inside the cavity made it possible to have sound on and sound off. In the constant heat flow approach, the shutter technique gave a direct value and measure of heat which was converted to sound. The constant temperature difference approach provided the most heat input for maximum sound output. The temperature profile of cold heat exchanger, hot heat exchanger, and stack was determined by thermocouples.

9:15

4aPA8. Sound propagation in a pipe with dynamically rough boundary. A. Romanova, K. V. Horoshenkov, and S. J. Tait (Dept. of Eng., Design and Technol., Univ. of Bradford, Bradford, BD7 1DP, United Kingdom, a.romashk@gmail.com)

The surface pattern of the water flow in partly filled circular pipe contains information on some key characteristics which are important for a better understanding of the hydraulic energy losses and turbulent processes. Surface pattern variation is a dynamic and nonstationary process which is difficult to measure directly. In this sense, airborne sound waves provide an attractive statistical mean which describes the apparent boundary roughness, its spatial correlation function, and frequency spectrum. These parameters can then be linked to key hydraulic characteristics of the flow. This work presents new experimental setup, which is used to study these characteristics under controlled laboratory conditions and allows for simultaneous measurements of the acoustic field in the pipe and water surface roughness. The acoustic technique makes use of Gaussian pulses which are transmitted in air above the turbulent flow of water over a carefully instrumented section and recorded on an intensity probe. The results obtained for a range of flow regimes illustrate that it is possible to relate unambiguously the variation in the recorded acoustic field to a short-term variance in the water surface roughness and its spectrum. A suitable theoretical foundation based on small perturbation theory is proposed to interpret the obtained data.

9:30

4aPA9. Acoustic radiation torque of arbitrary shaped waves. Glauber Silva (Inst. Fisica, Univ. Fed. Alagoas, Maceio, AL 57072-970, Brasil, glauber@pq.cnpq.br) and Farid Mitri (Los Alamos Natl. Lab. Acoust. & Sensors Technol. Team, MS D429 Los Alamos, NM 87545)

In this work, a general expression of the acoustic radiation torque produced over an object by an arbitrary shaped beam is presented. The object is immersed in an inviscid fluid. To obtain the expression, the stress tensor of the angular momentum is integrated over a farfield virtual sphere, which encloses the object. The incident and scattered acoustic fields are represented through the partial wave expansion in the spherical coordinates. After performing the integration, the radiation torque is given in terms of the beam-shape and the scattering coefficients, which come from the incident and scattered partial wave series, respectively. The method is applied to compute the torque upon a fluid sphere by a vortex (first-order) Bessel beam in both on- and off-axis configurations. It is shown that the torque only arises on absorbing spheres. In this case, the angular acceleration and velocity are obtained from the torque. It is found that the angular acceleration may reverse its direction depending on the wave frequency. In conclusion, the presented theory might be useful for describing the particle dynamics of a sphere subjected to acoustic vortex beams.

9:45

4aPA10. Non-adiabatic geometric phase of elastic waves. Jeremie Boulanger, Nicolas Lebihan (Gipsa-Lab, Grenoble Univ., FRANCE, Jeremie.Boulanger@gipsa-lab.grenoble-inp.fr), Stefan Catheline (ISTERRE, Grenoble Univ., 38000 Grenoble), and Vincent Rossetto (LPMMC, Grenoble Univ., 38000 Grenoble)

We study the transport of elastic waves in a waveguide with helical shape. Polarization exhibits a geometric phase (or Berry phase): The polarization plane rotates along the helix following a geometric rule called parallel transport. Whereas this experiment is similar to the first experimental evidence of a Berry phase, by Tomita and Chiao [Phys. Rev. Lett. **57** (1986)], there is a major difference: The evolution of polarization is not adiabatic. This experiment therefore addresses the universality of the geometric phase beyond the adiabatic regime. We show that properties of the observed geometric phase coincide with the ones predicted by the adiabatic theory. The measured value of the phase is consistent (up to experimental uncertainty) with the theoretical value and no dependency with frequency is observable either.

10:00–10:15 Break

10:15

4aPA11. Utilization of an acoustic tomography array as a large sonic anemometer/thermometer. Sergey N. Vecherin (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH, Sergey.N.Vecherin@usace.army), Vladimir E. Ostashev (Coop. Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 325 Broadway, Boulder, CO 80305), and D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755)

The temperature, wind velocity, and vertical and horizontal kinematic heat fluxes are important characteristics of the atmospheric surface layer. Point measurements of these meteorological parameters are often not representative due to their horizontal variations. For remote sensing of the area-averaged values of these parameters, we suggest using the acoustic tomography array at the Boulder Atmospheric Observatory (BAO). In this approach, the tomography array (with the horizontal size of 80 m by 80 m) is used, in essence, as a large sonic anemometer/thermometer for measurements of the area-averaged, instantaneous values of temperature and wind velocity. The area-averaged horizontal heat flux is then calculated from a time series of the area-averaged temperature and wind velocity. Feasibility of this approach is studied in numerical simulations of the BAO tomography array

with the use of LES fields of temperature and wind velocity. The results obtained show that the area-averaged values of temperature, wind velocity, and horizontal heat flux are reliably reconstructed. Numerical analysis of the LES fields indicates that the area-averaged vertical heat flux might be inferred from the horizontal flux. Preliminary experimental results obtained with the BAO acoustic tomography array show that this remote sensing technique is feasible.

10:30

4aPA12. Triggering of self-excited thermo-acoustic oscillations in a Rijke tube with a premixed laminar flame. Dan Zhao (Aerosp. Eng. Div. Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, zhaodan@ntu.edu.sg)

For a given Rijke tube, self-excited combustion oscillations could be caused by the transient growth of flow disturbances. A premixed laminar flame, anchored to a metal gauze, is considered to investigate the role of non-normality and the resulting transient growth in triggering such oscillations. The unsteady heat release is assumed to be caused by the flame surface variations, which results from the fluctuations of the oncoming flow. The flame is acoustically compact and its presence causes a jump in mean temperature. Coupling the flame model with a Galerkin series expansion of the acoustic waves enables the time evolution of the flow disturbances to be calculated. It was found that the nonlinear model can predict the mode shape and the frequencies of the excited oscillations very well. Moreover, the fundamental mode with the lowest frequency is the easiest one to be excited among all the acoustic modes. Linearizing the model and recasting it into the classical time-lag formulation provide insights on the mode selection and triggering. Finally, to gain insight about the stability behaviors of such non-normal Rijke tube, pseudo-spectra analysis is performed to obtain upper and lower bounds on the transient growth factor.

10:45

4aPA13. Synchronization of ultrasonic thermoacoustic devices. Myra Flitcroft and Orest G. Symko (Dept. of Phys. Astronomy, Univ. of Utah, 201 James Fletcher Bldg., 115 South 1400 East, Salt Lake City, UT 84112-0830)

The development of ultrasonic thermoacoustic devices opens up a new field of applications, in particular where heat can be converted acoustically to electricity in small systems. The power level for applications can be raised by incorporating such devices into array configuration. Since the acoustic devices are self-sustained oscillators, their phase at onset for oscillation is unpredictable; they are triggered on by a random fluctuation. Hence, maximum power output will be achieved by synchronizing the elements of an array. This can be accomplished by suitable coupling between them. Results are presented on the in-phase synchronization of ultrasonic thermoacoustic prime movers at $\sim$23 kHz, acoustically coupled by means of an acoustic cavity. Each engine has an inner volume of $\sim$2.7 mm^3. They were activated by means of separate wire heaters; the working fluid is air at one atmosphere. The demonstration of the synchronization of acoustic engines can be extended to many applications.

11:00

4aPA14. Photoacoustic spectrometer with a calculable cell constant for accurate absorption measurements of atmospheric aerosols and greenhouse gases. Keith A. Gillis (Temperature, Pressure, and Flow Metrology Div. NIST, 100 Bureau Dr., Gaithersburg, MD 20899-8360, keith.gillis@nist.gov) and Joseph T. Hodges (Chemical and Biochemical Reference Data Div. NIST, 100 Bureau Dr., Gaithersburg, MD 20899-8320)

A photoacoustic (PA) spectrometer for absolute optical absorption measurements of aerosols and greenhouse gases in ambient air has been developed. A theoretical analysis of the system in terms of the gas properties, continuous wave laser intensity modulation, and energy transfer relaxation rates will be discussed. The measured and predicted values for the PA system response differ by about 1%. The accuracy of the spectrometer is demonstrated by a probe of the absorption transitions of the *A*-band of O_2 in atmospheric humid air using reference line-shape parameters and accounting for reduced conversion efficiency due to relaxation effects. These transitions also provide a convenient method to monitor the system stability in the field. Observed detection limits are 3.1×10^{-9} W$\cdot$cm$^{-1}\cdot$Hz$^{-1/2}$ for absorption by gases and 1.5×10^{-8} W$\cdot$cm$^{-1}\cdot$Hz$^{-1/2}$ for absorption by soot particles (limited by fluctuations in the aerosol concentration). The sensitivity of the instrument is demonstrated with measurements of the amplified absorption resulting from ultrathin (>5 nm), nonabsorbing coatings on nanoscale soot particles. Applications to real-time monitoring of CO_2 concentration in ambient air and to measurements of the albedo of soot aerosols will be discussed. [Work is supported by NISTs Greenhouse Gas Measurements Program.]

11:15

4aPA15. Computational model for the dynamic stabilization of the Rayleigh-Bénard instability in rectangular enclosures. Randy M. Carbo (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804), Robert W. M. Smith, and Matthew E. Poese (Penn State Univ., PA 16804)

When fluid within a container is heated from the bottom, onset of convection occurs when Rayleigh number, $\frac{g\beta\Delta T\ell^3}{\nu\chi}$, exceeds some critical value. If an acoustic field is imposed on the fluid in the container, the critical Rayleigh number is a strong function of the frequency and amplitude of that acoustic field as noted by Swift and Backhaus [J. Acoust. Soc. Am. **126**(5), 2009]. Results will be reported for a linear model constructed to predict the modified critical Rayleigh number, based on a full field solution of the hydrodynamic equations using the approach of Gelfgat [*J. Comp. Phys.* **156**, 1999]. The spatial portion of the differential equations was solved using the Galerkin method, and the dynamic stability was determined using Floquet analysis. One of the benefits of the approach compared to the averaging methods used by Gershuni and Lyubimov, [*Thermal Vibration Convection* (Wiley, New York, 1998)] is that the parametric stability boundary can also be recovered. This study includes a variety of container aspect ratios, boundary conditions, and Rayleigh numbers ranging from 10^3 to 10^8. [Work supported by the Office of Naval Research and ARL Exploratory and Foundational Research Program.]

11:30

4aPA16. One channel spatio-temporal inversion of acoustic wave in reverberant cavities. Ruppin Matthieu, Catheline Stefan, and Roux Philippe (ISTERRE, Grenoble Univ., 38000 Grenoble, FRANCE, matthieu.ruppin@obs.ujf-grenoble.fr)

It has been recently shown that it was possible to optimally recover the Green functions from a complex wave field despite of a non-isotropic distribution of the noise sources. The method used is based on a particular use of the inverse filter (IF) formalism which is called the passive IF. Based on this formalism, we have investigated the possibility to control the spatio-temporal degrees of freedom in a reverberant cavity for the focusing of waves (active processes). The understanding of this phenomenon can be very useful in a lot of different applications like in acoustical imaging, seismology, or telecommunications. In the present work, the spatio-temporal focalization of ultrasounds in reverberant cavities is studied using medical arrays and water tanks. Through experiments, a complete spatio-temporal inversion is realized to synthesize optimized emitting signals. The result generalizes the focalization control over a spatial vector and during an arbitrary time window.

11:45

4aPA17. Acoustics cavitation in microfluidics for sonoluminescence and sonochemistry. S. W. Ohl, T. Tandiono (Inst. of High Performance Computing, Singapore), D. Ow (Bioprocessing Technol. Inst., Singapore), E. Klaseboer (Inst. of High Performance Computing, Singapore), V. Wong (Bioprocessing Technol. Inst., Singapore), and C. D. Ohl (Nanyang Technol. Univ., Singapore)

Strong ultrasound is applied to a microfluidic channel to generate nonlinear surface waves which entrap bubbles at the gas–liquid interface to form oscillating bubbles. The ultrasound is generated by the piezoelectric transducer on the side of polydimethylsiloxane microchannel. The microchannel is attached to a glass slide through plasma bonding, while the transducer is glued by epoxy for strong coupling. The high speed photography shows that continuous cavitation clusters are formed within the microchannel as gas is injected. As they collapse rapidly, they are able to produce very intense concentration of energy that is able to emit light. This phenomenon is known as sonoluminescence. Previously, sonoluminescence is achieved via a single bubble or multiple bubbles in a bulk liquid. The authors report a realization of sonoluminescence in a microfluidic device. The same oscillating bubbles can also be used as micro-labs. They can trigger chemical reactions that require high temperature and pressure. We achieve the formation of OH radicals in a lab-on-a-chip device. In conclusion, nonequilibrium microbubbles can be induced in a microfluidic system. They oscillate and collapse, and in the process provide a source of energy concentration for the emission of light and the activation of chemical reactions.

THURSDAY MORNING, 3 NOVEMBER 2011 CALIFORNIA, 9:00 A.M. TO 12:00 NOON

Session 4aPP

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

Andrew J. Lotto, Chair
Dept. of Speech, Language, and Hearing Science, Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721-0071

Contributed Papers

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

4aPP1. Psychophysical weights estimated for interaural cues in envelope slopes of amplitude-modulated waveforms. I-Hui Hsieh (Inst. of Cognit. Neurosci., Natl. Central Univ., 300 Jhongda Rd., Taoyuan, Taiwan, ihuihsieh@gmail.com) and Kourosh Saberi (Univ. of California-Irvine, Irvine, CA, saberi@uci.edu)

Current binaural theory contends that the auditory system encodes interaural delays in highpass-filtered complex sounds by phase locking to their slowly modulating envelopes. Spectrotemporal analysis of interaurally time-delayed highpass waveforms reveals the presence of a concomitant interaural level cue which could contribute to lateralization judgments. The current study systematically investigated the relative contribution of time and concomitant level cues carried by positive and negative envelope slopes of modified sinusoidally amplitude-modulated (SAM) high-frequency carriers. Psychophysical thresholds and observer decision weights were measured independently for the positive and negative modulation slopes of the acoustic signal. Decision weights were also measured to determine whether or not interaural delays are uniformly weighted at different temporal cycles of a SAM waveform. We found that lateralization of interaurally delayed SAM waveforms is influenced equally by ITDs in the rise and decay envelope slopes, and not by concomitant ILD cues, and that ITD cues are more heavily weighted in the initial few cycles of the SAM envelope.

4aPP2. Sensitivity to changing characteristics of Gaussian-shaped stimulus distributions in auditory categorization. Sarah C. Sullivan, Johnna A. Tanji, Andrew J. Lotto (Dept. of Speech, Lang., & Hearing Sci. , Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721-0071), and Randy L. Diehl (Univ. of Texas, Austin, TX 78712-0187)

This experiment examined the ability of human listeners to categorize sounds as a function of changing training distribution characteristics. Participants were presented non-speech sounds randomly sampled from two overlapping, Gaussian-like distributions. The sounds consisted of narrow-band noise bursts varying in center frequency. Participants mapped the distributions of sounds onto creatures in a video game receiving visual and auditory feedback about accuracy. The distributions were constant for half of the participants. For the other half, an unsignaled switch to distributions with a new optimal boundary occurred in the middle of the session. By examining obtained category boundaries one can distinguish between 3 possible responses to the switch: (1) adaptive switching resulting in the new optimal boundary being learned; (2) persistence of first learned boundary; and (3) averaging input across the entire session, resulting in an obtained boundary midway between the first and second optimal boundaries. The results demonstrate that most listeners could adaptively switch between distribution conditions with no external signal informing them of such a switch. The results indicate that most listeners are updating possible underlying distributions of input in a remarkably adaptive manner. [Work supported by NIH.]

4aPP3. Word production analysis of native speakers and second language learners by phonemic and categorical verbal fluency test. Keiko Asano (School of Medicine, Juntendo Univ., Hiragagauendai 1-1, Inzai-city, Chiba Pref ZIP270-1695, keiko_asano@sakura.juntendo.ac.jp)

This study investigated how different Japanese, Arabic, and Thai second language learners produce words orally from their native languages aspects of the Verbal Fluency Test. This test, which is in wide spread use in clinical and neuropsychological assessments, consists of two different tasks: phonemic and categorical sections. As a phonemic procedure, the participants were to produce orally as many different words as possible beginning with a certain letter within one minute. The names of specified items within categorical one were to be produced. These clinical field scores are only adopted as quantitative information. However, in this study, the relationship between the phonemic and categorical sections is also analyzed to evaluate the ability of second language learners' word production. It is commonly known that normal and healthy native speakers can produce more words in categorical section than that of the phonemic one while the speakers who are clinically disordered have results that are the reverse. The results show that native speakers produce more categorical words, which were the same results found in normal and healthy people. However among all three second language learners, there is different tendency. Categorical words were produced less frequency with no relevance to the proficiency level. The implication for the function of different brain area activation will also be discussed.

4aPP4. Monotic versus diotic thresholds in an amplitude modulation and quasi-frequency modulation discrimination task. Ewa Borucki, Allison I. Shim, and Bruce G. Berg (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697-5100)

This study investigated monaural and binaural differences in a task traditionally used to investigate the bandwidths of phase sensitivity. Subjects discriminated amplitude modulated (AM) tones and quasi-frequency modulated (QFM) tones presented diotically. An adaptive threshold procedure was used to estimate modulation depth needed to make the discrimination as a function modulation frequency for a 2000-Hz carrier. Threshold functions were often nonmonotonic, with nonmonotonicities observed at

higher-modulation frequencies (above 600 Hz). This is likely due to the effects of cubic difference tones (CDTs) creating spectral cues, yielding lower thresholds. Subjects then completed the task at higher modulation frequencies monotically, demonstrating differences between monotic and diotic thresholds. Within subject thresholds also differed between left ear and right ear presentations. Some subjects yield nonmonotonic thresholds in the monotic condition, even when diotic thresholds were monotonic. For these subjects, it is likely that the CDT interaction is not consistent between the two ears, rendering them an unusable cue when stimuli are presented diotically. Distortion product otoacoustic emissions (DPOAEs) were measured and support the hypothesis that nonmonotonicities found are the effect of a CDT interacting with the low tone of the stimulus.

4aPP5. Converging evidence from behavior and electroencephalography for differences in the storage of streams versus individual sound objects. Lenny A. Varghese (Dept. of Biomedical Eng., 677 Beacon St., Boston, MA 02215), Virginia Best (Dept. of Speech, Lang., and Hearing Sci., Boston, MA 02215), and Barbara G Shinn-Cunningham (Dept. of Biomedical Eng., Boston, MA 02215)

Previous psychophysical experiments from our lab have suggested a difference in the way perceptual sound streams and discrete sound events are stored in short-term memory. We explored differences in brain activity during memory retention of these two kinds of sequences using EEG. Listeners heard a target sequence composed either entirely of short pitched tones (forming a stream) or natural sounds (e.g., a horn or a scream, which are perceptually disconnected), while scalp activity was recorded using EEG. After a 2 s retention period, listeners were required to detect a change in the ordering of a probe sequence (same/different). For a given sequence length, performance for pitched tone sequences was consistently higher than for natural sound sequences, suggesting that memory retention of an auditory stream requires less cognitive effort than retention of a sequence of perceptually disconnected sounds. Retention period alpha (8–12 Hz) oscillatory activity generally increased from pre-target levels in parietal and occipital electrodes for both types of sound sequences, with a trend toward more widespread increases for natural sounds. These results may indicate that alpha activity is related to the amount of cognitive effort required to maintain sound sequences in short-term memory. [Work supported by NSSEFF grant to BGS-C.]

4aPP6. Auditory training effects on auditory steady state responses. Vidal I. Hinojosa and Su-Hyun Jin (Su-Hyun Jin Dept. of Commun. Sci. and Disord., 1 University Station A1100, Austin, TX 78712, vihinojosa@gmail.com)

Human auditory steady-state responses (ASSRs) have been linked to recognition scores in normal hearing and hearing impaired adults (Dimitrijevic *et al.* 2004). By taking this into account ASSRs should improve over time when speech perception improves. Computerized aural rehabilitation programs such as the LACE program by Neurotone have claims of being able to improve speech in noise perception, such as a 2.2 dB improvement using the QuickSin Test as an outcome measure. The main purpose of this study is to answer the question "Can improvements in speech perception be tracked by electrophysiological methods such as ASSR?". For this study, ASSRs and HINT scores will be tracked before and after auditory training in two groups of subjects. The first group will be hearing impaired people who have had hearing aid experience before. The second group consists of hearing impaired people who have just been fit with hearing aids for the first time. ASSR stimulus will be modeled after speech as in the Dimitrijevic 2004 study. By tracking these improvements in speech perception objectively using the ASSR and subjectively using HINT scores, the study will hopefully add validation to objectively testing speech perception using ASSR techniques.

4aPP7. Physiological arousal and laughter acoustics. R. Toby Amoss, Noel B. Martin, and Michael J. Owren (Dept. of Psych., Georgia State Univ., P.O. Box 5010, Atlanta, GA, 30302-5010, tobyamoss@gmail.com)

Laughter is a ubiquitous human phenomenon that has been little investigated scientifically. To examine the relationship between laugher arousal level and acoustic output, bouts of laughter were recorded from undergraduates viewing humorous video clips. These participants provided continuous subjective ratings of funniness while watching the clips, and heart rate (HR) was collected concurrently to provide an objective measure of physiological arousal. As a first approach, comparisons focused on voiced laughter from nine males and eight females. For each individual, one high-amplitude and one low-amplitude laugh bout was identified for which HR could be extracted. Beats per minute increased significantly more with high-amplitude than low-amplitude bouts, an effect that was not likely due to physical exertion or movement artifact. No sex difference was found in the magnitude of HR change for either amplitude condition. Both subjective ratings of funniness and fundamental-frequency measures were significantly higher for higher-amplitude bouts, while harmonic-to-noise ratios trended lower for these sounds. Overall, results are consistent with the intuition that higher physiological arousal in vocalizers is reflected in higher vocal amplitude and faster, potentially less stable vocal-fold vibration in voiced laughter.

4aPP8. Phonetic and acoustic differences in child and adult laughter. Caroline Menezes and Samantha Diaz (Dept. of Health and Rehabilitation Sci., Univ. of Toledo, 2801 W. Bancroft St., Toledo, OH 43606)

This is a preliminary study comparing the acoustic differences in recordings of child and adult spontaneous laughter. Altogether, 100 laughter calls where analyzed from one male and female adult and one male and female child. Results indicate that bout and call durations of children and adult laughter are similar in duration; however, segmental durations show developmental differences. Children show variation between vowel and consonant durations unlike adults. Moreover, child vowels are longer in duration when compared to the adult vowels. Surprisingly, between the adult and child vowels, no difference was observed in mean pitch and mean intensity values. The most prominent difference between child and adult laughter is observed in vowel quality where children's F1 values of laughter vowels are relatively higher than adult's. The consonantal resonance in children is similar to their vowels. However, in adults, the consonant resonant frequencies are much higher than the vowel resonances. Therefore, while children may employ more extreme placements of jaw or tongue they have relatively limited articulatory movement from consonant to vowel when compared to adult laughter. This suggests interesting insights into development of children's speech utterances which need to be explored further.

4aPP9. A modified model for predicting breathiness judgments using partial and noise loudness measures. Mark D. Skowronski, Rahul Shrivastav (Dept. Speech, Lang., and Hearing Sci. , Univ. of Florida, Gainesville, FL 32611, markskow@ufl.edu), and David A. Eddins (Dept. Comm. Sci. and Disord., Univ. S. Florida, Tampa, FL 32620)

Breathiness is one of three classifications, along with roughness and strain, used to characterize disordered voice quality. Breathiness has been measured using a magnitude estimation task and modeled with a power function of perceptual measures [Shrivastav *et al.*, J. Acoust. Soc. Am. **129**, 1605–1615 (2011)]. The experiment was repeated using a matching task which required listeners to match the perceived breathiness of a vowel to a sawtooth-in-noise comparison by adjusting the signal-to-noise ratio of the comparison. The matching task provides a more accurate measure of perception and can potentially accommodate all three dimensions of dysphonic voice quality (breathiness, roughness, and strain) in a single experimental paradigm. Breathiness was modeled with a power function of noise-to-partial loudness ratio. Breathiness judgments (in decibels) varied over about 4 orders of magnitude and exhibited a linear relationship with the log ratio of noise-to-partial loudness. This data was modeled with a logistic function which accounted for breathiness saturation at low levels of noise loudness. Accuracy of the models is discussed as well as sources of variation in the experiment (talker, listener) and strategies for mitigating their effects on modeling accuracy.

4aPP10. A computational model for evaluating the interaural disparities using coincidence detection. Zbynek Bures (College of Polytechnics, Tolsteho 16, 58601 Jihlava, Czech Republic, buresz@vspj.cz)

In mammals, spatial hearing is supported by evaluation of two types of binaural disparities, interaural time difference (ITD) and interaural level difference (ILD), which is performed by auditory neurons in the superior olivary complex. Using a complex auditory model, which includes a binaural evaluation stage, the output of binaural neurons from medial and lateral superior olive is simulated and quantitatively compared with available experimental data. Both binaural disparities are evaluated using detection of coincidence of spikes arriving from the two channels: in case of ITD, the coincidence of two excitatory spikes is detected; in case of ILD, the coincidence of one excitatory and one inhibitory spike is detected. It is shown that

using a physiologically realistic set of parameters, evaluation of both ITD and ILD based on coincidence detection is capable of reproducing the observed neuronal responses. Furthermore, the model is shown to qualitatively reproduce the just-noticeable differences of binaural parameters depending on frequency and intensity. [Work supported by the project "Podpora a individualni rozvoj perspektivnich akademickych pracovniku na VSPJ" at the College Of Polytechnics Jihlava.]

4aPP11. Comparison of a 3 dimension model versus a 2 dimension-axisymmetric finite element model of an occluded ear canal. Guilhem Viallet (Dept. of Mech. Eng., ETS, 1100 Notre-Dame West, Montreal, QC H3C 1K3, Canada, guilhem.viallet.1@ens.etsmtl.ca), Franck Sgard Pr (IRRST, Montreal, QC, H3A 3C2, Canada), and Frédéric Laville (ETS, Montreal, Montreal, QC, HC 1K3, Canada)

Due to low cost and simplicity, earplugs are a widespread solution to prevent the problem of hearing loss in the workplace environment. In practice, earplugs often perceived are being uncomfortable and/or do not always perform as desired. The attenuation, based on a laboratory measurement, is often overestimated compared to *in situ* measurements. The use of a model of an occluded ear canal can help to build an individual measurement system of the attenuation and to develop an earplug with optimized attenuation. It is established that the unoccluded auditory canal can be approximated by a cylinder to predict the interior pressure field up to 6 kHz. A remaining question is whether this approximation holds true for an occluded ear. First, a simplified 2-D-axisymmetric finite element model of an ear canal coupled to a cylindrical earplug is developed. Special emphasis is on the coupling between the earplug and the lateral walls of the auditory canal. Second, a 3-D model of a real ear canal coupled to a cylindrical earplug is developed to examine the limits of the simplified model. Several assumptions about the deformation of the ear canal/earplug system are tested when comparing the sound attenuation provided by both models.

THURSDAY MORNING, 3 NOVEMBER 2011 — PACIFIC SALON 4/5, 9:00 TO 11:30 A.M.

Session 4aSCa

Speech Communication: Forensic Acoustics—On the Leading Edge of the Tidal Wave of Change About to Hit Forensic Science in the US? I

Geoffrey Stewart Morrison, Chair

School of Electrical Engineering, Univ. of New South Wales, Sydney, NSW, 2052, Australia

Chair's Introduction—9:00

Invited Papers

9:05

4aSCa1. Legal standards for the admissibility of expert testimony: Implications of the 2009 National Research Council report on forensic sciences. William C. Thompson (Dept. of Criminology, Law and Society, School of Social Ecology, Univ. of California, Irvine, 5300 Social and Behavioral Sci. Gateway, Irvine, CA 92697-7050)

This talk will review formal legal standards for the admissibility of expert testimony in the United States, focusing on the Daubert standard that is applied in Federal courts and the Frye standard that is applied in several major states (including California and New York). It will also discuss the National Research Council's 2009 critique of judicial "gatekeeping," particularly the NRC's stunning claim that courts have violated their own purported standards by allowing forensic scientists to present scientifically dubious testimony based on inadequately validated methods. It will conclude by providing suggestions for researchers in emerging areas of forensic inquiry, like forensic acoustical science, who contemplate testifying in court.

9:25

4aSCa2. The response to *R v T*. Can forensic acoustics play a leading rôle in a new wave of adoption of the likelihood-ratio framework? Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommun., Univ. of New South Wales, UNSW Sydney, NSW 2052, Australia)

In 2010, the England and Wales Court of Appeals ruled in *R v T* that the likelihood-ratio framework should not be used for the evaluation of evidence except "where there is a firm statistical base." It was not favorable to the use of likelihood ratios even if a firm statistical base does exist. It was, however, willing to accept evidence without a firm statistical base, but on the condition that likelihood ratios are not used. In response, 31 leading forensic scientists published a statement asserting that the likelihood-ratio framework is the logically correct way to evaluate evidence (as a logical framework it is not itself dependent on data or statistical models). This statement has been endorsed by the board of the European Network of Forensic Science Institutes, representing 58 laboratories in 33 countries. Forensic scientists have argued that the ruling is based on misunderstandings of both the likelihood-ratio framework and statistics. The presenter proposes that the magnitude of the response to *R v T* potentially heralds a new wave of adoption of the likelihood-ratio framework by forensic scientists and by the courts (the first wave was DNA in the 1990s) and that forensic acoustics can potentially play a leading rôle.

9:45–10:00 panel-discussion

10:00–10:15 Break

Contributed Papers

10:15

4aSCa3. An introduction to forensic gunshot acoustics. Steven D. Beck, Hirotaka Nakasone, and Kenneth W. Marr (BAE Systems, 6500 Tracor Ln., MS 27-16, Austin, TX 78725, steve.beck@baesystems.com)

Due to the proliferation of audio recording devices in the military, law enforcement, and the civilian community, there has been an increase in the number of recorded gunshot sounds submitted for forensic analysis. A gunshot sound is composed of one or more discrete acoustic transient events. The two primary acoustic events are the muzzle blast (bang) and the ballistic shockwave (crack). The acoustic event characteristics depend on their source generating mechanisms and vary according to the firearm make, model, barrel length, and the specific ammunition characteristics. Forensic gunshot analysis deals with a single recorded shot lasting for a fraction of a second. These acoustic events are usually high intensity, often up to 160 dB SPL, are highly directional, and are often recorded in high distortion environments. Forensic gunshot analysis must take into account variations in the source generation characteristics and the sources of distortion for these recorded acoustic events in order to answer these fundamental forensic questions: Is this event a gunshot? Are two events from the same firearm? Who fired first? To illustrate the complex nature of the analysis, we present the gunshot data collected in a pristine controlled environment and the data collected in a forensic environment.

10:30

4aSCa4. Comparison of human-supervised and fully automatic formant-trajectory measurement for forensic voice comparison. Cuiling Zhang, Felipe Ochoa, Ewald Enzinger, and Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. (Telecom., Univ. of New South Wales, Sydney, New South Wales, Australia; Dept. of Forensic Sci.) Technol., China Criminal Police Univ., Shenyang, China)

Acoustic–phonetic approaches to forensic voice comparison often include analysis of vowel formants. Such methods depend on human-supervised formant extraction, which is often assumed to be reliable and relatively robust to transmission-channel effects, but requires substantial investment of human labor. Fully automatic formant trackers require less human labor but are usually not considered reliable. This study assesses the variability within and between four human experts and compares the results of human-supervised formant measurement with several fully automatic procedures, both on studio-quality recordings and transmission-channel degraded recordings. Measurements are made of the formant trajectories of /iau/ tokens in a database of recordings of 60 female speakers of Chinese. As well as directly comparing the formant-measurements results, the formant measurements are also used as input to likelihood-ratio forensic-voice-comparison systems, and the validity and reliability of each system is empirically assessed.

10:45

4aSCa5. Nasal spectra for forensic voice comparison. Ewald Enzinger (Forensic Voice Comparison Lab., School of Elec. Eng., Telecom., Univ. of New South Wales, Sydney, New South Wales, Australia) and Cuiling Zhang (Univ. of New South Wales, Sydney, New South Wales, Australia)

For features to be effective in forensic voice comparison, they must have relatively low within-speaker variability and relatively high between-speaker variability. An understudied source of features, which potentially meets these criteria is the acoustic spectrum of nasals. Nasals spectra contain poles and zeros dependent upon nasal cavities. The latter are complex static structures which vary from person to person. Theoretically, nasal spectra may therefore have low within-speaker and high between-speaker variabilities. This study evaluates different methods for extracting spectral features (e.g., pole-zero models, all-pole models, and cepstra) and using them as part of a likelihood-ratio forensic-voice-comparison system. The validity and reliability of each system is empirically evaluated using /m/ and /n/ token extracted from a database of voice recordings of 60 female speakers of Chinese.

11:00

4aSCa6. Intra- and inter-speaker variability in duration and spectral properties of English /s/. Colleen Kavanagh (Dept. of Lang. and Linguistic Sci. Univ. of York, Heslington, York YO10 5DD, United Kingdom, cd519@york.ac.uk)

This study investigates the speaker-specificity of acoustic characteristics of the English fricative /s/ and contributes background population statistics for use in forensic speaker comparison work. The intra- and inter-speaker variability in duration and spectral properties of /s/ was investigated in data from 30 young adult male speakers of Cambridge and Leeds English. Read speech was used in the present study to allow direct comparison across speakers. Segment duration was normalized for speaking rate. Spectra were filtered at 4 kHz in order to explore speaker discrimination performance at settings mimicking the bandpass filter effect of telephone transmission. Additional filters were applied at 8, 16, and 22.05 kHz to investigate discrimination with data from various frequency ranges. Spectral measures were calculated from a 40-ms window centered on the midpoint of each token. Although mean values display relatively little inter-speaker variation, the individuals at the extreme high and low ends of the distributions may be the best discriminated, particularly those at the extremes on more than one parameter. Discriminant analyses were conducted to determine the most speaker-specific predictors; relative performance was compared across the four filter conditions. The discriminatory ability of these parameters will also be presented using a likelihood ratio framework.

11:15

4aSCa7. Collecting population statistics: The discriminant power of clicks. Erica Gold (Dept. of Lang. and Linguistic Sci., Univ. of York, Heslington, York Y010 5DD, United Kingdom)

This research gathers population statistics on clicks for use in likelihood ratios (LRs). As reported in Gold and French (2011), clicks have been analyzed by 57% of experts in forensic speaker comparison cases and 18% of experts find them to be useful speaker discriminants. Eight minutes of speech from 100 male speakers of Southern Standard British English were analyzed from the DyVis Database, using categorical annotations of clicks (Wright, 2007). The distribution of click use in subjects is highly skewed with a large majority not clicking. However, the distribution of clicks is highly variable with *non-clickers* ranging from 25–44% of the population depending on the length of the speech sample. The same 100 speakers were also analyzed for click use when speaking with two additional interlocutors. Again the results are highly variable, which suggests the intra- and inter-speaker instability of clicks, the lack of overall robustness, and the accommodation of clicks in speech. This study serves as a beginning point in incorporating previously unreported population statistics into LRs, and specifically examining the potential of including higher order and paralinguistic features in a Bayesian framework. [Research funded by the European Community's Seventh Framework Program (FP7/2007-2013) under grant agreement #238803].

Session 4aSCb

Speech Communication: Acoustics of Speech Production and Acoustic Signal Processing (Poster Session)

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All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSCb1. Short and long diphthongs in Hainan Cham. Ela Thurgood (Dept. of English, California State Univ., Taylor Hall 209, Chico, CA 95929-0830, ethurgood@csuchico.edu)

This paper focuses on the acoustic features in the contrast between two pairs of phonemically distinctive diphthongs in Hainan Cham, /ai/ versus /a:i/ and /au/ versus /a:u/. The data from six native speakers of Hainan Cham show that the overall duration of long diphthongs does not statistically differ from the overall duration of short diphthongs. The differences, instead, lie in the durational differences between onsets and offsets. In Hainan Cham, the long diphthong onsets are longer than the short diphthong onsets, while their offsets are shorter. The transition duration does not differentiate short and long diphthongs. In Hainan Cham, it occupies only 25–34% of the duration of the whole diphthong, short or long. Another acoustic feature examined in diphthongs is the range and rate of F2 change. In Hainan Cham, /ai/ is well differentiated from /a:i/: The short diphthong /ai/ has a greater F2 range of change and a faster F2 rate of change than the long diphthong /a:i/. However, /au/ and /a:u/ have very similar F2 rate of change.

4aSCb2. The Canadian shift in Ontario: Transmission and diffusion. Rebecca V. Roeder (Dept. of English, Univ. of North Carolina at Charlotte, 9201 Univ. City Blvd., Charlotte, NC 28223, rroeder@uncc.edu)

The Canadian Shift, a sound change in progress that is affecting the front lax vowels of Canadian English, was initially characterized as a chain shift. However, recent studies have observed a different pattern of movement over apparent time, requiring an alternative theoretical model to explain the change. Relying on instrumental analysis of data from nearly 100 speakers across five Ontario cities and towns, this paper provides additional observations of parallel shifting across apparent time in these vowels and adopts vowel dispersion theory as a theoretical framework for positing phonetic movement toward a system that is balanced both phonetically and phonologically. This phonetically-based model has primarily been used to explain the relationship between phonological inventories and acoustic space, but the generalizations and principles that are emerging through such research are also useful in the interpretation of observations regarding sound change in progress. Findings also indicate that the geographical progression of the shift corresponds with a cascade or gravity model of diffusion.

4aSCb3. Stimulus and rate effects on central vowels in young adult and aged adult speakers. Phoebe Natzke and Marios Fourakis (Dept. of Com Dis, Univ. of Wisconsin, 1975 Willow Dr., Madison, WI 53706)

The central vowels caret and schwa occur frequently and contribute significantly to the rhythmic (prosodic) characteristics of English (Umeda, J. Acoustic. Soc. Am., 1975). Appropriate prosody contributes to the intelligibility of speech (Klopfenstein, Int. J. Speech Lang. Path., 2009), and there is evidence that it is disturbed in the speech produced by aged adults (Benjamin, Lang. Comm., 1986). The present study explores the effects of speech rate and speech material on these central vowels in the aged population, as compared to a group of young adult speakers. In addition, it contributes to the relatively scarce published research on the acoustics of speech produced by geriatric adults (Zraick *et al*., J. Med. Speech Lang. Path., 2005). The study reports temporal and spectral data of the vowels caret and schwa occurring in a variety of words produced at three different speech rates. Speakers included 10 healthy young speakers and 10 healthy geriatric speakers, with five men and five women in each group. [Work supported by NIH T32 DC 005359.]

4aSCb4. Dispersion and variability of vowels of different inventory sizes. Wai-Sum Lee and Eric Zee (Dept. of CTL, City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon Tong, Hong Kong, w.s.lee@cityu.edu.hk)

This study investigates dispersion and variability in the vowels of systems with different phonemic inventory sizes in three Chinese dialects. Yongding Hakka, Hong Kong Cantonese, and Wenling Wu have three, seven, and eleven vowel phonemes, respectively. Measurements of formant frequencies were obtained through a spectral analysis of speech data from twenty male and twenty female speakers of each dialect. Results show that the size of acoustical vowel space in the F1F2 plane increases with an increase in vowel inventory size. The finding supports the vowel dispersion theory (Lindblom, 1968; Liljencrants and Lindblom, 1972) which claims that the larger the inventory is, the more expanded the vowel space will be. However, the prediction by the theory that variability in vowel formants is inversely related to the vowel inventory size is not supported by the vowel formant data from the three Chinese dialects. In Fant (1966, 1975), female vowels are said to exhibit greater between-category dispersion in the F1F2 plane than male vowels. The observation is supported by the vowel formant data from the three Chinese dialects. However, this is true before vowel normalization, but not after. Lastly, no gender-related patterning of vowel dispersion is observed in the three Chinese dialects.

4aSCb5. Voice onset time production in Singapore English. Priscilla Z. Liu (2981 N Cardell Cir., Tucson, AZ 85712)

This current study investigates the effects of linguistic accommodation through the production of voice onset time (VOT). Six Singaporeans were recorded in three separate conversations that differed by interlocutors: 1) another Singaporean, the researcher, and a non-Singaporean. The two Singapore English dialects, Standard Singapore English (SSE), and Singlish, are investigated. SSE is influenced by British English while Singlish draws from a myriad of languages, "mother tongues" that are spoken in Singapore: Mandarin Chinese, Hokkien, Tamil, and Malay. Thus, Singaporeans were expected to shift production of VOT as motivated by linguistic accommodation to their interlocutors. Previous research indicates that VOT is a valuable acoustic demarcation of phonetic and phonological boundaries. Thus, speakers were found to significantly shift their production of VOT for /p t k/ towards phonetic categories of British English or Malay and Tamil dependent on their audience. (Though the trends for /b d g/ were not significant, the numerical data also shift in the same direction.) Lastly, the paper discusses the implications of these results, that although SSE and Singlish are not entirely discrete and both exist as heavily used dialects of English in Singapore, speakers manipulate VOT in order to accommodate their listeners.

4aSCb6. Effects of consonant context on vowel formant contours in spontaneous and read speech. Michael Kiefte (Sch. Human Commun. Disord., Dalhousie Univ., 5599 Fenwick St., Halifax, N.S. B3H 1R2 Canada, mkiefte@dal.ca) and Terrance M. Nearey (Univ. of AB, Edmonton AB TG6 2E7, Canada)

A database of recordings from 163 speakers from Nova Scotia, Canada was collected with the aim of comparing formant contours between spontaneous and read speech. In the reading task, participants were asked to produce a number of real and nonsense words spanning the inventory of vowels in this dialect in a variety of consonant contexts. In the second part of the reading task, speakers were asked to read 20 sentences from the TIMIT database. These latter recordings were used to assist in the training of the force-alignment system which used to segment the recordings into phonemes from a text transcript. In addition to the reading task, speakers also provided a monologue on a topic of their choice. These recordings were screened for disfluencies, noise, and disruptions, manually segmented into breathgroups, and then transcribed. Stressed vowels in /CVC/ contexts, where C corresponds to plosives, were sampled from both the read and spontaneous speech. Formants were tracked and measured automatically, and an analysis similar to that of Broad and Clermont [J. Acoust. Soc. Am., **81**, 155–165 1987] was performed in which consonant effects on formant transitions are treated as additive effects. [Work supported by SSHRC.]

4aSCb7. Effects of speaking rate, sentential position, and coda voicing on formant frequency. Keelan Evanini (Educational Testing Service, Rosedale Rd. MS-R11, Princeton, NJ 08541, kevanini@ets.org) and Eon-Suk Ko (Univ. at Buffalo, Buffalo, NY 14260)

This study examines the effects of speaking rate, sentential position, and coda voicing on formant frequency values in English. Some previous studies have found gestural undershoot for formant target values in words with shorter durations, e.g., Lindblom (1963), although other studies have shown little to no effect of duration, e.g., Gay (1978), and the effects of the other factors are less-studied. This study examines formant frequencies of three different vowels (/i/, /e/, and /ae/) in CVC words containing both voiced and voiceless codas produced in three different sentence positions (initial, medial, and final) and three different speaking rates (slow, habitual, and fast). In total, seven speakers (five female and two male) of the Northern dialect of American English produced 1295 tokens, and vowel formant measurements were extracted at 1/3 of the duration of each vowel token. Separate linear regression analyses of the three vowels for the male and female speakers show that F1 and F2 target values do not vary systematically with vowel duration. In many cases, however, sentence position and coda voicing do have significant effects: in general, F1 and F2 values are more peripheral before voiced codas and in sentence-initial position.

4aSCb8. Acoustic contrastivity in conversational and loud speech. Yunjung Kim (Dept. of Commun. Sci. and Disord., Louisiana State Univ., Baton Rouge, LA 70803)

Despite frequent clinical observations of improved speech intelligibility following high vocal intensity training in speakers with dysarthria, the mechanism by which loud speech results in increased speech intelligibility is little understood. Prior research has reported conflicting acoustic results of articulatory modifications in loud speech conditions such as changes in vowel durations, acoustic vowel space, and F2 transition duration/extent. More interestingly, the impact of an overall increase in amplitude of speech signals on perceptual judgment of speech intelligibility is unclear, especially when the entire speech signal is amplified as compared to the amplification of selected phonetic events (see Kim and Kuo, in press). This presentation focuses on the change of *relative* contrastivity within utterances that were produced at both conversational and loud levels to better understand the underlying mechanism of enhanced speech intelligibility secondary to greater vocal intensity. In this presentation, data on the ratio of vowel durations (long versus short), formant structures of vowels (tense versus lax), as well as the ratio of syllable intensity (stressed vs unstressed) will be compared between conversational and loud speech conditions produced by young adult speakers.

4aSCb9. Timing differences in read speech and spontaneous conversation: English, Japanese, Korean, and Mandarin. Dan Brenner (Univ. of Arizona, Dept. of Linguist., Douglass Bldg. 200E, Tucson, AZ 85721, dbrenner@email.arizona.edu)

Timing and rhythm in language reflect broad auditory properties of languages to which listeners acclimate very early on in acquisition (Nassi *et al*., 1998, 2000), and are heavily implicated in the differentiation of speech segmentation strategies cross-linguistically (Cutler and Butterfield, 1992; Otake *et al*., 1993; Saffran *et al*., 1996; Kim *et al*., 2008). The rhythmic properties of everyday conversational speech (as compared to read speech or motherese), however, are not well understood. The present work employs several measures developed to summarize rhythmic differences such as %V vs. δC (Ramus *et al*., 1999) and C-rPVI versus V-nPVI (Grabe and Low, 2002) in order to study the timing variation found in English, Japanese, Korean, and Mandarin (unrelated languages varying in purported "rhythm class") comparing rhythmic measures during careful read speech and in spontaneous casual conversation. This reveals the effect of highly variable conversational speech on the timing behavior of rhythmically diverse languages.

4aSCb10. Phonetic imitation in contexts of stimulus-directed and non-stimulus-directed attention. Jennifer Abel, Molly Babel, and Alexis Black (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada)

Research suggests that phonetic imitation is an automatic and subconscious process, but it is clearly a behavior that is variable across participants and conditions. This experiment explores how a participant's amount and type of attention to the speech signal moderates their amount and type of imitation. Six paired conditions varied the activities of participants (all native speakers of English) while listening to the model talker (a female speaker of North American English): blocked exposure (no instructions for listening block) vs immediate shadowing; math task vs picture-drawing task; and word-memorization task vs talker-description task. In all conditions, participants' baseline productions of the stimuli list were recorded prior to exposure to the model talker. Early analyses of whole word duration suggest participants are more likely to imitate the model when their attention is not directed toward the stimuli (i.e., no particular redirection or redirection through a picture-drawing task) than when their attention is stimulus-directed (i.e., being instructed to memorize the words or describe the talker). Further analyses of other acoustic parameters which may reveal imitative behavior are currently underway and include measures of vowel duration, vowel spectra, and f0.

4aSCb11. Visual speech influences on interactive speech alignment. James W. Dias, Therasa Cook, and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, 900 University Ave., Riverside, CA 92521)

Speech alignment describes the unconscious tendency of humans to produce speech that shares the acoustical characteristics of a perceived speech signal. Participants have been found to align more when interacting in full view of their partner than when conversing only through sound [J. W. Dias and L. D. Rosenblum, J. Acoust. Soc. Am. **128**, 2458 (2010)]. However, this previous study did not address what specific visual information enhances speech alignment. The present study evaluates whether being able to see visual *speech* gestures enhances alignment in a conversational setting. Pairs of participants performed an interactive search task, for which they were required to utter nine key words multiple times. Participants performed the task, while interacting face-to-face, or with visibility of the mouth and throat occluded using a small sound-permeable screen. Participants' utterances of the key words were recorded before, during, and after the interaction. Alignment was evaluated by naïve raters in an AXB matching task. In a second experiment, participants performed the task in a background of talker-babble noise. Preliminary results indicate that alignment increases with visibility of the mouth. The findings are consistent with evidence that alignment occurs to amodal, gestural properties of perceived speech.

4aSCb12. Examining the voice bar. Sean A. Fulop (Dept. of Linguist. PB92, Calif. State Univ., 5245 N Backer Ave., Fresno, CA 93740, sfulop@csufresno.edu) and Sandra Disner (Dept. of Linguist., Univ. of Southern Calif., Los Angeles, CA 90089)

In a spectrogram of a human vowel sound, it is possible to observe the formant resonances which define the vowel auditorily. It is usually also possible to observe an emphasized frequency below F1, which has often been

called the *voice bar*. Although recognition of the voice bar dates back to 19th century phonetics, it has never been the subject of a specific investigation. As a result, the nature and origin of the voice bar remain mysterious. Recent work on voice source synthesis [C. d'Alessandro *et al.*, *eNTERFACE 2005 Proceedings*, pp. 52–61] has explained the appearance of an emphasized frequency in the neighborhood of 200 Hz–simply, it results from the frequency peak of the radiated source spectrum. Yet many speech scientists continue to ignore the voice bar, even to the point of denying its reality. Measurements of the voice bar in a number of different speakers and languages will be clearly shown in this paper, using reassigned spectrograms and linear prediction spectral estimates. The voice bar has confused phoneticians for over a century, and has often confounded efforts to measure F1. The proper recognition of the voice bar can begin with this preliminary study.

4aSCb13. Sub-phonemic correlates of gender and regional identity in California. Grant L. McGuire (Dept. of Linguist., UC Santa Cruz, 1156 High St. Santa Cruz, CA 95064, gmcguir1@ucsc.edu), Angela R. Aiello (Dept. of Communicative Disord., San Jose St. U., 1 Washington St., San Jose, CA 95192), Jackie L. De Leon, Tariq El-Gabalawy, Lauren Negrete, and Kasondra Vanpykeren-Gerth (Dept. of Linguist., UC Santa Cruz, 1156 High St. Santa Cruz, CA 95064)

Given that the Northern and Southern California have large metropolitan areas geographically and culturally separated from each other, it is to be expected that each is developing a unique linguistic identity. Despite a handful of ethnographic studies showing otherwise (e.g., Hall-Lew 2009), the West has generally been lumped into a single dialect region (Labov *et al.* 2006). This paper presents data showing sub-phonemic differences between the regions that break along gender lines. Vowel productions from 14 (female = 8) Northern Californians (NCs) and 15 (female = 8) Southern Californians (SCs) were analyzed for regional differences in normalized vowel quality, voice quality (spectral tilt), pitch, and duration. No major differences in vowel quality were found. However, interactions were found between region and gender for duration and voice quality. Specifically, NC females had significantly longer word durations than NC males, with no difference between genders for SC. For voice quality, H1-H2 and H1-A3 measures both demonstrated significant differences between males and females for SC, with female voices being breathier, but with no differences for NC. Currently, a perception experiment is underway to determine if listeners can use these differences to categorize voices by region.

4aSCb14. Liquids as syllable peaks: Preconsonantal laterals in closed syllables of American English. Onna A. Nelson (Dept. of Linguist., UC Santa Barbara, South Hall, Santa Barbara, CA 93106, oanleson@umail.ucsb.edu)

Liquids occur in all syllable positions in English and may behave as syllable peaks or nuclei (Proctor, 2009). Previous work has examined syllabic liquids in open syllables like *little* and *doctor*, in the onset of closed syllables like *prayed* and *played* (Price, 1980) and has established that rhotics are syllabic in certain closed syllables like *bird*, *church*, and *learn*. However, little work has investigated whether laterals can serve as syllable peaks in preconsonantal position. This study examines potential syllabic liquids in closed and open syllables in the Santa Barbara Corpus of Spoken American English (Du Bois *et al.*, 2000, 2003, 2004, 2005), focusing on CVlC syllables, such as *bulk*, *filled*, and *help*, which are structurally similar to the aforementioned contexts containing syllabic rhotics. Vowel and lateral duration and intensity are measured to determine whether these laterals display properties associated with syllabicity (Price, 1980). Additionally, the first and second formants of the vowel and lateral are measured at 10 ms intervals to examine the vowel-like behavior of the liquids (Gick, 2002). Further influencing factors are considered, including morphology, surrounding vowel quality, place and manner of surrounding consonants, intonation, and other prosodic elements to determine the environments in which lateral syllabicity occurs.

4aSCb15. The production of Spanish–English code-switching. Page E. Piccinini (Dept. of Linguist., Univ. of California San Diego, 9500 Gilman D., La Jolla, CA 92093-0108)

It is generally assumed that in code-switching (CS) switches between two languages are categorical, however, recent research suggests that the phonologies involved in CS are merged and bilinguals must actively suppress one language when encoding in the other. Thus, it was hypothesized that CS does not take place abruptly but that cues before the point of language change are also present. This hypothesis is tested with a corpus of Spanish-English CS examining word-initial voiceless stop VOT and the vowel in the discourse marker "like." English VOTs at CS boundaries were shorter, or more "Spanish-like," than in monolingual utterances. Preliminary results suggest Spanish VOTs at CS boundaries were shorter than in monolingual utterances, thus even more Spanish-like than monolingual Spanish utterances. The vowel of "like" in English utterances was more monophthongal and had a lower final F2 as compared to "like" in Spanish utterances. At CS boundaries, "like" began similarly to the language preceding the token and ended similarly to the language following it. For example, in a "English-like-Spanishnn" utterance, initial F2 measurements were more English-like but final measurements more Spanish-like. These results suggest code-switching boundaries are not categorical, but an area where phonologies of both languages affect productions.

4aSCb16. Dynamic differences in the production of diphthongs by French–English bilingual children. Vincent Chanethom (Dept. of Linguist., New York Univ., 10 Washington Pl., New York, NY 10003)

This study examines the cross-linguistic phonetic interactions in the production of diphthongs by French–English bilingual children. Tautosyllabic vowel-glide combinations in English and French have different phonological statuses. This combination corresponds to a single segment (i.e., a diphthong) in English, but two separate segments (i.e., vowel+glide) in French. Using a picture-naming experiment, the study aims to investigate (1) whether English diphthongs (e.g., /aɪ/ as in bye) and French tautosyllabic vowel-glide combinations (e.g. /aj/ as in baille 'yawn') have different phonetic implementations and, if so, (2) whether bilingual children maintain two separate categories. Diphthongs were recorded for six monolingual speakers of French and English, and four 6-7 year-old bilingual French-English speakers living in the US. To best capture the dynamic properties of diphthongs, the curves corresponding to F1 and F2 trajectories were submitted to statistical comparisons using the Smoothing Spline ANOVA. The cross-linguistic comparisons from the adult monolingual data indicate distinct phonetic properties for the two categories. The child bilingual data, on the other hand, show variation from one child to another as a function of amount of input. Children who attend English-only schools show greater degrees of language interference than those who attend French–English bilingual schools.

4aSCb17. Vowel undershoot in production of English tense and lax vowels by Mandarin and American speakers. Chung-Lin Yang (Dept. of Linguist., Indiana Univ., Bloomington, IN 47405, cy1@indiana.edu)

Vowel undershoot (Lindblom, 1963), an effect where the articulatory gesture fails to reach the target due to the following contrary gesture (de Jong, 2004), is found to be particularly prominent (lowered F1 and shortened duration) in polysyllables (Lindblom, 1968; Moon and Lindblom, 1989). The current study investigates Mandarin and American speakers' production of English tense-lax vowels /i/-/ɪ/ and /eɪ/-/ɛ/, and examines undershoot in monosyllabic and disyllabic words. Three Mandarin and three American speakers were recorded. Fifteen target vowels were embedded in a voiced stop-V-voiceless stop context. All disyllabic words were first-syllable stressed. Carrier sentences were of variable length to create a natural-speech-like context. F1, F2, vowel duration and utterance duration were measured. The results show that Americans did show a significant distinction between /i/-/ɪ/ but tended to merge the formants of /eɪ/-/ɛ/ in disyllabic words, which was partly due to the coarticulation with the following syllable. They also demonstrated undershoot effect in /eɪ/-/ɛ/ but not much in /i/-/ɪ/. Mandarin speakers, however, could not make a significant tense-lax distinction, and showed formant undershoot in disyllabic words, especially tense vowels. One possible account for this effect is the influence from L1. The issue of Mandarin vowel inventory is discussed.

4aSCb18. Clear speech production by nonnative English speakers. Jenna Silver Luque and Ann R. Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, SLPJenna@gmail.com)

Clear speech is an intelligibility-enhancing mode of communication often used when speakers have trouble being understood. Previous work has

established that both native and non-native listeners can receive a clear speech perception benefit, though possibly to differing degrees (Bradlow and Bent, 2002). Few studies have looked at whether nonnative talkers can induce this clear speech benefit (e.g., Smiljanic and Bradlow, 2007; Rogers *et al*., 2010). The current study examined English clear and conversational speech by nonnative speakers from three language backgrounds (Japanese, Portuguese, and Turkish) and two proficiency levels to determine their effect on the inducement of a clear speech benefit. Native English listeners repeated back semantically anomalous sentences. The signal to noise ratio was adjusted to the level at which they could correctly repeat 50% of the words using an adaptive test similar to the Hearing in Noise Test (Nilsson *et al*., 1993). The results suggest that the speaker's native language may play a role in the size of the induced clear speech benefit independently of proficiency level. Additionally, accent ratings indicated a dissociation of intelligibility and accentedness. These results are consistent with the notion that variability in intelligibility is subject to language-specific knowledge by both the talker and the listener.

4aSCb19. Voice quality and pitch contrast in non-native Korean. Seung-Eun Chang (Dept. of East Asian Lang. and Cultures, Univ. of California, Berkeley, 3413 Dwinelle, Berkeley, CA 94720, sechang71@berkeley.edu)

In this study, I compare the effects of linguistic experience on voice quality (H1-H2) and fundamental frequency (f0) in Korean stops among native and non-native Korean speakers. Native speakers of Chinese, English, Korean, and Spanish produced Korean words in a /CVC/ context, and H1-H2 and f0 of the initial stops in each set of materials were measured. Korean and Chinese speakers showed creakiness (smaller H1-H2) for Korean tense stops and breathiness (larger H1-H2) for lenis and aspirated stops, whereas English and Spanish speakers showed relatively larger H1-H2 for all stops. For f0 values, Korean and Chinese speakers displayed a lower f0 for lenis, an intermediate f0 for tense, and a higher f0 for aspirated stops. For English speakers, however, lenis and tense stops were merged in the lower f0 region, and aspirated stops showed a higher f0. In Spanish speakers, tense and aspirated stops merged in the higher f0 region, and lenis stops showed a lower f0. These results demonstrate a strong effect of linguistic experience on voice quality and f0: speakers of Chinese were more accurate in replicating Korean stops than were speakers of English or Spanish, languages that lack phonemic voice quality and tone contrasts.

4aSCb20. Acoustic features of English sentences produced by native and non-native speakers. Yu-Fu Chen, Chang Liu, and Su-Hyun Jin (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX 78712)

Fundamental frequency (f0) contours and power envelopes of English sentences produced by young English-, Chinese-, and Korean-native speakers were measured. Sentences were selected from Hearing in Noise Test (HINT) and recorded from non-native speakers whose US residency within five years. Preliminary results showed that the f0 contours of non-native speakers were similar to those of native speakers, but with greater variation over the time, suggesting that non-native speakers may be able to follow native speakers' vocal pitch contour, but with higher temporal variability. Power envelopes of the sentences will be measured and the relationship between the acoustic features of speech and speech intelligibility will be discussed as well.

4aSCb21. Native English speakers learning German as a second language: Devoicing of word-final voiced stop targets. Bruce L. Smith and Elizabeth A. Peterson (Dept. of Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84107, bruce.smith@hsc.utah.edu)

In contrast to German and other languages that devoice underlying word-final, voiced obstruent targets, English has a surface contrast between voiced and voiceless obstruents. The present study investigated the issue of what occurs when native speakers of American English, in an early stage of learning German as a second language, produce word-final voiced and voiceless stop targets in German versus English. The fact that the underlying voicing contrast in German is reflected orthographically (e.g., "Tod" versus "tot") could make it difficult for native speakers of English to learn to devoice German word-final, voiced targets. The findings indicate that many of the 12 native English learners of German who were studied showed a tendency toward devoicing voiced targets in German relative to their productions of orthographically-similar words in English (e.g., "toad" and "tote"). In general, their partial devoicing in German (relative to their English productions) occurred due to a combination of producing somewhat shorter vowels before voiced consonant targets, reducing glottal pulsing during the closure of voiced consonant targets and/or shortening voiceless consonant closure durations. Subjects who produced more characteristically "voiced" consonants when speaking English (e.g., longer preceding vowel durations, etc.) tended to devoice German final stops to a lesser extent.

4aSCb22. Temporal characteristics of child-directed speech. Eon-Suk Ko (Dept. of Linguist., and Dept. of Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY 14260, eonsukko@buffalo.edu) and Melanie Soderstrom (Univ. of Manitoba, Winnipeg, MB R3T 2N2)

Child-directed speech (CDS) is produced with a slower tempo compared to adult-directed speech (ADS). Yet the characterization of CDS as simply slowly spoken speech masks a number of underlying subtleties. We investigated temporal characteristics of CDS as a function of speech register based on a highly controlled set of elicited data. six sentence forms containing five monosyllabic words were read several times in declarative and question intonation with three focus conditions in CDS and ADS. After an evaluation of the data through a perceptual rating task, 2301 sentences produced by one mother and five theater students were segmented at the word-level using forced-alignment tools (Yuan and Liberman, 2008). We found strong effects of the CDS register on duration across the entire sentence. Additionally, elongation in CDS applied even to the syllables without an explicit focal accent and to function words. Our data also demonstrated a highly consistent ratio of the final syllable to the sentence duration both in CDS and ADS across all subjects. These results suggest that the slow speaking rate in CDS cannot be attributed to any single effect such as the exaggerated utterance-final lengthening (Church *et al*., 2005) or effects of lexical categories (Swanson *et al*., 1992).

4aSCb23. Developmental courses of infants' articulations estimated by acoustic-to-articulatory inversions. Hiroki Oohashi, Hama Watanabe, and Gentaro Taga (Graduate school of Education, The Univ. of Tokyo, Tokyo 113-0033 Japan)

Knowledge about the actual movements of articulators and their developmental changes in early childhood is crucial to a better understanding of the relationship between speech productions and perceptions. In this study, we attempted to recover the shapes of articulators from the acoustic properties of infants' vocalizations by using a variable linear articulatory model (Boe, 1999). We randomly selected 30 samples of 5 Japanese vowels at three age groups (8, 24, and 50 months) from the NTT Japanese infants speech database (Amano *et al*., 2002) and those pronounced by adults. We modeled shapes of articulators and size of the vocal tract according to Maeda (1990) and synthesized vowels by forward transformation from the vocal tract cross-sectional area function to its acoustics. We performed inverse estimations of articulatory parameters from acoustic properties by using a pseudo-inverse of the Jacobian matrix. Statistical analysis of the estimated articulatory parameters revealed that the shapes of articulators during vowel pronunciations became closer to those of adults over developmental courses. Developmental courses of some articulators represent the u-shaped curve (e.g., tongue shape and tongue tip of /a/) and those of others represent the linear curve (e.g., tongue position of /o/).

4aSCb24. The acquisition of voiceless sibilant fricatives in children speaking Mandarin Chinese. Fangfang Li (Dept. of Psych., Univ. of Lethbridge, 4401 Univ. Dr. Lethbridge, AB, Canada, T1K 3M4)

The current study aims to describe Mandarin-speaking children's acquisition of voiceless sibilant fricatives, /s/, /ʂ/, and /ɕ/, as assessed by acoustics. Forty children, aged 2–5, participated in a word repetition task. The stimuli were fricative-initial words that are familiar to children. Children's speech sound productions were recorded and analyzed spectrally. Two acoustic parameters were obtained: the centroid frequency calculated over the middle 40-ms slice of the fricative noise spectrum and onset F2 frequency, the second formant frequency taken at the onset of the vowel following target fricatives. Centroid frequency indexes *where* the major lingual constriction is made in the oral cavity and is inversely related to the length of the front resonating cavity during the articulation of voiceless sibilant fricatives. Onset F2 frequency indexes *how* the major constriction is made and is sensitive to the lingual posture during constriction. These two parameters have been demonstrated to capture adults' fricative distinctions successfully. The results indicated an early separation between /ʂ/ and the other two fricatives in the centroid dimension, and an early separation between /ɕ/ and other two fricatives in the onset F2 dimension, The results suggested that children gradually implement their motor control in different articulation/acoustic dimensions.

4aSCb25. The development of neutral tone in Mandarin-speaking children. Jie Yang and Barbara Davis (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 2504 Whitis Ave., Austin, TX 78712, Babs@mail.utexas.edu)

Besides the four citation tones in Mandarin stressed syllables, neutral tone usually occurs in unstressed syllables. The fundamental frequency (F0) contour and height of neutral tone are determined by the preceding tones. Neutral tone was also considered to have lower intensity and shorter duration compared to citation tones (Cao, 1986; Chao, 1968). Li and Thompson (1977) suggested that neutral tone was not fully acquired by Mandarin-speaking children of age 2. However, the acoustic characteristics of neutral tone in the extended period of acquisition were not explored. The present study compared acoustic realizations of neutral tone in children's production with adults. Eight 5-yr-old and eight 8-yr-old mono-lingual Mandarin-speaking children and young adults participated. Bi-syllabic target words containing neutral tone in the second syllable was elicited by picture-naming tasks. F0, duration and intensity of neutral tone syllables were measured. The ratio of these acoustic parameters between the first and second syllable was calculated. Results indicated that 5-yr-old children started to produce F0 contour and height of neutral tone according to the preceding tone.

4aSCb26. Lexical effects in the production of emotional speech. Tatiana Kryuchkova and Benjamin V. Tucker (Dept. of Linguist., Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB, T6G 2E7, Canada, tatiana.kryuchkova@ualberta.ca)

Emotion in speech production has been shown to correlate well with fundamental frequency (F0), intensity, and duration [B. Zupan *et al*., J. Commun. Disord., **42**, 1–17 (2009)]. Studies in non-emotional speech have shown effects of lexical predictors on speech production (e.g., neighborhood density [B. Munson and N. P. Solomon. J. Speech, Lang. Hear. Res., **47**, 1048–1058 (2004)], and lexical frequency [R. H. Baayen *et al*., in *Proceedings of the Annual Meeting of the Chicago Linguistic Society* (2007), Vol. 43, pp. 1–29]. In the present study, we investigate the effects of lexical predictors in the production of emotional speech. Two professional male actors recorded 260 isolated words. Three emotional states were analyzed: neutral, anger, and joy. Measures of word frequency, morphological family size, number of synonyms and homophones, and ratings for danger and usefulness [L. Wurm, Psychon. Bull. Rev., **14**, 1218–1225 (2007)] were used as statistical predictors in modeling, using linear mixed-effects regression. Following previous literature, both speakers use F0, intensity, and duration to portray emotion. Lexical predictors such as word frequency, morphological family size, and danger ratings were found to significantly predict mean F0, mean intensity, and word duration across emotional types for both speakers.

4aSCb27. Executive abilities for spoken-word commands: Inhibiting conflicting responses in voice-tone classification by adolescents and adults. Blas Espinoza-Varas (Commun. Sci. Disord., OU Health Sci. Ctr., 1200 N. Stonewall Av., Oklahoma City, OK 73117), Hyunsook Jang (Hallym Univ., Chuncheon, South Korea), Caleb Lack (Univ. Central Oklahoma, Edmond, OK 73034), and Beatriz Luna (Western Psychiatric Inst. Clinic, Univ. Pittsburgh Medical Ctr., Pittsburgh, PA 15213)

The inhibition of responses prompted by conflicting laterality cues and response-mapping rules was studied while participants classified the voice-tone of word commands. Announced by the cue words /left/ or /right/, target word commands for impeding or instigating actions (e.g., /quit/ or /go/), spoken in stern or lenient tone, stimulated the left or right ear. Each trial presented a cue followed by a target (e.g., left-quit), and participants classified the target voice tone as lenient or stern with a left or right (or the reverse) response for impeding (or instigating) commands. Within a trial block, laterality-conflict conditions presented impeding or instigating words only and, on each trial, the cue and ear side were congruent or in conflict with the correct response side. Response-mapping conflict conditions presented impeding and instigating words within each trial block, and correct classification required adhering to the respective mapping. Without conflict, error percent was larger with instigating than impeding commands, and in adolescents than adults. With instigating commands, laterality conflict increased the errors in both groups but only in adolescents with impeding commands. Errors increased further with laterality and mapping conflict, especially in adolescents. The adolescent response inhibition is inferior to the adult. [Funded by ABMRF.]

4aSCb28. Normalized recognition of speech and audio events. Mark A. Hasegawa-Johnson, Jui-Ting Huang, Sarah King, and Xi Zhou (ECE Dept., Univ. of Illinois. Urbana, IL 61801)

An invariant feature is a nonlinear projection whose output shows less intra-class variability than its input. In machine learning, invariant features may be given *a priori*, on the basis of scientific knowledge, or they may be learned using feature selection algorithms. In the task of acoustic feature extraction for automatic speech recognition, for example, a candidate for *a priori* invariance is provided by the theory of phonological distinctive features, which specifies that any given distinctive feature should correspond to a fixed acoustic correlate (a fixed classification boundary between positive and negative examples), regardless of context. A learned invariance might, instead, project each phoneme into a high-dimensional Gaussian mixture supervector space, and in the high-dimensional space, learn an inter-phoneme distance metric that minimizes the distances among examples of any given phoneme. Results are available for both tasks, but it is not easy to compare them: learned invariance outperforms *a priori* invariance for some task definitions, and underperforms for other task definitions. As future work, we propose that the a priori invariance might be used to regularize a learned invariance projection.

4aSCb29. An automatic method for determining phonetic boundary for continuous speech utterances in an open source multi-language audio/video database. Montri Karnjanadecha and Stephen A. Zahorian (Dept. of Elec. and Comput. Eng., Binghamton Univ., P.O. Box 6000, Binghamton, NY 13902-6000)

Nine hundred video clips (approximately 30 h in each of English, Mandarin, and Russian) have been collected from Internet sources such as youtube.com and rutube.ru. This multi-language audio/video database has been orthographically transcribed by human listeners with time markers at the sentence level. However, the aim is to provide this database to the public with high accuracy time markers at the phonetic level, which will greatly increase the value of the database. This paper describes an approach to achieving high accuracy automatic phonetic labeling based on a Hidden Markov Model speech recognizer. This automatic method was developed due to the great length of time and tediousness of performing this task using only human listeners. One major challenge for the automatic method was that the audio data consists of spontaneous speech with unconstrained topics and the speech was spoken under various acoustic conditions. The approach begins with a well-trained acoustic model for each language. The acoustic model is then adapted to each passage and finally the phonetic labeling of the passage is determined. Comparison of the automatically determined phone time markers with those obtained by human listeners, for a subset of the speech materials, shows the accuracy of the automatic method.

4aSCb30. Spectral amplitude nonlinearities for improved noise robustness of spectral features for use in automatic speech recognition. Stephen Zahorian (Dept. of Elec. and Comput. Eng., Binghamton Univ., 4400 Vestal Parkway East, Binghamton, NY 13902, zahorian@binghamton.edu) and Brian Wong (Binghamton Univ., Binghamton, NY 13902)

Auditory models for outer periphery processing include a sigmoid shaped nonlinearity that is even more compressed than standard logarithmic scaling at very low and very high amplitudes. In some studies done at Carnegie Mellon University, it has been shown that this compressive nonlinearity is the most important aspect of the Seneff auditory model in terms of improving accuracy of automatic speech recognition in the presence of noise. However, in this previous work, the nonlinearity was trained for each frequency band of the Mel frequency cepstrum coefficients thus making it impractical to incorporate in automatic speech recognition systems. In the current study, a compressive nonlinearity is parametrically represented and constructed without

training, to allow various degrees of steepness and "rounding" of corners for low and high amplitudes. Using this nonlinearity, experimental results for various noise conditions, and with mismatches in noise between training and test data, were obtained for phone recognition using the TIMIT and NTIMIT databases. The implications of the results are that a fixed compressive nonlinearity can be used to improve automatic speech recognition robustness with respect to mismatches between training and test data.

4aSCb31. Restoration of intermittent speech signal relying on auditory perceptual capability. Mitsunori Mizumachi (Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, Fukuoka 804-8550, Japan, mizumach@ecs.kyutech.ac.jp) and Toshiharu Horiuchi (KDDI R&D Labs. Inc., 2-1-15 Ohara, Fujimino, Saitama 356-8502, Japan)

Speech signals are frequently transmitted through the IP network. In such situation, a packet loss causes a temporal break in receiving speech signals. The intermittent speech signals make speech communication uncomfortable. However, the intermittent speech signal can be heard smoothly under certain conditions, because the brain reconstructs the missing speech packet unconsciously. This auditory illusory phenomenon, that is, the phonemic restoration effect, occurs under severe noisy conditions [Miller and Licklider, 1950]. In short, a speech signal with a heavy background noise, of which signal-to-noise ratio (SNR) is less than 0 dB, overcomes the intermittency in receiving speech signals, although the noisy speech signal makes us uncomfortable in speech communication. In this paper, we propose a speech signal restoration scheme with auxiliary signal processing to positively enhance our phonemic restoration capabilities under less noisy condition. First, we discuss the characteristics of the background noises. Reducing noisiness and enhancing the phonemic restoration effect should be compatible with each other for achieving comfortable speech communication. Second, we propose to predict and reconstruct the principal parts of the missing signal components from peripheral information. Finally, synergistic contribution is discussed between the above considerations. [Work partially supported by NEDO, Japan.]

4aSCb32. Acoustic evidence for protracted development of monosyllabic Mandarin tone production by Taiwanese children. Xin Yu (Dept. of Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Columbus, Ohio 43212), Jing Yang (The Ohio State Univ., Columbus, OH 43210), and Puisan Wong (The Ohio State Univ., Columbus, OH 43212)

The current study examines the acoustic characteristics of monosyllabic Mandarin tones produced by 3-y-old children growing up in Taiwan. Four hundred monosyllabic tone productions were collected from 11 adults and 10 children and were judged by 5 Mandarin-speaking adults to determine tone accuracy. Seven acoustic parameters strongly associated with Mandarin tone perception were measured in these productions and compared among children's and adults correct and incorrect productions. The findings indicate that children do not produce the high level tone with fundamental frequencies (f0) as high or level as adults. Children's rising, falling, and dipping tones do not reach f0 ranges as low as adults. These results are largely consistent with the findings in our previous acoustic study on the monosyllabic Mandarin tones produced by 3-y-old children growing up in the U.S. Taken together, 3-y-old Mandarin-speaking children growing up in the U.S. and Taiwan do not produce adult-like monosyllabic Mandarin tones. Even children's productions in which the tonal targets are correctly perceived by adults are acoustically different than the adult forms. Children demonstrate more difficulties producing low f0 targets. The findings provide acoustic evidence to support a much more protracted process for Mandarin lexical tone acquisition than most studies have suggested.

THURSDAY MORNING, 3 NOVEMBER 2011 — PACIFIC SALON 3, 8:00 TO 10:15 A.M.

Session 4aUWa

Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Characterization of Noise Radiation and Quieting Techniques for Unmanned Underwater Vehicles

Christopher Barber, Cochair
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Jason D. Holmes, Cochair
Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02138

Chair's Introduction—8:00

Invited Papers

8:05

4aUWa1. An international standard for the measurement of vessel underwater noise. Michael Bahtiarian (Noise Control Eng., Inc., 799 Middlesex Turnpike Billerica, MA 01821, mikeb@noise-control.com)

The development of a new international commercial standard for "Underwater Noise Measurement of Ships" started in April 2011. This effort follows on an American National Standard, ANSI/Atomic Security Agency (ASA) S12.64-2009/Part 1, Quantities and Procedures for Description and Measurement of Underwater Sound from Ships, Part 1; General Requirements. Currently, no international standards exist for performing underwater noise measurements of ships. For many years, the field of underwater noise from ships has been the exclusive specialty of the Navy. However, non-navy vessels are looking to be just as quiet so that they can perform better science. Green ships are being conceived in order to have less emission into the ocean. The goal of the project is to develop an International Organization for Standardization (ISO) standard for the measurement of underwater noise levels of ships using commercial technology. One aim is that the standard would be applicable to any open ocean site in the world and not require traveling to special acoustic test ranges. The committee' scope of work will include neither regulatory actions nor the development of any underwater noise level but will address both deep and shallow water sites. This presentation provides an update of the committee work to date and outreach to the acoustical community.

8:30

4aUWa2. Modeling noise from underwater vehicles. Raymond Fischer (Noise Control Eng., Inc., 799 Middlesex Turnpike Billerica, MA 01821, rayf@noise-control.com)

To predict and control the radiated noise from underwater vehicles can be a complicated task. As opposed to a surface ship with a hull surrounding the machinery sources, these vehicles have multiple acoustic sources in direct contact with water. Any surrounding hull-type structure, excited by the source, then also radiates into the acoustic media. This paper discusses these potential sources, e.g., propulsors, motors, pumps, controllers, high-pressure air system, and electronics. The transmission path from the sources to the ocean also needs to be defined and understood. Airborne, fluidborne, and structureborne paths exist for most vehicles and they are often cross-coupled. These vehicles can also have exotic hull materials whose radiation characteristics are quite different from the standard metallic hulls. Accurate models can identify which sources and paths are important and over which frequency ranges. To reduce the signature in an optimal manner and diminish the adverse impacts on space, payload, and cost of typical treatments, one must have a good understanding of the process of modeling the radiated noise of a vehicle. This paper discusses these critical factors.

8:55

4aUWa3. Underwater gliders as acoustic receiving platforms. Georges A. Dossot, James H. Miller, Gopu R. Potty, Kristy A. Moore (Univ. of Rhode Island, Dept. of Ocean Eng., Narragansett, RI), Scott Glenn (Rutgers Univ., NB, NJ), and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02882)

Acoustic data were collected on a single hydrophone towed by a Webb Slocum glider deployed by Rutgers University, during the shallow water experiment (SW06), on the continental shelf, of New Jersey. The geometry of the experiment provided for adequate recording of the 224 and 400 Hz tomography sources. A follow-up study of the New Jersey Tuckerton Field Station provided a rudimentary noise analysis showing the glider's capabilities as an acoustic receiving platform. The glider's saw-tooth glide profile allows for vertical sampling of the water column with periodic surfaces for GPS fixes and data transfer via satellite phone. The glider provides a low-noise and low-speed platform, potentially enabling detection of low level signals. [Work sponsored by the Office of Naval Research.]

9:20

4aUWa4. Application of boundary layer suction for reducing hydrophone sensing noise. Craig N. Dolder (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, dolder@utexas.edu), Meagan A. Villanueva (Dept. of Aerosp. Eng. and Eng. Mech., The Univ. of Texas at Austin, 1 University Station C0600, Austin, TX), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029), and Charles E. Tinney (Dept. of Aerosp. Eng. and Eng. Mech., The Univ. of Texas at Austin, 1 University Station C0600, Austin, TX)

One of the leading noise sources for hydrophone arrays on moving vessels is hydrodynamic noise which results from the presence of turbulent boundary layers. This ongoing experimental study explores the impact boundary layer suction has on reducing the hydrodynamic pressure fluctuations and thereby increasing the signal to noise ratio for sensing. A custom hydrophone array is used to capture the pressure signatures with and without boundary layer suction under 2D fully developed turbulent boundary layers with momentum thickness Reynolds numbers ranging from 2000–4000. The first generation suction device shows a reduction in noise of up to 50% with moderate suction intensities. The current focus is on observing the effect suction has on the boundary layer velocity field using both single point laser Doppler anemometry and 2D particle image velocimetry. This information will provide insight into how the hydrodynamic structures are being removed and will provide a basis for the development of future optimized suction devices.

Contributed Papers

9:45

4aUWa5. Radiated noise measurements in a harbor environment using a vertical array of omnidirectional hydrophones. Brian Fowler (Grad. Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, bef5000@arl.psu.edu) and Christopher Barber (Penn State Univ., P.O. Box 30, State College, PA 16804)

Measurement of the radiated noise of a ship or submerged vessel in shallow water is complicated by the presence of multiple surface and bottom reflections, requiring that environmental effects on propagation either be minimized in the measurement or accounted for in the result, or perhaps both. In a harbor environment, there are additional sources of reflection and noise that may degrade the ability to obtain meaningful measurements. The combination of multiple reflected paths, high background noise, and a possible inability to assume far-field behavior due to a shortened range present a significant challenge to acquiring high confidence measurements of the radiated acoustic field. This work presents preliminary results from a radiated noise measurement test conducted at the U.S. Navy's Acoustic Research Detachment in Bayview, Idaho during summer 2010. A line array of 14 equispaced omnidirectional hydrophones was deployed from a barge tied up adjacent to a moored test vessel to obtain radiated noise measurements. A series of test signals was also transmitted through a calibrated acoustic source deployed at various depths in the harbor to evaluate the effectiveness of vertical line array measurements in minimizing reflected path contributions and improving signal-to-noise ratio. Preliminary results and conclusions are presented. [Work sponsored by the Office of Naval Research, Code 331.]

10:00

4aUWa6. Review of methods for making radiated noise measurements of submerged vessels. Christopher Barber (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, cbarber@psu.edu)

There are a variety of challenges associated with measuring the radiated noise of surface ships and submerged vessels in order to obtain a measurement and environment-independent parameter such as the far-field equivalent source level. While measurements of large ships and naval vessels have routinely been conducted at deep water test sites in what approximates a far field measurement in a free-field environment, open ocean, fixed range measurements are not always practical for smaller vessels and particularly for autonomous or unmanned undersea vehicles. This presentation provides a brief review of several existing methodologies and examines some of the specific challenges associated with obtaining high quality estimates of vessel radiated noise associated with various methods and measurement scenarios.

Session 4aUWb

Underwater Acoustics: Autonomous Underwater Vehicle Navigation

Brian T. Hefner, Chair
Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

10:30

4aUWb1. Navigation and sonar applications of an acoustical spiral wave front beacon. Benjamin R. Dzikowicz (Naval Res. Lab., Physical Acoust. Branch Code 7136, 4555 Overlook Ave. SW, Washington, DC 20375, benjamin.dzikowicz@nrl.navy.mil), Brian T. Hefner (Univ. of Washington, Seattle, WA 98105), and Robert A. Leasko (Naval Surface Warfare Ctr., Panama City, FL 32407)

A spiral wave front beacon consists of a transducer whose phase depends on the azimuth at which it is received and a reference transducer whose phase does not. [J. Acoust. Soc. Am. **129**:(6), 2011]. A spiral wave front can be produced by using a cylindrical transducer whose radius advances by one wavelength over one revolution, or by phasing a circular array of elements out of phase such that they generate a spiral wave front. Several experiments using both of these types of beacons are carried out at the Navy's Dodge Pond facility in Connecticut. This facility provides a range of environments where the robustness of the signals in multipath and reverberation can be tested. Navigation experiments are carried out using a remotely operated USV equipped with a hydrophone and a data acquisition system which triggers upon receipt of an incoming signal. The vehicle is also equipped with a differential global positioning system (DGPS) receiver to determine its exact position. Several types of outgoing signals are employed. The beacons are also tested for use in passive and active sonar applications where the phase advance between the transducers gives an indication of direction. [Work supported by the Office of Naval Research.]

10:45

4aUWb2. A robust phase gradient bearing estimation algorithm for a tri-axis cross-dipole acoustic array, with application to a long range autonomous underwater vehicle homing and tracking system. Carmen Lucas, Garry Heard, Nicos Pelavas, and Richard Fleming (Defence RD Canada - Atlantic, P.O. Box 1012, Dartmouth, NS, Canada B2Y 3Z7, carmen.lucas@drdc-rddc.gc.ca)

A bearing estimation algorithm was developed as part of a long range acoustic bearing (LRAB) homing system implemented on an autonomous underwater vehicle (AUV). A tri-axis cross-dipole acoustic array with seven digital hydrophones was developed and mounted in the AUV for the homing system. The Phase Gradient algorithm was implemented on the vehicle's Acoustic Homing and Localization System processor and run in real-time. The algorithm was designed to estimate the bearing and elevation angles to a continuous wave (CW) signal from a beacon source. The algorithm directly estimates the three Cartesian components of the incoming signal wave-vector from estimated cross-spectra between the hydrophone elements. The algorithm is robust against hydrophone failure, and every hydrophone in the array is used to estimate each component of the wave-vector. In this paper, the theoretical development of the Phase Gradient algorithm is presented, as well as the results from real applications of the algorithm as part of an AUV homing and tracking system.

11:00

4aUWb3. Linear drift error for cornerstone autonomous underwater vehicle. Nicos Pelavas, Garry J. Heard, Carmen E. Lucas, and Derek Clark (Defence Res. Development Canada Atlantic, PO Box 1012 Dartmouth, NS, Canada B2Y 3Z7, nicos.pelavas@drdc-rddc.gc.ca)

Autonomous underwater vehicles (AUVs) have a promising future in their use of collecting bathymetric data in remote regions of the Arctic Ocean. In accordance with the United Nations Convention on the Law of the Sea, Defence Research & Development Canada Atlantic has partnered with Natural Resources Canada (NRCan) and Department of Fisheries and Oceans to use AUVs in support of Canada's Arctic claim. In this article, we investigate the AUV linear drift error that accumulates as a result of a misalignment between the Inertial Navigation Unit and the Doppler Velocity Log. Data collected during the 2010 Cornerstone Arctic field trial is used to quantify the linear drift error associated with one of the AUVs. The linear drift error was determined to be 0.67% to stern and 0.21% to starboard, and this result was then applied to correct the track for the AUV survey mission.

11:15

4aUWb4. Modeling a spiral wave front source in an ocean environment. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105 USA) and Benjamin R. Dzikowicz (Physical Acoust. Branch, Code 7136, Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375)

A spiral wave front source generates a pressure field which has a phase that depends on the azimuthal angle at which it is measured [J. Acoust. Soc. Am. **129**(6), (2011)]. This type of source can be used in conjunction with a reference source to form a navigation beacon. A remote receiver can determine the direction to the beacon from the phase difference between the pulses transmitted from each of the sources. To determine the accuracy of this navigation technique, it is necessary to model the output of the spiral wave front source in ocean environment. To this end, the spiral wave front analogue of the acoustic point source is examined and is shown to be related to the point source through a simple transformation. This makes it possible to transform the point source solution in a particular ocean environment into the solution for a spiral source in the same environment. This transformation is applied to simple cases, such as reflection from the sea surface, as well as to the more general case of propagation in a horizontally stratified waveguide. [Work supported by the Office of Naval Research.]

Session 4aUWc

Underwater Acoustics: Acoustic Propagation Modeling

Matthew A. Dzieciuch, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

8:00

4aUWc1. Recent developments in the seismo–acoustic parabolic equation. Michael D. Collins (Naval Res. Lab., Stennis Space Ctr., MS 39529, mike.collins@nrlssc.navy.mil)

The development of the seismo–acoustic parabolic equation is currently proceeding in several directions. Since the stability of the parabolic equation is an issue for problems involving relatively thin elastic layers, special rational approximations are being designed for problems involving ice cover. Range dependence has previously been treated with approaches based on single scattering, energy conservation, and coordinate changes that are of limited use for certain cases. For test problems involving gradual range dependence (a convenient reference solution is available in this limit), promising results have been obtained with a single-scattering approximation that conserves quantities across a vertical interface in a mean sense. One of the advantages of this approach is that its simple physical interpretation facilitates generalizing to different cases, such as problems involving sloping fluid–solid interfaces and anisotropic and poro-elastic layers. [Work sponsored by the Office of Naval Research.]

8:15

4aUWc2. Three-dimensional underwater sound propagation using a split-step Padé parabolic equation solution. Jon M. Collis (Colorado School of Mines, 1500 Illinois St., Golden, CO 80401)

The majority of current three-dimensional (3-D) parabolic equation propagation model development work has been focused on implementations. Recent attention has been on the modeling of internal wave fields and propagation in the presence of strong internal tides and eddies. State of the art solutions are experimental water-tank comparisons against azimuthal schemes [Sturm *et al*., JASA **113**] and split-step Fourier based propagation models [Lin *et al*., JASA **126**]. A current thrust is to establish 3-D benchmarks for propagation simulations in range-dependent environments where effects due to generic bottom features are present. In this paper, a 3-D extension to current 2-D split-step Padé solutions is developed and benchmarked. Transverse and depth operator discretizations are performed using a Galerkin method. Cartesian versus cylindrical coordinate systems are discussed as is the nature of the acoustic source in either geometry.

8:30

4aUWc3. Propagation of coupled modes in three dimensions. Ronald F. Pannatoni (540 Mark Dowdle Rd., Franklin, NC 28734, elliptic@alum.mit.edu)

A time-harmonic acoustic field in an ocean can be represented in terms of local normal modes. This representation may include leaky modes to account for radiation into the bottom. A system of coupled elliptic equations governs the amplitudes of these modes. If there is a primary horizontal direction of propagation, it may be possible to approximate these equations by parabolic equations. The resulting system of coupled parabolic equations can be integrated in the direction of propagation only if a certain linear transformation is nonsingular. This constraint limits the size of the system and the amount of coupling among the modes that are consistent with the parabolic approximations. A method for integrating the coupled parabolic equations numerically is discussed, and it is applied to a problem of three-dimensional propagation and scattering around a conical seamount in a deep ocean.

8:45

4aUWc4. A three-dimension propagation model using stepwise coupled modes. Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

A propagation model has been developed that is applicable to shallow-water waveguides characterized by three-dimensional inhomogeneities which induce horizontal refraction and mode coupling. A normal-mode approach is chosen for this work because the field decomposition into modal amplitudes provides insight into the effects of the environment on the acoustic field. The model is based on the stepwise coupled-mode technique implemented with the single-scatter approximation [R. B. Evans, J. Acoust. Soc. Am. **74**, 188–195 (1983)]. The stepwise technique discretizes a range-dependent environment into a series of range-independent segments. The single-scatter solution is obtained by treating each pair of segments as an independent problem, thus neglecting the higher-order terms resulting from multiple scattering at other interfaces. The field is propagated using the parabolic equation in cylindrical coordinates. Thus, at each range step, horizontal refraction is accounted for in the angular direction and mode coupling is included in the radial direction. Examples illustrating the effects of horizontal refraction and mode coupling will be presented. [Work supported by ONR.]

9:00

4aUWc5. Discrete transparent boundary conditions for parabolic equations. Ronald F. Pannatoni (540 Mark Dowdle Rd., Franklin, NC 28734, elliptic@alum.mit.edu)

There are simple algorithms for constructing transparent boundary conditions (TBCs) for a partial discretization of the basic parabolic equation that is known as a "semi-discrete" parabolic equation. This equation and some of these algorithms are reviewed. Solutions of a semi-discrete parabolic equation in a long rectangular strip subject to TBCs at the long edges of the strip are then considered. These solutions can be computed accurately and efficiently with a pseudospectral method that is based on expansions in Chebyshev polynomials. It is beneficial to combine this method with a conventional split-step FFT solution of a parabolic equation subject to Neumann boundary conditions at the long edges of the strip. This hybrid approach will be called the "decomposition method." It is demonstrated in a computation of radiation modes from the termination of a truncated nonlinear internal gravity wave duct in a shallow water area.

9:15

4aUWc6. Equation, describing temporal evolution of the sound pulse in horizontal plane in shallow water. Boris Katsnelson (Voronezh State Univ., Universitetskaya Sq. 1, Voronezh 394006, Russia, katz@phys.vsu.ru)

It is shown that for comparatively narrow-band pulses in shallow water sound propagation, it is possible to introduce modal pulses, corresponding to variation (or evolution) of space-time dependence of the amplitude of separate waveguide modes. In ray approximation, in the presence of

perturbation depending on horizontal coordinates, these signals propagate along different trajectories in horizontal plane (space-time horizontal rays) depending on mode number and frequency. In the paper equation, describing evolution of amplitude of modal pulses as a function of time in horizontal plane outside the ray approximation is obtained. This equation can be considered as extension of well-known parabolic equation in horizontal plane. Examples of solutions for some shallow water models are shown; applicability of this equation is discussed.

9:30

4aUWc7. Seismo-acoustic propagation near thin and low-shear speed ocean bottom sediments. Jon M. Collis (Colorado School of Mines, 1500 Illinois St., Golden, CO 80401), Adam M. Metzler (Rensselaer Polytechnic Inst., Troy, NY 12180), Paden Reitz, and Rezwanur Rahman (Colorado School of Mines, Golden, CO 80401)

Accurate and efficient parabolic equation solutions exist for complex propagation environments featuring elastic and porous elastic sediment types. Because of numerical stability issues, areas of concern have been with thin and low-shear wave speed sediments. In certain situations, both of these common features of the seafloor can make obtaining an accurate solution difficult. At low frequencies, layers of this type can be treated as a massive interface between the water and higher-shear speed sediment basement layers. To satisfy interface conditions across the layer, Rayleigh jump conditions are imposed [F. Gilbert, Ann. Geofisica **XL**, 1211 (1997)]. This approach is only valid for a single layer, but is able to handle shear wave speeds as they tend to zero. In this talk, a massive elastic interface parabolic equation implementation is benchmarked along with classical bottom treatments to quantify the effects of ocean acoustic propagation over thin sediment layers. It is demonstrated that in certain situations, it is sufficient to consider the thin layer as part of adjacent, thicker layers.

9:45

4aUWc8. Robust computation of acoustic normal modes in attenuating ocean waveguides. Thomas J. Hayward and Roger M. Oba (Naval Res. Lab., Washington, DC 20375)

The normal mode representation [Jensen *et al.*, *Computational Ocean Acoustics*, AIP Press, 1997] provides for the mathematical analysis and numerical computation of acoustic fields in a range-independent ocean waveguide. Existing algorithms [e.g., M. B. Porter and E. L. Reiss, J. Acoust. Soc. Am. **77**, 1760] provide for efficient computation of the mode eigenvalues and eigenfunctions for narrow-band acoustic fields. However, for extensive computations involving a large set of environmental parameter values or acoustic frequencies, reliability issues, such as eigenvalues omitted in the calculation, have been noted in the case of attenuating media. In this work, a computational method is presented that computes the normal modes by approximate solution of the mode depth-dependence equation on a discrete computational grid, using a selected discrete basis. Empirical evidence of the robustness of the method is provided by comparisons with established numerical benchmarks and by examining the acoustic parameter dependence of the mode spectra over thousands of parameter values. Theoretical support for the reliability of the computation is then discussed. [Work supported by the Office of Naval Research.]

10:00–10:15 Break

10:15

4aUWc9. An approach to computing acoustic wave propagation in shallow water. Cathy Ann Clark (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841)

A shallow water mode solution is presented, which approximates the water column sound speed variation using isovelocity layers. Within a layer, two fundamental solutions are seen to satisfy the separated depth-dependent wave equation and complex analogs apply in evanescent regions. Solutions which satisfy an upper boundary condition are extended continuously through isovelocity layers to the bottom, matching functions and derivatives at layer boundaries. The value of the Wronskian for the Green's function thus obtained is used to locate the eigenvalues of the normal modes comprising the propagating field. Acoustic mechanisms which are dominant in shallow water such as forward scattering and range dependence are incorporated as matrix multiplications to implement horizontal coupling between modes. Agreement between the shallow water approach and a benchmark deep water mode solution is shown for a number of shallow environments.

10:30

4aUWc10. Sonar propagation modeling using hybrid Gaussian beams in spherical/time coordinates. Sean M. Reilly (Dept. of Ocean Eng., Univ. of Rhode Island, Bay Campus, Narragansett, RI 02882)

This paper defines a new undersea acoustic transmission loss model that is optimized for real-time, sonar simulation/stimulation systems in littoral environments. The ray solutions to the eikonal equation are computed in latitude, longitude, and altitude coordinates to match wide-area environmental databases. Hybrid Gaussian beam techniques for transmission loss calculation are used to extend the applicability of ray theory to lower frequency regimes. Numerical integration of the wave equation is performed in the time dimension to support broadband signal modeling. This 3-D approach also supports out-of-plane reflection from the ocean bottom. This paper derives the eikonal solution from first principles to ensure a complete understanding of the coordinate system's impact.

10:45

4aUWc11. Underwater acoustic propagation in Arctic environments. Christie A. O'Hara (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD 20723) and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Developments in underwater acoustic modeling for the Arctic have been limited due to the complicated nature of the polar extreme. In Arctic regions, the sound speed minimum occurs at or near the ice-covered surface. The upward refracting sound speed profile causes any long-range propagation to repeatedly interact with the ice cover. This paper presents an overview of the derivation of a 2-D normal mode propagation model to the range-independent wave equation for a source and receiver in the water column. To more accurately model the Arctic ocean acoustic environment, we consider a modified Pekeris waveguide. As with the Pekeris waveguide, the bottom is considered as an infinite fluid halfspace and the top is considered to be a fluid layer of finite thickness overlying the water column. Results will be benchmarked against a fluid parabolic equation solution. An application is discussed as a means to track marine mammals using received signals.

11:00

4aUWc12. Perth-Bermuda revisited again: Global adiabatic mode parabolic equation results. Kevin D. Heaney and Richard L. Campbell (Ocean Acoust. and Instrumentation Systems, Inc., 11006 Clara Barton Dr. Fairfax Station, VA 22039, oceansound04@yahoo.com)

In 1960, a set of explosives were detonated off the coast of Perth Australia, and multi-pulse receptions were recorded from moored hydrophones off of Bermuda. The Perth-Bermuda experiment demonstrated the capability of trans oceanic acoustic propagation. The two-pulse arrival, separated by approximately 25 s, was explained by Heaney, *et al*. (JASA, **90**(5), 2586–2594) in terms of two disparate paths: a northern path refracting off islands in the southern Indian Ocean and the other refracting off the shelf-break on the coast of Brazil. In this paper a global adiabatic mode-parabolic equation hybrid model is used to compute the multi mode, broadband full-field pressure response from Perth to Bermuda. The PE field demonstrates that Bermuda is in the acoustic shadow of the refracted geodesics, yet observations of arrivals were made. PE results demonstrate that two significant scattered paths, one to the north passing by the Cape of Good Hope and one to the south, passing by the coast of Brazil yield strong strong arrivals, confirming the results of Heaney *et al*.

11:15

4aUWc13. Acoustic modes in a curved internal wave duct. Ying-Tsong Lin (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, ytlin@whoi.edu), Kara G. McMahon, William L. Siegmann (Rensselaer Poly. Inst., Troy, NY 12180), and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Observations and numerical simulations have shown that nonlinear internal waves in continental shelf and shelfbreak regions can form 3-D acoustic ducts. The strength of ducting depends on the size of internal waves, the width of the gap between waves and the curvature of the wave front, and also on the acoustic frequency and the vertical mode number. It has been seen in numerical simulations and simplified ray theory that for a given internal wave structure and a given frequency, higher vertical modes are easier being trapped in a curved internal wave duct. Also, the number of the lowest mode trapped between curved waves increases as the frequency goes up. In this talk, a 3-D normal mode theory is employed to analyze these observed characteristics. The analysis is carried out in a cylindrical coordinates, and two types of horizontal modes are found: whispering-gallery modes and full bouncing modes. Both types of modes can be described by Bessel functions, and the asymptotic formulas can be used in some limiting cases. [Work supported by the ONR.]

11:30

4aUWc14. Detection performance modeling and measurements for convergence zone (CZ) propagation in deep water. Kevin D. Heaney, Richard L. Campbell, James J. Murray (Ocean Acoust. and Instrumentation Systems, Inc. 11006 Clara Barton Dr. Fairfax Station, VA 22039, oceansound04@yahoo.com), Gerald L. D'Spain (Scripps Inst. of Oceanogr., UCSD), and Arthur B. Baggeroer (Massacheussets Inst. of Technol., Cambridge MA)

A novel parabolic equation algorithm in the C programming language has been developed based upon the RAM model. This model (CRAM) permits modeling of the full-field sonar equation to estimate towed array performance of the detection of quiet targets in a dynamic environment with both environmental range dependence and source/receiver kinematics. During an experiment in the northern Philippine Sea in 2009, a ship towing Penn State's Five-Octave Research Array (FORA) was towed at various depths in a star pattern about the station-keeping source ship, thereby sampling the first CZ in range, depth, and azimuth. Measurements and modeling of the CZ arrivals will be compared. A simple detection processor is applied to the CZ receptions. Comparison of passive ASW performance modeling results with measurements will be made. One of the primary science issues in the statistics associated with probabilistic detection is the time between independent samples, or the sample-to-sample correlation. This will be evaluated from the data for a portion of the test where the receiver was towed in an arch-fixing the source–receiver range for several hours. [Work supported by ONR.]

11:45

4aUWc15. An investigation of the effects of rough seas and bubble injections on high frequency propagation using a parabolic equation method. Joseph M. Senne, Aijun Song (CEOE, Univ. of Delaware, Robinson Hall, Newark, DE 19716, sennejm@udel.edu), Kevin Smith (Naval Postgrad. School, Monterey, CA 93943), and Mohsen Badiey (Univ. of Delaware, Newark, DE 19716)

High frequency underwater acoustic transmissions (>10 kHz) are heavily influenced by scattering from both rough surfaces and bubbles. These interactions are recorded through the prevalence of micro multi-paths in observed data. To study these scattering effects, a rough-surface variant of the Monterey Miami parabolic equation model was combined with a hydrodynamic surface model that produces non-linear waves along with depth- and range-dependent bubble distributions. Parabolic equation setup parameters were taken from collected environmental data, while a wave-rider buoy was used for time-evolving sea surface generation. Bubble plume densities were calculated using surface white-cap distributions along with a bubble evolution scheme. Comparisons of the simulated results are made against collected acoustic data for calm and rough sea states [Work supported by ONR Code 322OA.]

THURSDAY AFTERNOON, 3 NOVEMBER 2011 — SUNSET, 1:00 TO 5:25 P.M.

Session 4pAAa

Architectural Acoustics, Noise, and Committee on Standards: Networking in Soundscapes—Establishing a Worldwide Collaboration II

Gary W. Siebein, Cochair
Dept. of Architecture, Univ. of Florida, 231 Arch, P.O. Box 115702, Gainesville, FL 32611

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square, Ste. 103, Vernon, CT 06066

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Getting it together—Interdisciplinary sound environment research. Frans Mossberg (Dept. of Cultural Studies, Lund Univ., Biskopsgat 7, 22100 Lund, Sweden, fransmo@glocalnet.net)

The Sound Environment Center at Lund university is an interdisciplinary center created to coordinate research on sound and soundscape issues and is known to be the first of its kind worldwide. Ranging from acoustics to medicine, psychology, and cognitive sciences, as well as humanities like musicology and linguistics, soundscape research adresses many interdependent areas and touches upon health as well as philosophical, aesthetic, and technical issues. To get a holistic comprehension, these perspectives need to be synchronized. Therefore, the center has an interdisciplinary board and a mission to study sound environments from multidisciplinary perspectives. Focus lies on research and contact between researchers. The center has external funding for larger research collaborations on topics such as teachers voice strain and rooms acoustics, health effects of combined exposure to noise and airborne particles, cognition, and sound exposure. In addition to initiating research projects, the center arranges symposiums adressing topics such as Noise and health, Seductive Sounds, Operational Sounds, Dangerous Sounds' and Sound, Cognition and Learning. Further topics have been Sound Design, Sounds and Silence for Mental Recreation, Teachers Voice Comfort, and recently Wind Turbine Noise. The symposiums facilitates cross disciplinary contacts and discussions, many of them producing published papers and reports.

1:25

4pAAa2. Soundscape ecology: A worldwide network. Catherine Guastavino (McGill Univ., School of Information Studies and CIRMMT, 3661 Peel, H3X 1X1, Montreal, QC, Canada, catherine.guastavino@mcgill.ca) and Bryan C. Pijanowski (Purdue Univ., West Lafayette, IN 47906)

The overarching objective of our network is to bring together acousticians, cognitive psychologists, ecologists, and creative artists to integrate how they study and perceive soundscapes and use this knowledge to help shape a research agenda for the conservation of soundscapes. Many natural soundscapes are being threatened from various directions, e.g., habitat destruction, climate change, invasive species. This project aims at recording and documenting soundscapes in remote locations and identifying conceptualizations of these soundscapes across different cultures and disciplines. The network will help to (1) foster open communication between different disciplines and communities about soundscapes; (2) coordinate soundscape monitoring sites where acoustic data are being collected long-term; (3) develop a common vocabulary, long-term monitoring standards, and metadata standards for acoustic data for use by ecologists; (4) increase awareness of this new field among ecologists and social scientists; and (5) increase public awareness of the importance of their acoustic connection to nature. This project on the soundscapes of natural ecosystems is a logical complement to research underway at the European level on urban soundscapes (COST action TD0804). Together, this worldwide network of researchers will be uniquely situated to contribute decisively to a cross-cultural conservation framework for soundscapes.

1:45

4pAAa3. Mapping the landscape of soundscape research. Hill Kobayashi and Saoru Saito (World Forum for Acoust. Ecology (WFAE) P.O. Box 268, Fairfield Victoria, 3078 Australia)

In this paper, I give a personal view on what could be the landscape of soundscape research. I will describe the philosophical content of this discipline, the current state of aesthetic aspect, the challenges in technical issues and the future direction of Soundscape activities among researchers, practitioners, designer, and composers. Given that to map a landscape of this discipline with discussion is a very challenging task. It requires us to connect knowledge from different disciplines with perceptional sense. The paper aims to address recent collaboration opportunities for interdisciplinary engagement and key topics for debate.

2:05

4pAAa4. Sounds in cause: Soundscape and evolution. José Manuel Berenguer (Caos-Sonoscop. CCCB. Montalegre, 5. 08001. Barcelona. Spain)

Beyond the therm of landscape, that has often been defined as "a painting, drawing, or photograph depicting natural scenery," the concept of soundscape should lack any aesthetic significance and be thought as a technical therm naming the whole sonic experience of animals having sense of hearing. In general, from a methodological point of view, approaches describing landscape as "a view of some natural place" does not seem to be useful, because artificial and natural are often indistinguishable. Soundscape does not need to be considered natural or artificial. For instance, it seems evident that in a city, most sources of sound objects in soundscapes are human activities; anyway, even in a city, sounds produced by non-human species can easily be found. Soundscapes are complex structures that can be considered in terms of evolution. They evolve sound sources adapt their sonic productions to sonic productions of other sound sources sharing the same environment. It happens in human and non-human soundscapes. This way of thinking underlies sounds in cause, a project that started building a database of soundscapes recorded following a strict methodology and now proposes an Internet2 Network of Laboratories and Stations for Permanent Listening to the Soundscape.

2:25

4pAAa5. On the soundscape of urban parks. J. Luis Bento Coelho and Mohammed Boubezari (CAPS, Instituto Superior Tecnico, TULisbon, Lisbon, Portugal)

Urban parks are parts of every city fabric and usually well appreciated by the citizens for providing restoration and some "quiet," at least when compared to most other city areas. Research on the soundscape of parks in cities in Portugal and in Brazil have been conducted in order to assess what makes such areas sonically interesting. Work is being directed to the differentiation of the sound components of the overall sound environment in different types of parks and on techniques for mapping the perceived sound components. The work also aims at understanding effects of climate and culture on the perception of the soundscape. Results are presented and discussed.

2:45

4pAAa6. A livingscape approach to characterize urban historical places. F. La Malva, A. Astolfi, P. Bottalico, and V.R.M. Lo Verso (TEBE Res. Group, Polytechnic of Turin, Dept. of Energetics, Turin, Italy)

The quality of life concerned with open spaces has more and more become an essential part of urban culture. The evaluation of environmental effects as perceived by people is primarily a subjective issue, rather than being simply based on objective parameters. This paper presents an approach called livingscape, used to assess the quality of urban spaces. It consists of analyzing and correlating psychometric tools to measure the perception of environmental quality with different aspects related to the urban blight (both in architectural and environmental terms) and objective investigation of environmental quality through the measurement of acoustic, visual, thermal, and IAQ physical parameters. Livingscape data were collected in 13 key-spaces of St. Salvario, an historical district in Turin (Italy), during summer 2010 and winter 2011, selected based on an historical analysis to characterize the past and present district soundscape and subdivided in nodes, paths, and edges. The subjective environmental perceptions are delineated through the analysis of the questionnaires submitted to the users of the area, outdoor. Objective measures (acoustical, lighting, and thermal parameters) were combined to subjective responses, thus providing a more complete key-spaces characterization. The investigation aims to describe the changes in the key-spaces characterization from 19th century to nowadays.

3:05–3:15 Break

3:15

4pAAa7. Using a soundscape approach to develop an acoustic ecology plan for a city. Lisa R. Lavia (Noise Abatement Society, Ste. 2, 26 Brunswick Terrace, Brighton, England, BN3 1HJ, United Kingdom), Max Dixon (Independent Consultant, London, England, United Kingdom), Osten Axelsson (Decorum Commun.s, Passvagen 30, SE-147 53 Tumba, Sweden), and Harry Witchel (Brighton and Sussex Med. School, Falmer, Brighton, BN1 9PS, United Kingdom)

Sounding Brighton is a collaborative project exploring practical approaches toward better soundscapes focusing on soundscape issues related to health, quality of life, and restorative functions of the environment. The project provides the opportunity to raise awareness and promote communication on soundscapes among the general public, stakeholders and those involved in policy, including encouraging exploration of new ways of listening in local soundscapes, and new ways of tackling noise and improving local soundscape quality. The project is working to provide opportunities to discuss how soundscape concepts might, alongside tackling conventional noise problems, contribute to local planning and environmental improvement as part of a city wide engagement process in the city of Brighton and Hove in England in the United Kingdom. A range of environments, e.g., seafront, foreshore, historic terraces, squares, lanes, parks, and gardens, are being considered. A soundmap of the city is being developed utilizing the Swedish Soundscape-Quality Protocol (developed by Osten Axelsson, Mats E Nilsson and Birgitta Berglund); a public outreach exhibition is being developed; and a night noise intervention study is planned to explore the relationship between soundscapes and the brain, community well being, social cohesion, and the physical and mental health of individuals.

3:35

4pAAa8. The introduction of the concept of soundscape to urban design analysis. Maria Tomalova MA (46 Leinster Gardens, W2 3AT, London, the UK, m_tomalova@hotmail.com)

The objective of this paper was to provide pragmatic approach through the analysis of Covent Garden Piazza's soundscape with intention to show the way of its implementation into urban design analysis, whereas analysis of soundscape is missing part of urban design analysis. So, soundscape suppose to be a new tool for designer to create more pleasant sound ambient environments not only for users who are able to perceive urban space visually but also for those who can use another senses only like aural and tactile to identify relevant quality of urban places. In this research were employed methods of subjective evaluation and spatial analysis of soundscape (entirely qualitative research). The data were collected through exploration of the soundscape preference and level of awareness of ambient sound environment within explorative single case study and thorough observation of activities on the study site and investigation of urban form. Thus, the urban uses and activities of the location and its acoustic identification are closely linked. Recent urban analyses generally take only human and technical noises into consideration, excluding the rest of noises of sonic environment and way of their perception by human beings. Acoustic dimensions must be available for design.

3:55

4pAAa9. Urban sound design—Utopia or urgent need. Nina Hllgren (Dept. of Architecture, KTH Royal Inst. of Technol./Konstfack Univ. of Arts Crafts & Design, Box 3601, 12627 Stockholm, Sweden, nina.hallgren@konstfack.se)

Today we are facing the consequences of about 100 years of urbanization, confronting questions about quality of life in relation to efficiency and economical benefits. Hard facts are considered to be more reliable than values which are not so easily measurable. The quality of sound is one of them. Urban designers and architects are currently not fully aware of the interaction between the built outcomes of their work and the process of propagation and perception of sound. Designing houses, relations between houses, connections between places, whole neighborhoods, and cities is a serious task affecting many different aspects of life. But we can still note a recurring absence of knowledge regarding the complex relation between visual and sonic realities. This reality is in fact what surrounds the urban inhabitant for an entire lifetime. But what can or should the architects and planners do? As has been recently pointed out in a report on the subject, this professional group is lacking the tools, language, and guiding examples for being able to implement anything in reality. Implementing what? What can really be done or designed to improve the sonic environment and for what purpose?

4:15

4pAAa10. Piazza del Marchese Paolo: An architectural and soundscape design to redevelop an outdoor public space. Achille Sberna, Francesco Asdrubali (Dept. of Industrial Eng., Univ. of Perugia, Via G. Duranti 67, 06125 Perugia, Italy), and Brigitte Schulte-Fortkamp (Technische Universitaet Berlin, Germany)

This paper will report about a design procedure regarding the redevelopment of an open square located in the historical center of Città di Castello, Italy. The square has five entrances and it is surrounded by old buildings. The Public Library of the town is also located at this place. Currently, the square is used as a parking lot. The goal of the design is to redevelop this square matching the given context. For the design procedure, first, the visual and acoustical status of the place will be described. Second, binaural recordings will be carried out to measure the acoustical climate and third, soundwalks will be conducted to help to detect the soundmarks of the area. Moreover, the idea is to transform the space in a pedestrian area and to consider the square as an acoustical "outdoor floor" for the library. The design process will be focused on the preservation of the genuine Soundscape.

4:35

4pAAa11. How the soundscape approach enlightens the knowledge about the feeling of safety in urban spaces: A matter of networking and interdisciplinarity. Kay Sebastian Voigt and Brigitte Schulte-Fortkamp (Technische Universite, Inst. Fluid Mech. and Eng., Secr TA7, Einsteinufer 25, Berlin, Germany)

In a current project on dynamics of urban safety and its arrangements funded by the Federal Ministry of Education and Research, Germany http://www.dynass-projekt.de/projekt-dynass/ Dynamische Arrangements städtischer Sicherheitskultur/, the Soundscape approach is one of diverse approaches to investigate areas of different cities with respect to the perception and production of safety in such environments. As to the perception of safety, areas as well as its production special attention will be given to the reconstruction of

decisive factors. The current status of the international standardization process of the ISO/TC 43/SC 1/WG 54 and the contribution of the COST network are used here as the basis to improve procedures and measures in the applied Soundscape approach. Moreover, this paper will discuss the Soundscape research as an interdisciplinary approach and in an interdisciplinary approach with regard to investigations upon safety management in urban areas. First results give hints to the importance of the acoustic, visual, and social structures of an area in the face of the Soundscape procedure.

Contributed Papers

4:55

4pAAa12. Water features and acoustic diversity of urban parks. Osten Axelsson (Passvaegen 30, SE-147 53 Tumba, Sweden) and Mats E. Nilsson (Stockholm Univ., SE10691 Stockholm, Sweden)

Water features are well-acknowledged in architecture and urban planning for their visual characteristics. But, how do water features contribute to acoustic diversity and soundscape quality? Visitors in an urban park were recruited to complete a questionnaire on how they perceived the park including its soundscape. Meanwhile, the soundscape was manipulated by turning a fountain on or off at irregular hours. The fountain sounds had a positive effect on soundscape quality in an area close to the fountain, by masking background road-traffic noise. The fountain sound also masked other natural sounds, which may have a negative influence on acoustic diversity and soundscape quality. In addition, some participants may have mistaken the fountain sounds for distant road-traffic noise. Hence, when introducing a water feature in an urban park it is necessary to consider the acoustic characteristics of the water sounds, as well as the placement of the water feature.

5:10

4pAAa13. Researching sound in silence. Jurgen De Blonde (Aifoon vzw, Nieuwevaart 117a, B-9000 Gent, Belgium, jurgen@aifoon.org)

Aifoon is an educational arts organization that investigates sound in silence. We investigate poetic and communicative possibilities of everyday sounds. Our aim is to have people communicate about their immediate surroundings in a non-verbal way by using sounds taken from those surroundings. In our workshops, we teach people to record sounds, we teach microphone awareness, we open ears, draw sounds, and compose with these drawing, compose with sound, make a montage on a computer, and talk about the results. We avoid using music and words since these two elements are too coded. We take the results and the questions that rise from our work back to artists, researchers, and the public for further interpretation and investigation. We have taken our core philosophy as a basis for an exposition, a number of events, an ensemble, and a couple of installations in an attempt to take our educational role beyond the workshop session into the public space. This has proven successful and has often helped us to explain our ideas and outsiders to understand them.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 SUNRISE, 1:25 TO 5:45 P.M.

Session 4pAAb

Architectural Acoustics: Variable Acoustics—Methods for Effective Collaboration

Roger W. Schwenke, Chair
Research and Development, Meyer Sound Laboratories, 2832 San Pablo Ave., Berkeley, CA 94702

Chair's Introduction—1:25

Invited Papers

1:30

4pAAb1. What you call it does matter: A vocabulary for active acoustics. Kurt M. Graffy (Arup Acoust., 560 Mission St., San Francisco, CA 94105, kurt.graffy@arup.com)

Active acoustics (Electronic Architecture) systems have reached a maturity in terms of their acoustic performance which makes them a viable option for multi-use venues as well as renovations of existing facilities. However, the awareness on the part of owners and facilities regarding the capabilities and potential acoustic benefits of such systems is still developing. More significantly, the corresponding level of awareness for designers and architects, which active acoustics can actually represent options for how architectural, structural, and building service designs may proceed, is even less developed. At this intersection of past techniques versus current abilities, we are in need of a common vocabulary for active acoustics; a common vocabulary to interact with the design team, a common vocabulary to assist venue operators and owners understand the benefits, and a common vocabulary to interact with the musicians and artists in tuning and configuration. The presentation references some recent projects utilizing active architecture primarily focused on the interaction with the design team and subsequently with the artists, and the development of a common language as the projects moved forward.

1:50

4pAAb2. 2006 International Music Festival. Edward Dugger III (4490 SW Long Bay Dr. Palm City, FL 34990, edward@edplusa.com)

Throughout the 3-week run of the 2006 International Music Festival held in Boca Raton, Florida, the 4000 seat Mizner Park Amphitheatre was fitted with an active acoustic system. We will present a case study that explores many of the elements of the project including the initial audio system selection, installation, rehearsals, and performances. Since this was an unusual project both in its exterior application and large scale, we will present the design features that had significant implications for the musicians, concert presenters,

4p THU. PM

stage hands, and audience. At the request of the venue operators, a new design has been developed to install a permanent active acoustic system for the amphitheater. This design will also be presented.

2:10

4pAAb3. Successfully merging architectural and electronic acoustical treatments. Steve Barbar (30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

In enclosed volumes, the integration of electronic acoustical components with architectural surface treatments forms a hybrid system that produces the perceived acoustical conditions. Since the underlying operating principles for electro-acoustic enhancement systems differs considerably between manufacturers, the requirements for system infrastructure are not germane, nor is the optimum integration of architectural treatments. As a result, the nature of the work performed by the acoustical consultant changes to accommodate optimum performance of the specific "hybrid" system, which may also include other forms of variable treatments.

2:30

4pAAb4. Working with musicians to increase low-frequency performance of an active acoustic system in a music practice room. Ron Freiheit (Wenger Corp., 555 Park Dr. Owatonna, MN 55060, ron.freiheit@wengercorp.com)

To enhance the performance of an active acoustic system for music practice rooms, a new speaker was developed with extended low-frequency response. To better understand the performance desired from certain musicians, a small number of cellists provided observations and opinions about what environment was most pleasing to them, specifically related to the expanded low-frequency response of the active system. A system was developed that allowed the tuning of the low-frequency response to better ascertain at which point it was optimized and below or above which very little improvement was noted. Once these parameters were optimized, design criteria for the speakers were determined. Also discovered during this research were challenges musicians faced in discriminating between the direct sound and active sound field.

2:50

4pAAb5. The care and feeding of clients with variable room acoustics systems. Edward Logsdon (D. L. Adams Assoc., Inc., 1701 Boulder St., Denver, CO 80211, elogsdon@dlaa.com)

The care and feeding of clients with variable room acoustics systems. Now that the Variable Room Acoustics System (VRAS) is designed and installed in the room, what do you do? An Acoustical Consultant's Perspective One of the primary educational goals for the Colorado College, Edith Kinney Gaylord Cornerstone Arts Center in Colorado Springs, CO, is to encourage collaboration between the music, drama, dance, film, and visual artists who perform and display their work in the facility. The building offers many opportunities for film/video and theater students, or music and sculptural artists to work together on multimedia presentations. We used "collaboration" to further encourage the student and faculty performers and artists to communicate their needs and priorities for use of the multipurpose main theatre early on in the design. Subjective opinions of how the room sounds, or how best to control the VRAS system, had to be carefully addressed and assessed to avoid confusion and to achieve unanimity. We will review our process of establishing an artistic vocabulary that allowed a jury of users to advise us on the types and number of room presets needed for the various uses of the theatre and the validation and approval of the system.

3:10–3:30 Break

3:30

4pAAb6. Various applications of active field control. Takayuki Watanabe and Masahiro Ikeda (Spacial Audio System Group, Yamaha Corp., 10-1 Nakazawa-cho, Nakaku, Hamamatsu, Japan, watanabe@beat.yamaha.co.jp)

Several types of various active field control (AFC) applications are discussed, while referring to representative projects for each application. (1) Realization of acoustics in a huge hall to classical music program, E.g., Tokyo International Forum: This venue is a multi-purpose hall with approximately 5000 seats. AFC achieves "loudness" and "reverberance" equivalent to those of a hall with 2500 seats or fewer. (2) Compensation of acoustics on stage without rigid shell using the electro-acoustic method. E.g., High school auditoriums: In these renovation projects, AFC achieves "acoustical support" for performer on stage and "uniformity" throughout the auditorium from the stage to the audience area, etc. (3) Improvement of the acoustics under the balcony in auditoria. E.g., Experiments on a full-scale model and the school auditorium. The system is a non-regenerative system, and the loudspeakers, located at positions corresponding to measurement points across the balcony, recreate the reflecting sound from above the balcony area, which otherwise fail to reach to the listeners under the balcony. The results of the experiment show that the system is significantly better for all tests to the use of no system and that the system is superior to a standard PA (delay system).

3:50

4pAAb7. Acoustic repair: Recent experience with the acoustic control system (ACS) for improving acoustic conditions in two existing venues. Timothy E. Gulsrud (Kirkegaard Assoc., 954 Pearl St., Boulder, CO, 80302 tgulsrud@kirkegaard.com) and Arthur van Maurik (Acoust. Control Systems BV, Speulderweg 31 3996 LA Garderen, The Netherlands)

Active acoustics systems are becoming more prevalent in architectural acoustics practice, particularly in the context of repairing or improving acoustics in existing venues. Governmental policies to reduce funds and subsidies put into new facilities for the performing arts are another reason for designers to consider the use of active acoustics. This paper highlights two recent examples of such installations of ACS systems, one at the Sydney Opera House Concert Hall, and the other at MBCCH, Winnepeg, Canada. Collaboration between the system designer, the musicians, and the acoustics consultant will be emphasized, along with techniques used to evaluate the systems' performance in the halls.

4:10

4pAAb8. Indoor outdoor acoustics: Active acoustics at the New World Center, Miami Beach Soundscape. Frederick R. Vogler (Sonitus, fred@sonitusconsulting.com), John Pellowe (Meyer Sound Labs, Inc.), and Steve Ellison (Meyer Sound Labs, Inc.)

The Frank Gehry designed New World Center is home to the New World Symphony in Miami Beach, FL. This facility includes an intimate 756 seat concert hall and is used as a training platform both for symphonic conductors and musicians. The adjacent park uses an active acoustics system to allow a similar number of people in the park to simultaneously experience the indoor concert experience in an open air environment. The immersive sound of the reproduced concert experience is accompanied by a 7000-square-foot projection wall that carries live video of the performance. The system captures the natural acoustic of the concert hall using microphones distributed throughout, and these signals are processed and then transmitted to the park utilizing a set of 160 distributed loudspeakers. The successful design, commissioning, and tuning of the system relied on a team approach between the architect, consultants, manufacturer, installer, and venue operators. Scope within the team is explored, challenges revealed, and suggestions offered to help ensure the success of new multi disciplinary ventures such as this. Similarities and differences to a surround sound broadcast transmission of the Los Angeles Philharmonic are also reviewed.

Contributed Papers

4:30

4pAAb9. Acoustical design of New World Center, Miami Beach, FL. Daniel F. Beckmann, Kayo Kallas, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoust., 2130 Sawtelle Bl. Ste 308, Los Angeles, CA 90025, beckmann@nagata.co.jp)

The New World Center opened in January 2011 after an 8-year process to design and build the $160 million facility. The Frank Gehry-designed facility for the New World Symphony, "America's Orchestral Academy," has at its heart the 756-seat auditorium, designed in the arena style. Also in the facility are one large orchestral rehearsal room, seven medium-sized Chamber/Ensemble rehearsal rooms, and 24 individual coaching/practice rooms. Administrative offices and production spaces complement the spaces for music to bring the building to 100 641 square feet. The acoustical design of the 756-seat auditorium was performed in close collaboration with Gehry and the founder of the New World Symphony, acclaimed conductor Michael Tilson Thomas. To ensure validity of the acoustical design, a 1:24 scale model test was performed. Flexibility in rehearsal and performance was one of the prime requirements for the space as the program includes much more than just orchestral performance, which is met by a curtain system for variable acoustics, a highly flexible stage lift system, 247 retractable seats, and four alternate "Performance Platforms." Five large "sails," which double as projection surfaces, a proper ceiling above the stage, and a steeply raked audience area are amongst the acoustical design elements that are reported.

4:45

4pAAb10. Acoustical design of Soka University Performing Arts Center. Kayo Kallas, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoust. America, 2130 Sawtelle Blvd., Ste. 308, Los Angeles, CA 90025, kimotsuki@nagata.co.jp)

The Soka University Performing Arts Center and Academic Building will open in September 2011. The $73 million performing arts center is open to the public, hosting various types of performing arts. For lovers of the performing arts, the center will become another choice among the many fine venues in Orange County. The building houses a 1000-seat multipurpose hall, a 150-seat black box theater, support spaces, and classrooms. The multipurpose hall was designed primarily as a concert hall, and later to become suitable for dance, plays and musicals. To satisfy these flexible programs, the design features a curtain system for variable acoustics and an automated stage lift to accommodate concert, thrust, and convocations stage settings. The seating layout of the multipurpose hall was arranged in the arena style. The room shape and interior materials were carefully selected to optimize the acoustics of the space, including the two layered ceiling design: one for aesthetics and the other for room acoustics. Acoustical design and characteristics of the new multipurpose hall will be reported.

5:00

4pAAb11. The effect of reverberation enhancement on the diffusion of the sound field. Hugh Hopper, David Thompson, and Keith Holland (I.S.V.R., University of Southampton, University Rd., Southampton, SO17 1BJ, UK)

Reverberation enhancement is a technology which allows the reverberation time of a room to be increased. It is important to consider the effect of this technology on the other measurable attributes of the room response. The spatial variation of steady state sound pressure level and reverberation time within the room can be used to measure the extent to which the room approximates a diffuse field. A theoretical value of these quantities can be predicted for an ideal diffuse field, and the ratio between the measured and theoretical values gives a normalized measure of the diffusion of the sound field. This work investigates the changes in these measures when reverberation enhancement is applied to a room. Experimental results have shown that the normalized measures of diffusion increase with the introduction of reverberation enhancement. This implies a reduction in the homogeneity and isotropy of the sound field which may be perceived as a reduction in subjective quality.

5:15

4pAAb12. Variable acoustics at hauppauge high school. Richard F. Riedel (Riedel Audio & Acoust., LLC, 443 Potter Blvd., Brightwaters, NY 11718)

The author will discuss how variable acoustic solutions were used to solve room anomalies and provide a means to control room size. Once plagued by a significant rear wall reflection that interfered with stage musicians' timing, due to its later arrival time, the author will explain how a unique custom designed retractable acoustical diffuser reduced this problem. Another recurring issue in the space was the need to reduce the size of the seating area, normally 1100 seats, to one which provided a more intimate setting for dramatic presentations, which were normally not attended by large audiences. Using a commercially available acoustical product, the issue was solved and a means provided to alter the room's acoustical environment. This paper presents details about the methods used to provide variation to the room's acoustics and cites specific measurements of the space with and without the variable acoustical elements.

5:30

4pAAb13. Sound absorbing vertically retracting drapery: A comparative study. Liz L. Lamour (Univ. of Kansas, School of Architecture, Design and Planning, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, lizlamour@gmail.com), Ben Bridgewater (Univ. of Kansas, Lawrence, KS 66045), and Ben Brooks (Univ. of Kansas, Lawrence, KS 66045)

The use of drapery as a variable sound absorbing material is widespread in theaters, university music halls, and other spaces where variable reverberation time is desired. As a class project at The University of Kansas, the authors and their classmates tested two types of sound absorbing vertically retracting drapery manufactured by a theatrical contracting firm using reverberant room methods. Fabric used in these two tests was cotton velour and acrylic velour. Coefficients of absorption were compared with coefficients of absorption published by a firm specializing in the fabrication of vertically retracting sound absorbing velour drapery. The measured and published data allowed both types of vertical retracting drapery to be specified for a multipurpose auditorium renovation project at a college in Kansas. The college auditorium renovation project will use the drapery fabricated by the specialty firm and the on-site determined coefficients of absorption will be presented and compared with the manufacturer's published data and also compared with the data obtained in the reverberant room for the somewhat different retracting drapery design by the theatre contracting firm. And if it proves to be possible, the on-site sound absorption measurements will be made at another college auditorium which uses cotton velour retracting drapery produced by the theatrical contracting firm. These data will be useful in specifying sound absorbing vertically retracting drapery for future projects.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 1, 1:30 TO 3:15 P.M.

Session 4pAB

Animal Bioacoustics: Long-Term Acoustic Monitoring of Animals II

Simone Baumann-Pickering, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720

Invited Papers

1:30

4pAB1. Diel and lunar variations of marine ambient sound in the North Pacific. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Ana Širović (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Marie A. Roch (San Diego State Univ., San Diego, CA 92182), Anne E. Simonis, Sean M. Wiggins (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Erin M. Oleson (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI 96822), and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

Marine ambient sound was recorded on autonomous high-frequency acoustic recording packages (bandwidth 10 Hz to 100 kHz) during long term deployments at multiple sites across the North Pacific, from the high latitude Aleutian Islands to tropical Palmyra Atoll in depths of 600–1 000 m. Most intertropical but no temperate locations showed a distinct diel pattern in ambient sound. The soundscape at each location was unique, yet there was a similar recurring sound of unknown origin in lower latitude locations. This sound had a peak frequency around 3–5 kHz and was recorded only for several hours after sunset. Additionally, at some locations, a broadband acoustic signal with bandwidth up to 60 kHz was recorded at night with crepuscular peaks. Both sound patterns were lunar dependent with lower acoustic levels during full moon phases. Site-specific diel and seasonal acoustic patterns have been observed for various odontocete species. Correlations between odontocete presence and levels of ambient sound are investigated. [Work supported by NOAA-Pacific Islands Fisheries Science Center, US Navy-N45/PACFLT, ONR, Pacific Life, Ocean Foundation, University of California, San Diego.]

1:50

4pAB2. Long-term passive acoustic monitoring of nearshore ecosystems in the Northwestern Hawaiian Islands. Marc O. Lammers, Lisa Munger (Hawaii Inst. of Marine Biology, P.O. Box 1346, Kaneohe, HI 96744, lammers@hawaii.edu), Pollyanna Fisher Pool (Univ. of Hawaii, Honolulu, Hawaii), Kevin Wong (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii), Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), and Russell E. Brainard (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii)

Monitoring the changing state of marine habitats in remote areas is, in most cases, a challenging task due to limited and/or infrequent opportunities to make direct observations. Passive acoustic monitoring is sometimes the best means of establishing long-term biological trends in such areas. Since 2006, an effort has been underway to monitor the neashore ecosystems of the Northwestern Hawaiian Islands (NWHI) using a network of Ecological Acoustic Recorders. A wide range of acoustic signals are being monitored to infer biological trends and to gauge the relative stability of the ecosystem. Among the variables measured are the acoustic activity of snapping shrimp, the incidence of cetaceans and the extent of spectral and temporal partitioning of the acoustic space by different taxa, measured as the “acoustic entropy” of the habitat. Multiyear time series of the different measures provide baseline levels of biological activity at each location and also reveal periods of anomaly. Observed trends are then examined for corollary relationships with oceanographic and meteorological parameters measured both in situ and remotely. The data obtained thus far are providing valuable insights that will help assess the long-term response of ecosystem in the NWHI to both natural and anthropogenic factors

2:10

4pAB3. Eavesdropping on coconut rhinoceros beetles, red palm weevils, Asian longhorned beetles, and other invasive travelers. Richard W Mankin (USDA-ARS-CMAVE, 1700 SW 23rd Dr., Gainesville, FL 32608, richard.mankin@ars.usda.gov)

As global trade increases, invasive insects inflict increasing economic damage to agriculture and urban landscapes in the United States yearly, despite a sophisticated array of interception methods and quarantine programs designed to exclude their entry. Insects that are hidden inside soil, wood, or stored products are difficult to detect visually but often can be identified acoustically because they produce 3–30-ms, 200–5 000-Hz impulses that are temporally grouped or patterned together in short bursts. Detection and analysis of these sound bursts enables scouts or inspectors to determine that insects are present and sometimes to identify the presence of a particular target species. Here is discussed some of the most successful acoustic methods that have been developed to detect and monitor hidden

insect infestations. Acoustic instruments are currently available for use in rapid surveys and for long-term monitoring of infestations. They have been useful particularly for detection of termites, coconut rhinoceros beetles, red palm weevils and Asian longhorned beetles in wood, white grubs and Diaprepes root weevil in soil, and stored product insects.

Contributed Papers

2:30

4pAB4. Acoustic monitoring of dolphin populations in the Gulf of Mexico. Kaitlin E. Frasier (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr. La Jolla, CA 92093, kefrasie@ucsd.edu), Melissa S. Soldevilla (Protected Resources and Biodiversity Div, NMFS/SEFSC, Miami, FL 33149), Mark A. McDonald (WhaleAcoust., Bellvue, CO 80512), Karlina P. Merkens, Sean M. Wiggins, John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093), and Marie A. Roch (San Diego State Univ., San Diego, CA 92182)

High-Frequency Acoustic Recording Packages (HARPs) continuously monitored delphinids at five sites in the northeastern Gulf of Mexico during and after the *Deepwater Horizon* oil spill. Surface oil reached two sites, while the three unexposed sites functioned as “controls.” Presence of dolphin vocalizations (clicks, whistles, and burst pulses) was documented at exposed and unexposed sites over the course of a year following the oil spill. These sites are within the known habitat ranges of 11 species of delphinids. Broadband towed array recordings with visual identifications were used to determine species-specific vocalization characteristics, which were then compared with autonomously recorded vocalizations. Two species have distinctive vocalizations that match between towed array and autonomous recordings. At least four more unique vocalization patterns were detected autonomously, which may be species-specific. Both clicks and whistles were explored for identifying features. The data provide a comparative view of delphinid presence relative to the oil spill.

2:45

4pAB5. Passive acoustic monitoring of sperm whales during and after the Deepwater Horizon oil spill. Karlina Merkens (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, kmerkens@ucsd.edu), Mark A. McDonald (Whale Acoust., Bellvue, CO 80512), Simone Baumann-Pickering, Kaitlin Frasier, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0205)

The Deepwater Horizon oil spill during the summer of 2010 impacted a region of sperm whale habitat along the continental slope and deep waters of the Gulf of Mexico. Passive acoustic monitoring was used to study the potential impact of the oil spill on sperm whales by recording trends in their characteristic sounds, such as echolocation clicks and foraging creaks. High-frequency Acoustic Recording Packages (HARPs) were deployed shortly after the oil spill began; one was located close to the Deepwater Horizon well, above which the sea surface was contaminated by oil throughout the summer of 2010, and another was deployed in a region of sperm whale habitat that remained unexposed to surface oil to function as a “control” site. At both sites, sperm whales were detected on a majority of days during the nearly year-long recording period. Sperm whale presence was evaluated from detected clicks and creaks, and changes in these sounds over time and between sites were compared.

3:00

4pAB6. Long-term acoustic monitoring of marine mammal response to the 2010 oil spill in the Northern Gulf of Mexico. Natalia Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy S. Ackleh, Baoling Ma (Univ. of Louisiana at Lafayette, LA 70504), Christopher Tiemann (Univ. of Texas at Austin, Austin, TX 78713), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

The 2010 deep water horizon (DWH) oil spill in the Northern Gulf of Mexico brought a need for assessing spill impact and recovery timeline for the deepwater ecosystem, including marine mammals. Passive acoustics is emerging as a viable technology to monitor short-term and long-term abundance dynamics and to assess different factors that may cause an observed response. Multi-year pre-spill and post-spill acoustic data collected at different distances from the DWH incident site by the Littoral Acoustic Demonstration Center (LADC) are used to compare first-year oil spill response by three different groups of marine mammals: sperm whales, beaked whales, and dolphins. Densities of acoustic phonations by these animals are extracted from collected data and used for point estimates of the resident population density. As an example, a regional abundance estimate shows a decrease in the number of sperm whales at the site nearest to the DWH (9 mi away) which exceeds statistical uncertainties and can be accepted as an existing trend. The use of acoustic data to extract information about environmental factors, such as anthropogenic noise level or food call densities, that may contribute to the explanation of existing trends is also discussed. [Work is partially supported by NSF.]

4p THU. PM

Session 4pBA

Biomedical Acoustics: Therapy and Applications

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

Contributed Papers

1:30

4pBA1. Effects of physical properties of the skull on high intensity focused ultrasound for transcranial sonothrombolysis. Prashanth Selvaraj (Dept. of Mech. Eng., Etcheverry Hall, Univ. of California, Berkeley, CA pselvaraj@me.berkeley.edu), Kohei Okita (Ctr. for Intellectual Property Strategies RIKEN 2-1 Hirosawa, Wako, Saitama 351-0198, Japan), Yoichiro Matsumoto (Univ. of Tokyo, 7-3-1 Hongo, Bunkyo, Tokyo 113-8656, Japan), Arne Voie, Thilo Hoelscher (Univ. of California, San Diego, 212 West Dickinson St.), Hope Weiss, and Andrew J. Szeri (Etcheverry Hall, Univ. of California, Berkeley, CA)

The use of high intensity focused ultrasound (HIFU) in transcranial sonothrombolysis is emerging as a promising therapeutic intervention after stroke. Of interest in the present study is the evolution of the wave from transducer to focus, with special attention to two aspects. One is the attenuation of the wave before it reaches the focus, the other is the scattering of the wave at tissue interfaces leading to alteration of the focus. A code developed for tissue ablation (Kohei Okita, Kenji Ono, Shu Takagi, and Yoichiro Matsumoto, Int. J. Numer. Methods Fluids 65:43–66 (2011)), has been modified to study the effect of the physical properties of the skull on the focusing of the HIFU waves. Phase delay of the array transducer is employed to focus the waves. A basic model illustrative of the calvaria of the skull has been used as only the physical properties of the bone are of interest here. Microbubble cavitation has been shown to enhance sonothrombolysis; hence, the altered wave is examined from the point of view of the bubble dynamics it engenders.

1:45

4pBA2. Vascular permeability with targeted contrast agents—The effect of physiologically relevant dynamic shear stress. Pavlos Anastasiadis (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 302, HI 96822, pavlos@hawaii.edu), Joshua J. Rychak (Targeson, 3550 General Atomics Court, San Diego, CA 92121), and John S. Allen III (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, HI 96822)

Targeted ultrasound contrast agents (UCAs) may be able to facilitate an early noninvasive diagnosis of atherogenesis. The coronary arterial branches might be routinely scanned for a clinical diagnosis since plaques often first form at disturbed flow regions within bifurcations. However, many outstanding questions exist on the targeting efficacy subject to pulsatile and potentially pathological flow in these areas. Targeted ultrasound contrast agents conjugated to the antibody of the intercellular adhesion molecule-1 (ICAM-1) are injected into y-shaped flow chambers that model the coronary heart artery bifurcations. Simultaneous measurements of the barrier function in physiologically relevant dynamic shear stresses are monitored with electric cell impedance (ECIS) measurements. The study aims at quantifying variations of the barrier function at regions with exposure to different shear stresses and correlating this information to the binding efficacy of targeted UCAs conjugated to the antibody of ICAM-1. These results are discussed with respect to reported targeted ultrasound contrast agent studies of inflammation and plaque in Apolipoprotein E-deficient mice.

2:00

4pBA3. Synergistic interaction between stress waves and cavitation is important for successful comminution of residual stone fragments in shock wave lithotripsy. Jaclyn Lautz, Georgy Sankin, and Pei Zhong (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, jaclyn.lautz@duke.edu)

To assess the role of stress waves and cavitation in comminuting residual fragments during shock wave lithotripsy (SWL), cylindrical 4×4 mm BegoStone phantoms were treated in an electromagnetic lithotripter either at the focus ($z = 0$, $p_+ = \sim 45$ MPa) or pre-focally ($z = -30$ mm, $p_+ = \sim 24$ MPa). The treatment was performed with the stone immersed either in degassed water or in Butanediol, which has similar acoustic impedance to water but much higher viscosity to suppress cavitation. At the focus, the first fracture was observed after 26 ± 9 shocks, both in water and Butanediol ($p = 0.7$). However, when stones were moved pre-focally where comparable cavitation is produced (based on high speed imaging), the average shock number required for the initial fracture was increased to 66 ± 10 in water and 122 ± 20 in Butanediol ($p = 0.002$). Below -40 mm prefocally ($p_+ < 20$ MPa), stones did not fracture in water even after 2,000 shocks, although cavitation was observed. Furthermore, stone comminution at the focus after 250 shocks was $\sim$35% in water compared to $\sim$5% in Butanediol ($p < 0.001$). Altogether, these findings suggest that a synergistic interaction between stress waves and cavitation is critical in producing effective stone comminution during SWL. [Work supported by NIH and NSF GRFP.]

2:15

4pBA4. Investigation on the effect of specular reflections from stone surface on twinkling artifact. Wei Lu, Oleg A. Sapozhnikov, John C. Kucewicz, Bryan W. Cunitz, Peter J. Kaczkowski, Lawrence A. Crum, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle WA 98105)

The twinkling artifact can highlight kidney stones during ultrasound color Doppler imaging with high sensitivity for stone detection. The mechanism of the twinkling artifact is still under debate. It was reported previously that twinkling appeared distal to the echogenic reflection from the stone surface in cases with no signal saturation. [Lu *et al.*, JASA **129**(4), p. 2376]. In this report, the effect of specular reflections on twinkling was investigated. Human kidney stones (5-9 mm in length) were embedded in a polyacrylamide gel phantom. Radio-frequency (RF) data were recorded from pulse-echo ensembles using a software-programmable ultrasound system. The variability within the beamformed Doppler ensemble, which is responsible for twinkling, was traced back to the unbeamformed RF channel data to identify whether variability arose disproportionately on channels receiving the specular reflection. The results showed that the specular reflection did not saturate individual channels and that the variability was observed on most channels with similar magnitude, which indicates that the appearance of twinkling does not rely on the specular reflection from the stone surface. Instead in the beamformer, the varying signals have the appearance of arising from a point source within the stone. [Work supported by NIH DK43881, DK086371, DK092197, and NSBRI through NASA NCC 9-58.]

2:30

4pBA5. Inhibition of breast cancer cell proliferation by low-intensity pulsed ultrasound (LIPUS). Amit Katiyar, Kausik Sarkar (Mech. Eng., Univ. Of Delaware, Newark, DE 19716), and Krishna Sarker (Biological Sci. , Univ. of Delaware, Newark, DE 19716)

Cancer is the second leading cause of death in the United States, preceded only by heart disease. Cancer cells display an uncontrolled proliferation, controlling which has been a big challenge for cancer treatment. Ultrasound is best known for its application in diagnostic imaging; it is also a vehicle for delivering high frequency mechanical stimulation toward beneficial bio-effects. Unlike high intensity focused ultrasound, which is recently being investigated for thermal ablation of solid tumors, low intensity pulsed ultrasound (LIPUS) is directed toward cellular mechanisms. The effects of LIPUS on cancer cell proliferation are not known. Here, we demonstrate that LIPUS dose-dependently inhibits proliferation of breast cancer cell T47D as determined by several biochemical assays such as MTS, Alamar Blue, and BrdU assay. Statistically significant inhibition of T47D cell proliferation is observed when cells are exposed to 50–100 mW/cm^2. For this intensity range, LIPUS excitation inhibits the proliferation of T47D cells upto 50%. We also notice that inhibition of cell proliferation by LIPUS depends on its exposure time on cells. Minimum exposure time of LIPUS excitation for pronounced inhibitory effects on T47D cell proliferation is approximately 10 min.

2:45

4pBA6. Acoustical assessment of body water balance. Armen Sarvazyan (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618)

A new medical application of acoustics has recently emerged: assessment of body hydration status by ultrasonic measurement of muscle water content. The need for an easy-to-perform method for the detection of water imbalance is of the utmost clinical importance. Body hypohydration may cause severe health and performance problems, decreasing cognitive and physical work capabilities, while excessive hydration is a common symptom of many other diseases. The speed of longitudinal acoustic waves in muscle, as well as in other soft tissues, is defined by tissue molecular composition because both the density and bulk compressibility of tissue depend mainly on short range molecular interactions. Skeletal muscle is the largest water compartment in the body; it comprises 40% of body mass and 75% of muscle is water. The ultrasound velocity in muscle is a linear function of water content with the slope of about 3.0 m/s per 1% change in water content. We will describe the design, measurement principles, and testing results for new acoustic devices for assessment of hydration status of elderly and infants, two most vulnerable groups of population. Advantages and disadvantages of acoustical method of hydration over currently available methods are discussed.

3:00

4pBA7. Use of highly nonlinear solitary waves for the assessment of dental implants. Bruk Berhanu (942 Benedum Hall, Dept. of Civil and Environ. Eng., 3700 O'Hara St., Pittsburgh, PA 15261, bruk.berhanu@gmail.com)

This paper presents a noninvasive technique based on the propagation of highly nonlinear solitary waves (HNSWs) to monitor the stability of dental implants. HNSWs are mechanical waves that can form and travel in highly nonlinear systems, such as one-dimensional chains of contacting spherical particles (i.e., granular crystals). In this study, a granular crystal-based actuator/sensor, designed and built at the University of Pittsburgh, was used to introduce HNSWs into dummy implants that were inserted into either hardened plaster or treated beef bones. The waves reflected at the interface between the particle and implant were monitored to estimate the change in stiffness of the material. The hydration of the plaster was monitored because it can be considered largely similar to the osseointegration process that occurs in the oral connective tissue once a dental-endosteal threaded implant is surgically inserted. In the experiment using bone, the implant-bone system was immersed in an acid bath causing decalcification of the bone and, therefore, reduced stiffness of the bone itself, simulating the inverse of osseointegration. Positive correlations were found, in both experiments, between certain properties of the HNSWs and the stiffness of the test object, demonstrating that HNSWs show promise for use in assessment of dental implants.

3:15

4pBA8. Artificial flesh material selection for hearing protection evaluation system. Mehmet M. A. Bicak, Josiah M. Oliver, and Kevin R. Shank (Adaptive Technologies. Inc., 2020 Kraft Dr. Ste. 3040, Blacksburg, VA 24060)

Developing an advanced hearing protection evaluation system (HPES) in the form of an acoustic test fixture (ATF) allows for characterizing either circumaural or insert-type hearing protection devices (HPDs) in both impulse and continuous noise environments across the dynamic range of human hearing. This is a challenging task since the acoustical transfer paths through flesh contribute to the dynamic response of the system. Current ATFs do not account for the transfer paths through flesh to ear canal. In this study, we investigated several visco-elastic flesh materials numerically using coupled vibro-acoustic simulations, and experimentally using vibration and acoustic excitation methods. Geometrically representative prototypes are being developed using volume computed tomography (VCT) that include detailed features of the skull and flesh structure, so that flesh conducted sound transmission paths can be physically modeled. The HPD on ATF dynamic behavior is compared with the HPD on subject behavior using finite element simulation models developed using the VCT images. The material selection is validated using noise reduction and vibration experiments on the subjects.

3:30

4pBA9. Cough count as a marker for patient recovery from pulmonary tuberculosis. Brian H. Tracey (ECE Dept., Tufts Univ., 196 Boston Ave., Rm., 4330 Medford, MA 02155, brian.tracey@tufts.edu), German Comina (Laboratorio de Ingeniería Fsica, Facultad de Ciencias, Universidad Nacional de Ingeniería, Rimac, Lima, Per), Sandra Larson (Michigan State Univ., College of Osteopathic Medicine, East Lansing, MI), Marjory Bravard (Massachusetts General Hospital, Boston, MA 02114), Jose W. Lopez (Unidad de Epidemiología, Hospital Nacional Dos de Mayo, Lima, Per), and Robert H. Gilman (Johns Hopkins Bloomberg School of Public Health, Baltimore, MD)

In regions of the world where tuberculosis (TB) poses the greatest disease burden, clinicians often lack access to skilled laboratories. This is particularly problematic for patients with drug-resistant tuberculosis, as these patients will otherwise receive standard TB medication and will not respond to treatment. Thus, a lab-free method for assessing patient recovery during treatment would be of great benefit. We hypothesize that cough analysis may provide such a test, and have carried out a pilot study to record coughs from a cohort of patients in Lima, Peru. We describe algorithm development for cough data analysis and compare several event detection and classification strategies. Results from our ongoing validation efforts suggest that cough count (cough/hour) decreases noticeably after the start of treatment in drug-responsive patients. Our long-term goal is development of a low-cost ambulatory cough analysis system that will help identify patients with drug-resistant tuberculosis.

Session 4pEA

Engineering Acoustics and Underwater Acoustics: Vector Sensors, Projectors, and Receivers II: Receivers, Reception, and Transmission

Stephen C. Butler, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Roger T. Richards, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Invited Papers

1:05

4pEA1. Acoustic intensity vector probes. Gary W. Elko (Mh Acoust. LLC, 25A Summit Ave., Summit, NJ 07901)

Modern acoustic intensity measurement techniques began in the early 1980's with the realization that the imaginary part of the cross-spectral density was directly related to the active intensity component along the axis between two closely-spaced pressure microphones. Since acoustic intensity is a vector quantity, it was obvious that one would like to measure all three orthogonal components and to graphically represent acoustic power flow through space. Needing an interesting topic for a Ph.D. thesis at Penn State (and a way of funding it ..., thank-you US Navy), the task of investigating and developing the estimation of the acoustic intensity vector field fortunately came my way. This talk will present some of the early interesting and fun things that came out of working on in-air acoustic vector sensors. It will conclude with some more recent developments that have direct connections to the early acoustic vector probes that were built, tested, and used at Penn State.

1:25

4pEA2. Vector sensors for airborne surveillance applications. James McConnell, Scott Jensen, Thomas McCormick, and Brendan Woolrich (Appl. Physical Sci. Corp., 475 Bridge St., Groton CT 06340)

The use of vector sensors for airborne surveillance applications (e.g., frequencies below 500 Hz) is discussed with emphasis on transducers that measure the acoustic pressure-gradient. Traditional approaches such as ribbon microphones, hot-wire anemometers, and finite-difference techniques will be reviewed. The crux of the presentation concerns a discussion of diffraction type pressure-gradient microphones that go beyond the classic ribbon microphone and utilize large format membrane transduction elements comprised of piezoelectric and electret materials. The results of analytical, numerical, and experimental evaluations will be presented.

1:45

4pEA3. Estimation of sea floor properties using acoustic vector sensors. Steven E. Crocker (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841), James H. Miller, John C. Osler (NATO Undersea Res. Ctr., La Spezia, Italy), Gopu R. Potty (Univ. of Rhode Island, Narragansett, RI 02882), and Paul C. Hines (DRDC Atlantic, Dartmouth, NS B2Y3Z7, Canada)

Information in the acoustic vector field can be used to estimate properties of the environment with which the field interacts. A study was performed to understand the value of vector field data when inverting for sea floor geoacoustic properties. The study compared results obtained with new inverse methods based on measurements of complex acoustic transfer functions and specific acoustic impedance. Acoustic field data consisted of gated continuous wave transmissions acquired with four acoustic vector sensors that spanned the water-sediment interface during the Sediment Acoustics Experiment 2004 (SAX04). Motion data provided by the buried vector sensors were affected by a suspension response that was sensitive to the sediment density and shear wave speed. The suspension response for the buried vector sensors included a resonance within the analysis band of 0.6–2.4 kHz. The response was sufficiently sensitive to the local geoacoustic properties, that it was exploited by the inverse methods developed for this study. Inversions of real and synthetic data sets indicated that information about sediment shear wave speed was carried by the suspension response of the buried sensors, as opposed to being contained inherently within the acoustic vector field. [Work supported by ONR.]

2:05

4pEA4. Comparing vector- and pressure-sensor arrays: The 2009 cooperative array performance experiment. Daniel Rouseff, Russell Light (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), Zhongkang Wang, and Shihong Zhou (Hangzhou Appl. Acoust. Res. Inst., Hangzhou, China)

The 2009 cooperative array performance experiment (CAPE'09) was designed to compare performance between vector- and pressure-sensor arrays. The experiment was a joint effort of Chinese and American investigators; both arrays were designed and assembled by the Hangzhou Applied Acoustics Research Institute (HAARI), while the source systems and signal processing/recording systems were designed and assembled by Applied Physics Laboratory, University of Washington (APL-UW). The two arrays, both approximately 7 m in length, were deployed vertically off the stern of the APL-UW's R/V Robertson in Lake Washington, Seattle. Various transmitted signals in the 1.5–4 kHz band were recorded simultaneously on the two arrays at ranges between 10 m and 4 km. The signals included repeated linear frequency-modulated chirps and communications sequences. The pressure- and vector-sensor arrays had 32 and 8 uniformly spaced elements, respectively. Because each element in the vector-sensor array recorded both pressure and the three components of particle velocity, the two arrays made the same number of measurements over a similar vertical aperture. In the present talk, the design features of the vector-sensor array are emphasized. Sample results for both arrays are presented. [Work supported by ONR.]

2:25

4pEA5. Acoustic particle velocity amplification with horns. Dimitri M. Donskoy (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030)

Previously [Donskoy and Cray, J. Acoust. Soc. Am., **129**(4), Pt. 2, 2644 (2011)] the authors numerically investigated an acoustic particle velocity amplification effect with conical open-ended horns. Here, using Webster's approach, an analytical solution for the particle velocity response of conical horns (single and double) is derived and analyzed. The solutions are verified by comparison with direct numerical computations and supported with experimental measurements. It is shown that small horns, compared to the acoustic wavelength, are capable of providing substantial particle velocity amplification. For example, a 20 cm horn, in-water, can deliver nearly 10 dB of amplification over a very broad frequency range (from zero to 1000 Hz) without significant amplitude and phase distortion. Another unique feature of the velocity horn is its dipole directionality. The paper presents a thorough analysis of horn's amplification versus geometrical parameters. [This work is supported in part by the ONR summer fellowship program].

Contributed Papers

2:45

4pEA6. Analysis of the advantages and complexities of acoustic vector sensor arrays. Andrew J. Poulsen (Appl. Physical Sci. Corp., 49 Waltham St., Ste. 4, Lexington, MA 02421, apoulsen@aphysci.com) and Arthur B. Baggeroer (MIT, Cambridge, MA 02139)

The hydrophone, an omnidirectional underwater microphone, is the most common sensor for listening to underwater sound. Directional sensors, however, have many important applications. Acoustic vector sensors, one important class of directional sensors, measure acoustic scalar pressure along with acoustic particle motion. With this additional vector measurement, vector sensors feature many advantages over conventional omnidirectional hydrophone sensors: improved array gain/detection performance, enhanced bearing resolution, the ability to "undersample" an acoustic wave without spatial aliasing, and the capability of attenuating spatial ambiguity lobes, e.g., left/right ambiguity resolution for a linear array. Along with their advantages, however, vector sensors also pose additional practical complexities: greater sensitivity to non-acoustic, motion-induced flow noise at low frequencies, requisite knowledge/measurement of each sensor's orientation, management of different sensor types (pressure and particle motion) that each with different noise properties/calibration requirements, and adaptive processing can become difficult in a snapshot limited regime since each vector sensor is made up of up to four data channels. This paper will explore the virtues and limitations of vector sensor arrays in the presence of realistic ocean noise fields and system imperfections, including their effects on array performance (gain, beampatterns, etc.) supported by theoretical analysis and illustrative examples.

3:00

4pEA7. Vector wave measurements on landmine detection with an array of loudspeakers focused on the ground. Martin L. Barlett, Justin D. Gorhum, Wayne M. Wright, Mark F. Hamilton, and Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

An array of 16 loudspeakers, deployed along a segment of the base of a right circular cone, was used to focus sound on soils overlying buried targets lying along the conical axis of the source. Measurements were made at several incident angles tending toward grazing to examine long range detection for humanitarian de-mining applications. Targets and soils were instrumented with triaxial geophone and accelerometer sensors. The transmission of airborne sound into the soils produced vertical and radial vibrations in both the soil and the targets, which included rigid and compliant mine cases. Several waveform transmission types were utilized. The compliant targets provided resonances amenable to optical detection, depending on the acoustic, geometric and environmental parameters, which are discussed. [Work supported by the IRD program at ARL:UT Austin, in cooperation with the National Center for Physical Acoustics.]

3:15

4pEA8. Capacity and statistics of measured underwater acoustic particle velocity channels. Huaihai Guo, Chen Chen, Ali Abdi (Elec. & Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ 07102, hg45@njit.edu), Aijun Song, Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716), and Paul Hursky (Heat, Light, and Sound Res., Inc., La Jolla, CA 92037)

Acoustic particle velocity channels can be used for communication in underwater systems [A. Abdi and H. Guo, IEEE Trans. Wireless Communi. **8**, 3326–3329, (2009)]. In this paper, the information (Shannon) capacity of underwater acoustic particle velocity channels is studied using measured data. More specifically, the maximum achievable data rates of a compact vector sensor communication receiver and another communication receiver with spatially separated scalar sensors are compared. Some statistics of particle velocity channels such as amplitude distribution and power delay profile are investigated using measured data and proper models are suggested as well. The results are useful for design and simulation of vector sensor underwater communication systems in particle velocity channels. The work is supported in part by the National Science Foundation (NSF), Grant CCF-0830190.

Session 4pNS

Noise and Physical Acoustics: Launch Vehicle Noise II

R. Jeremy Kenny, Cochair
Marshall Space Flight Center, Bldg. 4203, Huntsville, AL 35812

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

Invited Papers

1:10

4pNS1. Overview of the Ares I scale model test program. Douglas D. Counter (Bldg. 4203, M.S. ER42, Marshall Space Flight Ctr., Huntsville, AL 35812, douglas.d.counter@nasa.gov) and Janice Houstion (Jacobs Eng., Huntsville, AL 35812)

Launch environments, such as lift-off acoustic (LOA) and ignition overpressure (IOP), are important design factors for any vehicle and are dependent upon the design of both the vehicle and the ground systems. LOA environments are used directly in the development of vehicle vibro-acoustic environments and IOP is used in the loads assessment. The NASA Constellation Program had several risks to the development of the Ares I vehicle linked to LOA. The risks included cost, schedule, and technical impacts for component qualification due to high predicted vibro-acoustic environments. One solution is to mitigate the environment at the component level. However, where the environment is too severe for component survivability, reduction of the environment itself is required. The Ares I scale model acoustic test (ASMAT) program was implemented to verify the Ares I LOA and IOP environments for the vehicle and ground systems including the mobile launcher (ML) and tower. An additional objective was to determine the acoustic reduction for the LOA environment with an above deck water sound suppression system. ASMAT was a development test performed at the Marshall Space Flight Center (MSFC) East Test Area (ETA) Test Stand 116 (TS 116). The ASMAT program is described in this presentation.

1:30

4pNS2. Ares I Scale Model Acoustic Tests Instrumentation for Acoustic and Pressure Measurements. Magda B. Vargas (Bldg. 4203, M.S. ER42, MSFC, Huntsville, AL 35812, magda.b.vargas@nasa.gov) and Douglas D. Counter (MSFC, Huntsville, AL, 35812)

The Ares I Scale Model Acoustic Test (ASMAT) was a development test performed at the Marshall Space Flight Center (MSFC) East Test Area (ETA) Test Stand 116. The test article included a 5% scale Ares I vehicle model and tower mounted on the Mobile Launcher. Acoustic and pressure data were measured by approximately 200 instruments located throughout the test article. There were four primary ASMAT instrument suites: ignition overpressure (IOP), lift-off acoustics (LOA), ground acoustics (GA), and spatial correlation (SC). Each instrumentation suite incorporated different sensor models which were selected based upon measurement requirements. These requirements included the type of measurement, exposure to the environment, instrumentation check-outs and data acquisition. The sensors were attached to the test article using different mounts and brackets dependent upon the location of the sensor. This presentation addresses the observed effect of the sensors and mounts on the acoustic and pressure measurements.

1:50

4pNS3. Measurements of the ground acoustic environments for small solid rocket motor firings. Bruce T. Vu (NASA Kennedy Space Ctr., NE-M1, KSC, FL 32899, Bruce.T.Vu@nasa.gov) and Kenneth J. Plotkin (Wyle Labs., Arlington, VA 22202, Kenneth.Plotkin@wyle.com)

During the ground launch of a space vehicle, the mobile launcher deck and tower are exposed to severe acoustic environments. These environments, if not properly managed, can weaken ground support equipment and result in structure failure. The ground acoustic environments are different than the vehicle acoustic environments. They are typically more severe because of the close proximity of the rocket plume, which often involves direct impingement. They are more difficult to predict, and their measurement and data reduction remain challenging. This paper discusses these challenges and describes the methods of processing ground acoustic data during a series of static firings of a 5-percent scale solid rocket launch vehicle and mobile launcher, known as the Ares Scale Model Acoustic Test.

2:10

4pNS4. Ares I scale model acoustic test lift-off acoustics. Douglas D. Counter (MSFC, M.S. ER42, Huntsville, AL, 35812, douglas.d.counter@nasa.gov) and Janice Houston (Jacobs Egnineering, Huntsville, AL, 35812)

The lift-off acoustic (LOA) environment is an important design factor for any launch vehicle. For the Ares I vehicle, the LOA environments were derived by scaling flight data from other launch vehicles. The Ares I LOA predicted environments are compared to the Ares I scale model acoustic test (ASMAT) preliminary results.

2:30

4pNS5. Ares I Scale Model Acoustic Test above deck water sound suppression results. Douglas D. Counter (MSFC, M.S. ER42, Huntsville, AL 35812, douglas.d.counter@nasa.gov) and Janice Houston (Jacobs Eng., Huntsville, AL 35812)

The Ares I Scale Model Acoustic Test (ASMAT) program test matrix was designed to determine the acoustic reduction for the LOA environment with an above deck water sound suppression system. The scale model test can be used to quantify the effectiveness of the water suppression system as well as to optimize the systems necessary for LOA noise reduction. Several water flow rates were tested to determine which rate provides the greatest acoustic reductions. Preliminary results are presented.

2:50

4pNS6. Recovering the spatial correlation of liftoff acoustics from the Ares Scale Model Acoustics Test. Brian Prock, Paul Bremner (ATA Eng., Inc.), and Thomas L. Philley (NASA JSC)

Accurately predicting structural vibrations due to acoustic loads requires knowledge about the overall sound pressure levels, the amplitude of the sound pressure levels as a function of frequency (auto-spectra), and the spatial correlation of the sound pressure levels (cross-spectra). When dealing with the liftoff acoustics of launch vehicles, a large amount of historical data is available in terms of overall levels and auto-spectra, but only a limited amount of data exists for cross-spectra. ATA Engineering has used data taken during NASAs Ares Scale Model Acoustic Test to recover the spatial correlation of liftoff acoustics for a typical launch vehicle. By assuming the measured liftoff acoustics that are a combination of propagating waves and diffuse acoustic fields, curve-fitting algorithms are used to recover spatial correlation parameters required by modern vibro-acoustic analysis software.

3:10–3:30 Break

3:30

4pNS7. 5% Ares I scale model acoustic test: Overpressure characterization and analysis. David Alvord (Jacobs Eng., M.S. ER42, MSFC, Huntsville, AL 35812, david.alvord@nasa.gov), Matthew Casiano, and David McDaniels (MSFC, Huntsville, AL, 35812)

During the ignition of a ducted solid rocket motor (SRM), rapid expansion of injected hot gases from the motor into a confined volume causes the development of a steep fronted wave. This low frequency transient wave propagates outward from the exhaust duct, impinging the vehicle and ground structures. An unsuppressed overpressure wave can potentially cause modal excitation in the structures and vehicle, subsequently leading to damage. This presentation details the ignition transient findings from the 5% Ares I scale model acoustic test (ASMAT). The primary events of the ignition transient environment induced by the SRM are the ignition overpressure (IOP), duct overpressure (DOP), and source overpressure (SOP). The resulting observations include successful knockdown of the IOP environment through use of a space shuttle derived IOP suppression system, a potential load applied to the vehicle stemming from instantaneous asymmetrical IOP and DOP wave impingement, and launch complex geometric influences on the environment. The results are scaled to a full-scale Ares I equivalent and compared with heritage data including Ares I-X and both suppressed and unsuppressed space shuttle IOP environments.

3:50

4pNS8. Simulation of acoustics for Ares I scale model acoustic tests. Gabriel C. Putnam and Louise L. Strutzenburg (Marshall Space Flight Ctr., MSFC / ER42, Huntsville, AL, 35812)

The Ares I scale model acoustics test (ASMAT) is a series of live-fire tests of scaled rocket motors meant to simulate the conditions of the Ares I launch configuration. These tests have provided a well documented set of high fidelity acoustic measurements useful for validation including data taken over a range of test conditions and containing phenomena like ignition over-pressure and water suppression of acoustics. To take advantage of this data, a digital representation of the ASMAT test setup has been constructed and test firings of the motor have been simulated using the LOCI/CHEM computational fluid dynamics software. Results from ASMAT simulations with the rocket in both held down and elevated configurations, as well as with and without water suppression have been compared to acoustic data collected from similar live-fire tests. Results of acoustic comparisons have shown good correlation with the amplitude and temporal shape of pressure features and reasonable spectral accuracy up to approximately 1000 Hz. Major plume and acoustic features have been well captured including the plume shock structure, the igniter pulse transient, and the ignition overpressure.

4:10

4pNS9. Hybrid computational fluid dynamics and computational aero-acoustic modeling for liftoff acoustic predictions. Louise L. Strutzenberg (MSFC, M.S. ER42, MSFC, Huntsville, AL 35812, louise.s@nasa.gov) and Peter A. Liever (CFD Res. Corp., Huntsville, AL 35812)

This paper presents development efforts at the NASA Marshall Space flight Center to establish a hybrid computational fluid dynamics and computational aero-acoustics (CFD/CAA) simulation system for launch vehicle liftoff acoustics environment analysis. Acoustic prediction engineering tools based on empirical jet acoustic strength and directivity models or scaled historical measurements are of limited value in efforts to proactively design and optimize launch vehicles and launch facility configurations for liftoff acoustics. CFD based modeling approaches are now able to capture the important details of vehicle specific plume flow environment, identify the noise generation sources, and allow assessment of the influence of launch pad geometric details and sound mitigation measures such as water injection. However, CFD methodologies are numerically too dissipative to accurately capture the propagation of the acoustic waves in the large CFD models. The hybrid CFD/CAA approach combines the high-fidelity CFD analysis capable of identifying the acoustic sources with a fast and efficient boundary element method (BEM) that accurately propagates the acoustic field from the source locations. The BEM approach was chosen for its ability to properly account for reflections and scattering of acoustic waves from launch pad structures. The paper will present an overview of the technology components of the CFD/CAA framework and discuss plans for demonstration and validation against test data.

4:30

4pNS10. Validation study on computational aeroacoustics of acoustic waves from sub-scale rocket plumes. Seiichiro Morizawa (Dept. of Aerosp. Engineering, Tohoku Univ., 3-1-1 Yoshinodai, Chuo-ku, Sagamihara 25205210, morizawa@edge.ifs.tohoku.ac.jp), Taku Nonomura (Inst. of Space and Astronautical Sci., Sagamihara 2525210), Seiji Tsutsumi (JAXA's Eng. Deigital Innovation, JAXA, Sagamihara 2525210), Nobuhiro Yamanishi, Keita Terashima (Space Transportation Mission Directorate, Tsukuba 3058505), Shigeru Obayashi (Tohoku Univ., 2-1-1 Katahira Aoba-ku Sendai, 9808577), and Kozo Fujii (Inst. of Space and Astronautical Sci., Sagamihara 2525210)

In this paper, the comparative study of prediction based on computational aeroacoustics (CAA) and experimental results for acoustic waves from modeled rocket motors is conducted, and prediction accuracy of CAA is discussed in the framework of JAXA-CNES collaboration. Experimental data of flow and acoustic fields of solid motor by JAXA and H2-AIR liquid motor by CNES are used as the reference. Two types of computational codes are adapted in this study. The predictions of sound pressure level by both computational codes agree reasonably with corresponding experimental data, whereas the errors are approximately less than 5 dB. In addition, each aeroacoustic field of CAA results in this study is discussed in detail.

Contributed Papers

4:50

4pNS11. Measurement and propagation of supersonic aeroacoustic noise sources using continuous scanning measurement technologies and the fast multipole boundary element method. Michael Y. Yang, Havard Vold, and Parthiv N. Shah (11995 El Camino Real, San Diego, CA 92130)

ATA Engineering has developed a technique which uses a continuous scanning robot to take high-resolution measurements of supersonic jet plumes. The jet noise was modeled using a reduced-order model and propagated to far field microphone locations in the free-field. It is shown that the pressure at these microphones was successfully reconstructed across a range of frequencies. The capability to make predictions when scattering surfaces are present is also demonstrated using the fast multipole boundary element method in VA One. This work was originally designed for supersonic jets but can also be used for static firing tests of launch vehicle engines. The measured data could then be used for analytic predictions of the liftoff environment.

5:05

4pNS12. On the jet-wake similarity. Ballard W. George (1367 Boblink Circle, Sunnyvale, CA 94087)

This paper is concerned with similarities and differences between jets and wakes, as indicated by a sampling of the literature, including what tests and studies have been made in each case. As noted by Franken, jets and wakes are both characterized by a region of shear and (depending on the Reynolds number) high turbulence. Jet noise has been extensively studied under the impetus of a strong concern with community noise. Jet noise studies have examined numerous quantities including, for example, spectra, directivity, source location, and total power. Lighthill published widely quoted papers on sound generated aerodynamically, primarily in the context of jet noise and without solid boundaries. Literature referred to for this paper was primarily for airborne sound and included underwater sound, in which case cavitation plays a significant role and tends to mask turbulent wake noise due to the vehicle body. Propellers are a source of wake as well as lift noise. A readily noticeable difference between jets and wakes involves the fact that jets can be tested statically, while for wakes there has to be relative motion. Also jets, though not jet engines, can be tested "cold."

THURSDAY AFTERNOON, 3 NOVEMBER 2011 ESQUIRE, 1:55 TO 4:45 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Perceptual Aspects of Sound

Elizabeth A. Strickland, Chair
Dept. of Speech, Language, and Hearing Sciences, Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—1:55

Contributed Papers

2:00

4pPP1. Mistuned harmonic detection and the role of tonotopically local neural synchrony. William M. Hartmann (Depatment of Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824)

A mistuned harmonic in an otherwise periodic complex tone is a simple stimulus for the study of segregation and integration of complex tones by human listeners. In a mistuned harmonic *detection* experiment, a listener must discriminate between two tones—one perfectly periodic and the other with a mistuned harmonic. This detection paradigm is subject to known artifacts. However, with careful experiment design the artifacts can be controlled, and the paradigm becomes an efficient way to explore the emergence of segregated mistuned spectral components. The results of mistuned harmonic detection experiments support a tonotopically local model in which detection is mediated by the dephasing of a neural spike sequence in a tonotopically tuned channel. Evidence for this model comes from (1) the functional dependences of detectability on the amount of mistuning and the tone duration, (2) the non-monotonic level dependence of detectability, (3) the lowpass character of detection—indicating an essential role for neural synchrony, and (4) the need for tonotopically local interaction as evidenced by experiments using mistuned distracters, mistuned harmonics in spectral gaps, and dichotic presentation. [Work supported by the NIDCD and the AFOSR.]

2:15

4pPP2. Parametric issues in measuring the olivocochlear reflex with a masking technique. Elin M. Roverud and Elizabeth A. Strickland (Dept. of Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr. West Lafayette, IN 47907, eroverud@purdue.edu)

Previous studies in this laboratory have suggested that forward masking occurs by two mechanisms, excitatory masking and gain reduction via the medial olivocochlear reflex (MOCR), that operate over different time courses. In this laboratory, a forward masking technique is used in which a precursor, intended to elicit the MOCR, is followed by a fixed off-frequency forward masker and signal. A prior study [Roverud and Strickland, J. Acoust. Soc. Am. **128**, 1203–1214] found that for short precursors, signal threshold increased then decreased (buildup) as the signal and masker were delayed from the precursor. This result could be consistent with the sluggish onset of the MOCR. When the masker is removed, however, buildup is not observed as the signal is delayed from a precursor, even though the precursor should still be eliciting the MOCR. This raises the question, what is the role of the masker and why is it necessary to observe buildup? The current study examines the frequency and level characteristics of the masker required to observe the buildup effect. Results will be discussed in terms of gain reduction and temporal window models of forward masking. [Research supported by a Grant from NIH(NIDCD) R01 DC008327.]

2:30

4pPP3. Perceiving auditory distance using level and direct-to-reverberant ratio cues. Andrew J. Kolarik, Silvia Cirstea, and Shahina Pardhan (VERU, Eastings 204, Anglia Ruskin Univ., East Rd., Cambridge, CB1 1PT, United Kingdom, andrew.kolarik@anglia.ac.uk)

The study investigated how level and reverberation cues contribute to distance discrimination, and how accuracy is affected by reverberation cue strength. Sentence pairs were presented at distances between 1 and 8 m in a virtual room simulated using an image-source model and two reverberation settings (lower and higher). Listeners performed discrimination judgments in three conditions: level cue only (Level-Only), reverberation only (Equalized), and both cues available (Normal). Percentage correct judgment of which sentence was closer was measured. Optimal distance discrimination was obtained in the Normal condition. Perception of the difference in distance between sentences had a lower threshold (i.e., performance was significantly better, $p<0.05$) for closer than further targets in Normal and Level-Only conditions. On the contrary, in the Equalized condition, these thresholds were lower for further than closer targets. Thresholds were lower at higher reverberation in the Equalized condition, and for further targets in the Normal condition. Data indicate that level generally provided more accurate discrimination information than direct-to-reverberant ratio. Direct-to-reverberant ratio provided better information for sounds further from the listener than for nearer sounds, and listeners were able to use direct-to-reverberant ratio as effectively as level in highly reverberant rooms when discriminating far sound sources.

2:45

4pPP4. Infants' vowel discrimination in modulated noise. Lynne A. Werner (Dept. Speech & Hearing Sci., Univ. Washington, 1417 NE 42nd St., Seattle WA 98105-6246)

Adult listeners detect and discriminate target sounds better in amplitude modulated noise than in unmodulated noise. This study examined infants' ability to take advantage of masker modulation to improve sensitivity to a target. Listeners were 7–9-month-old infants and 18–30-year-old adults. Vowel discrimination in noise was tested. Listeners learned to respond when a repeated vowel changed from /a/ to /i/ or from /i/ to /a/. An observer-based method was used to assess sensitivity to the vowel change. The maskers were speech-spectrum noise either unmodulated, amplitude modulated with the envelope of single-talker speech or sinusoidally amplitude modulated at 8 Hz with a 75% modulation depth. The overall level of the maskers was 60 dB SPL. The level of the vowels, chosen to yield an average d' of 1 in the unmodulated masker, was 46 dB SPL for adults and 58 dB SPL for infants. Adults' d' was substantially and significantly higher in both modulated maskers than in the unmodulated masker. Infants' d' was also significantly higher in the two modulated maskers than in the unmodulated masker, but the improvement due to modulation was significantly smaller for infants than for adults. [Work supported by NIDCD R01DC00396 and P30DC04661.]

3:00

4pPP5. Effects of silent interval on human frequency-following responses to voice pitch. Fuh-Cherng Jeng and Ronny P. Warrington (Commun. Sci. and Disord., Ohio Univ., 1 Ohio Univ. Dr. Athens, OH 45701, jeng@ohio.edu)

Human frequency-following responses (FFRs) to voice pitch have provided valuable information on how the human brain processes speech information. Recordings of the FFR to voice pitch, however, may overlap when insufficient silent intervals are used. To determine the shortest silent interval that can be used with no overlap between adjacent response waveforms, FFRs were recorded from 12 Chinese adults using a wide range of silent intervals. The stimulus token was a Chinese monosyllable with a rising pitch of 117–166 Hz and a duration of 250 ms. A high stimulus intensity at 70 dB SPL was used to maximize overlaps in the response waveforms. A total of seven silent intervals, ranging from the full length of the stimulus duration down to approximately half period of the fundamental frequency of the stimulus token, were administered at a random order across participants. Two distinct methods (Hilbert transform and root-mean-square amplitudes) were used to delineate the envelopes and overlaps of the response waveforms. A one-way repeated measures analysis of variance was significant ($p = 0.038$) in defining the magnitude of overlaps for the 10 ms pre-stimulus interval. The results indicated the shortest silent interval that could be used without compromising the response is between 35 and 45 ms.

3:15

4pPP6. Psychoacoustics of chalkboard squeaking. Christoph Reuter (Musicological Inst., Univ. of Vienna, Vienna, Austria) and Michael Oehler (Univ. of Cologne, Cologne, Germany)

At least since 1975 the "pleasantness" of a sound is discussed from many different angles (Ely 1975; Aures 1984; Halpern *et al*. 1986; Vaschillo 2003; Neumann & Waters 2006; Cox 2008), but often chalkboard squeaking or scratching a chalkboard with finger nails tops the list of unpleasant sounds. The aim of the presented study is to detect specific parts of the sounds that make chalkboard squeaking particularly unpleasant. With a combination of perception experiments and electro-physiological measurements, it was analyzed to what extent the knowledge about the sounds influenced the subjects' judgments and/or the physiological reactions. Basically the study is a replication of Halpern *et al*. (1986), whose methods were extended by several sophisticated sound analysis and re-synthesis techniques and the measurement of some electro-physiological parameters (heart rate and skin resistance) during listening. First results show that especially the modification of the tonal parts as well as applying a filter between 2000 and 4000 Hz led to a more pleasant sound perception. Almost all stimuli were rated more unpleasant if the subjects knew about the nature of the sounds.

3:30–3:45 Break

3:45

4pPP7. The shopping sound experience. Ralf Jung, Gerrit Kahl, and Lbomira Spassova (DFKI, Campus D 3 4, 66123 Saarbrücken 66123 Germany, Ralf.Jung@dfki.de)

The attempt to influence the shopping behavior of customers in a supermarket through music often fails due to the different music preferences. In this work, a method to bring personalized music in the supermarket is presented by providing a location-aware playback and notification service. By using a web interface, customers are able to create an electronic shopping list, associate list items with music tracks, and thus create an individual shopping playlist. These product-associated music tracks start playing at the customer's instrumented shopping cart when he enters specific product departments where an item from the shopping list is located. Additionally, a service to provide product-awareness through non-speech audio cues is presented. This location-aware service notifies customers when they come closer to products that are listed on their shopping list. A two-stage notification approach is used to create user-centric notification zones. Depending on the customer's distance to the product, he gets notified by an ambient or arousal noise that is mixed into the background music. In addition, further information about the detected product is displayed on the screen that is mounted at our instrumented shopping cart.

4:00

4pPP8. Psychological factors influencing the evaluation of electric vehicle interior noise. Jochen Steffens (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Dsseldorf, Germany, jochen.steffens@fh-duesseldorf.de), Thomas Kueppers (Daimler AG, 70372 Stuttgart, Germany), and Sabrina Skoda (Duesseldorf Univ. of Appl. Sci., 40474 Dsseldorf, Germany)

Increasing global awareness of the benefits of electromobility has brought about the need for new concepts in terms of the acoustic design of future vehicle generations. This includes both the creative design process and the development of suitable methods for the subjective evaluation of target sounds. The main difference between e-car sound surveys and those carried out on familiar sound categories is the potential consumer's lack of experience with electric vehicles. Thus, the consumer has no, or very unspecific, expectations in this regard. Several studies have shown that many subjects have to construct their personal frame of reference for evaluation within the listening experiment. However, this is possibly at odds with experience-based expectations relating to sounds of conventional combustion engines. The result is a conflict of objectives between the traditional and the modern, familiarity and strangeness, and not least between driving freedom and ecological awareness. In this context, the authenticity of the sound and the subjective interpretability of the sound information also appear as moderator variables. Moreover, associations with other vehicle categories, for example, streetcars, also influence the perceived sound quality. Within this contribution, these factors will be expounded and their influence on the evaluation of interior noise discussed.

4:15

4pPP9. Auditory-proprioceptive interaction—How do acceleration forces influence the evaluation of driving sounds? Sabrina Skoda, Jochen Steffens (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany, sabrina.skoda@fh-duesseldorf.de), Thomas Kueppers (Daimler AG, Mercedesstrasse 122, 70372 Stuttgart, Germany), and Joerg Becker-Schweitzer (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany)

One fundamental requirement for a widespread use of electric powered vehicles is a high degree of social acceptance of alternative drive concepts. As part of this, perception and evaluation of comfort and quality in a vehicle become increasingly important. The customers' judgment on these factors is strongly influenced by the noise and vibration behavior of the vehicle and always is passed in the context of multiple sensory impressions which are processed in the human brain consciously and unconsciously. The interaction mechanisms of sensory perception are highly complex and raise several scientific questions. During the past years, valuable insights about the interaction of visual and auditory perception have been obtained and there are also a number of theories about the auditory-tactile interaction. In contrast, the connection between auditory and proprioceptive perception remains largely unexplored. This paper deals with the question how acceleration forces influence human auditory perception. In order to make evident possible crossmodal effects, a listening experiment was conducted in different kinds of driving simulators. The results of this study will be presented and discussed.

4:30

4pPP10. Structurized sound design process of electric vehicle interior sound. Thomas Kueppers (Daimler AG, NVH Electric Powertrain, Stuttgart, D-70372 Germany, thomas.kueppers@daimler.com), Jan-Welm Biermann (IKA, RWTH Aachen Univ.), and Jochen Steffens (SAVE Inst., FH Duesseldorf Univ.)

The sound character of electric vehicles extremely differs from internal combustion engine vehicles. While the electric vehicle development is still in the beginning, there is potential to consider customer expectations very early in the development process, to have an impact on the typical desired electric vehicle sound on market launch and to generate unique selling propositions in competition between vehicle manufacturers. The current electric powertrain noise is perceived as inconvenient and nonaesthetical especially on long journeys. A complete reduction of this powertrain noise emphasizes the perception of wind and rolling noise and avoids feedback of velocity and load dependency, which decreases emotional impression. Futuristic sounds by sound design systems have to be evaluated by customers and revised in their acceptance and authenticity. Besides the improvement of the sound character, sound design systems can additionally be used to influence the informative acoustic feedback of driving and vehicle parameters. This article introduces the upcoming scenarios for interior sound development and focuses the contradiction between a great amount of freedom in design and a structurized workflow for designing electrical powertrain sounds.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 4/5, 1:30 TO 3:20 P.M.

Session 4pSCa

Speech Communication: Forensic Acoustics—On the Leading Edge of the Tidal Wave of Change About to Hit Forensic Science in the US? II

Geoffrey Stewart Morrison, Chair
School of Electrical Engineering, Univ. of New South Wales, Sydney, NSW, 2052, Australia

Chair's Introduction—1:30

Contributed Papers

1:35

4pSCa1. Selection of speech/voice vectors in forensic voice identification. James Harnsberger (Dept. Linguist., U. Florida, Gainesville, FL 32607) and Harry Hollien (Instit. Adv. Study Comm. Proc., U. Florida, Gainesville, FL 32611)

The case for the use of speech/voice vectors in speaker identification was made by Hollien and Harnsberger (2010). Those vectors found most robust in capturing speaker-specific characteristics were voice quality, vowel quality, speaking fundamental frequency, and temporal features. In this study, the speech cues for each of the four vectors will be compared to each other with respect to their predictive power. In addition, different vector algorithms and/or processing approaches for each will be contrasted in terms of their effects on identification robustness. One example is conversion of the vowel formant frequencies and bandwidth measurements to geometric scaling (semits). Finally, a second dataset of 18 male voices obtained from evidence recordings, paired with exemplars recorded from a speaker pool, were used to test a modified form of this speech/voice vector approach. The

data from these subjects will be compared to those from the 1993 study (substantial improvement) and those from the 2010 experiment (confirmation).

1:50

4pSCa2. When to punt on speaker comparison? Reva Schwartz (U.S. Secret Service, Forensic Services Div. 950 H St. NW, Ste. 4200, Washington, DC 20223), Joseph P. Campbell, and Wade Shen (MIT Lincoln Lab., Lexington, MA, 02420)

In forensic speaker comparison, it is crucial to decide when completion of the examination may not possible (punt). We explore the factors that make speaker comparison decisions difficult or impossible. These factors may include duration, noise, speaking style, language/dialect, mental state, number of speakers, type and quality of recording, and deception. The analyst needs criteria to decide to reject case work. We present analysis of some of these factors and their impact on automatic speaker recognition systems. We propose a methodology for setting objective thresholds by which comparison examples can be rejected. This methodology could be used by forensic analysts to decide whether or not to proceed with speaker comparisons involving these factors.

2:05

4pSCa3. Defining the default defense hypothesis in likelihood-ratio forensic voice comparison. Felipe Ochoa and Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecom., Univ. of New South Wales, Sydney, New South Wales, Australia)

In forensic DNA comparison, the person submitting samples for evaluation does not know what properties the samples will have when they are analyzed at the laboratory; samples are submitted as a matter of routine. In contrast, in forensic voice comparison the decision to submit samples for evaluation is based on prior screening: Typically a police officer, a lay person with respect to forensic voice comparison, has listened to the questioned-speaker recording and the known-speaker recording and decided that they sound sufficiently similar that they could be the same speaker and merit evaluation by a forensic scientist. If they do not sound sufficiently similar they are not submitted for evaluation. Unless the defense proposes a more restrictive hypothesis, the forensic scientist should therefore adopt the following as the default defense hypothesis and select a background database accordingly: The known speaker is not the same person as the questioned speaker but is one member of a population of speakers whom to a lay person sound sufficiently similar to the voice on the questioned-voice recording that they would submit recordings of these speakers for forensic comparison with the questioned-voice recording. Examples of how this theory might be applied are discussed.

2:20–2:35 Break

2:35

4pSCa4. Human error rates for speaker recognition. Wade Shen, Joseph P. Campbell (MIT Lincoln Lab., 244 Wood St, C-290A, Lexington, MA, 02420), and Reva Schwartz (U.S. Secret Service, Forensic Services Div. Ste. 4200, Washington, DC 20223)

It is commonly assumed that speaker identification by human listeners is an innate skill under certain conditions. As such, human listening tests have served as the benchmark for automatic recognition systems. In recent evaluations comparing human and machine performance on a speaker comparison task, error rates of naïve human listeners far exceed those of machines [special session on Human Assisted Speaker Recognition, IEEE ICASSP, Prague, 2011]. In this presentation, we quantify the performance of naïve listeners in a variety of challenging channel conditions and we compare these results against automatic systems and trained human listeners. The results of these experiments impact the admissibility of both forensic voice analysis and courtroom testimony by human listeners.

2:50

4pSCa5. Investigating the acoustic and phonetic correlates of deceptive speech. C. Kirchhuebel (Dept. of Electronics, Audio Lab., Univ. of York, Heslington, York, UK YO10 5DD, ck531@york.ac.uk)

The following study describes an initial investigation into the acoustic and phonetic correlates of deceptive speech using auditory and acoustic analysis. Due to the lack of extant data suitable for acoustic analysis, a laboratory-based experiment was designed which employed a mock-theft paradigm in conjunction with a "security interview" to elicit truthful and deceptive speech as well as control data from a total of ten male native British English speakers. Using PRAAT, the control, truthful, and deceptive speech samples were analyzed on a range of speech parameters including f0 mean and variability, intensity, vowel formant frequencies, and speaking/articulation rate. Preliminary analysis suggests that truth-tellers and liars cannot be differentiated based on these speech parameters. Not only was there a lack of significant changes for the majority of parameters investigated but also, if change was present it failed to reveal consistencies within and between speakers. The remarkable amount of inter and intra-speaker variability underlines the fact that deceptive behavior is individualized and very multifaceted. As well as providing a basis for future research programs, the present study should encourage researchers and practitioners to evaluate critically what is (im)possible using auditory and machine based analyses with respect to detecting deception from speech.

3:05

4pSCa6. Progress toward a forensic voice data format standard. James L. Wayman (Office of Graduate Studies and Res., San Jose State Univ., San Jose, CA 95192-0025), Joseph P. Campbell, Pedro Torres-Carrasquillo (MIT Lincoln Lab., Lexington, MA 02420-9108), Peter T. Higgins (Higgins and Assoc., Int., 3900-A, #7E, Watson Pl. NW, Washington, DC 20016), Alvin Martin (Information Access Div. Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8940), Hirotaka Nakasone (Operational Technol. Div. Federal Bureau of Investigation, Quantico, VA 222135), Craig Greenberg, and Mark Pryzbocki (Information Access Div. Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8940)

The de facto international standard for the forensic exchange of data for biometric recognition is ANSI/NIST ITL-1/2, "Data Format for the Interchange of Fingerprint, Facial, and Other Biometric Information." This format is used by law enforcement, intelligence, military, and homeland security organizations thoughout the world to exchange fingerprint, face, scar/mark/tatoo, iris, and palmprint data. To date, however, there is no provision within the standard for the exchange of audio data for the purpose of forensic speaker recognition. During the recent 5-year update process for ANSI/NIST ITL-1/2, a consensus decision was made to advance a voice data format type under the name "Type 11 record." Creating such an exchange format type, however, is far from straight forward—the problem being not the encoding of the autio data, for which many accepted standards exist, but rather in reaching a consensus on the metadata needed to support the varied mission requirements across the stakeholder communities. In this talk, we'll discuss the progress that has been made to date, the questions that remain, and the requirements for additional input from the broader stakeholder communities.

Session 4pSCb

Speech Communication: Prosody, Articulation, and Articulatory Modeling (Poster Session)

Meghan Sumner, Chair
Dept. of Linguistics, Stanford Univ., Stanford, CA 94305-2150

Contributed Papers

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

4pSCb1. Reduction of consonants and vowels in the course of discourse. Michael McAuliffe and Molly Babel (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, mcauliff@interchange.ubc.ca)

There is a clear link between the discourse status of a word and the degree of reduction. For instance, Gregory [Dissertation (2002)] provided evidence that hearer knowledge affected reduction in production for discourse-old items. Lexical information, such as word frequency, also plays a crucial role in the degree of reduction [Fosler-Lussier and Morgan, Speech Commun. (1999)]. These previous studies either looked at short discourses or words isolated from context. Therefore, the current study investigates longer discourses, using the VIC Corpus [Pitt *et al*., Corpus (2007)]. The primary question is to what degree do repeated uses cause further reductions, and if the reductions are syllabically and segmentally uniform across a word. Many studies on reduction rely on the intuition that reductions occur when information load is light, such as when the word was repeated recently or has a high probability of occurrence, so the prediction is that unstressed syllables would show more reduction than stressed syllables, due to their lighter load of information. Likewise, as vowels vary more than consonants across dialects, the informational load of vowel quality may be less than that of consonant quality, so the prediction is that vowels would show greater reductions than consonants.

4pSCb2. Automatic analysis of constriction location in singleton and geminate consonant articulation using real-time magnetic resonance imaging. Christina Hagedorn, Michael Proctor, Louis Goldstein, and Shrikanth Narayanan (Dept. of Eng., USC, Los Angeles, CA 90089, shri@sipi.usc.edu)

Research on geminate consonants has attempted to establish whether the control of their articulation differs from that of corresponding singletons in temporal parameters, spatial parameters, or both. One piece of evidence supporting spatial control in Italian geminates is EPG results revealing that the location of maximal constriction (CL) of coronal geminates along the palate exhibits small differences from the CL of singletons (Payne 2005). However, our recent work investigating Italian using real-time magnetic resonance imaging (MRI) has shown that when measuring CL dynamically, the CL of singletons and geminates are identical. Dynamic CL is defined as the region of the image that exhibits maximum intensity change during constriction formation and release. These results can be reconciled with the EPG findings if we hypothesize that differences in CL at the moment of maximal constriction are due to compression effects (in the longer geminates) involving the tongue tip sliding along the palate during the closure duration. We evaluate this hypothesis by testing whether CL differences between singletons and geminates can also be found in MRI data, when CL is measured statically (region on the palate contacted in the most constricted frame) in the same utterances in which dynamic CL is invariant. [Work supported by NIH.]

4pSCb3. Development of speech motor control: Lip movement variability. Anders Lfqvist (Dept. Logopedics, Phoniatrics and Audiol., Lund Univ., Lund, Sweden), Johan Frid, and Susanne Schötz (Humanities Lab., Lund Univ., Lund, Sweden)

This study examined variability of lip movements across repetitions of the same utterance as a function of age in Swedish speakers. Subjects were 37 typically developed Swedish children and adults (19 females, 18 males, aged 5–31 yr). Lip movements were recorded during 15–20 repetitions of a short Swedish phrase using articulography, with a sampling rate of 200 Hz. After correction for head movements, the kinematic records were expressed in a maxilla-based coordinate system. Movement onset and offset of the utterance were identified using kinematic landmarks. The Euclidean distance between receivers on the upper and lower lips was calculated and subjected to functional data analysis [Ramsay *et al*., J. Acoust. Soc. Am. **99**, 3718–3727 (1996)] to assess both temporal and spatial variabilities. Results show a decrease in both indices as a function of age, with a greater reduction of amplitude variability. [Work supported by grant 349-2007-8695 from the Swedish Research Council.]

4pSCb4. Imaging and quantification of glottal kinematics with ultrasound during speech. Benjamin Parrell (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089), Adam Lammert (Univ. of Southern California, Dept. of Comput. Sci. 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089), Louis Goldstein, Dani Byrd (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089), and Shrikanth Narayanan (Dept. of Elec. Eng., Univ. of Southern California, Los Angeles, CA 90089)

Most examinations of glottal abduction and adduction during speech have employed laryngoscopic video or transillumination (Hoole, 2006). While these provide accurate information about timing of glottal movements, they are invasive and cannot provide absolute measurements about glottal width. At the same time, recent medical studies have used ultrasonic imaging to accurately capture glottal movements and laryngeal anatomy [Hu, J. Ultrasound Med. (2010); Jadcherla *et al*., Dysphagia (2006)]. We demonstrate novel methods of using ultrasound to measure both temporal and spatial aspects of glottal movements during speech. While previous work on glottal ultrasound has been limited by the need to manually analyze each acquired frame, we present methods to automatically quantify glottal aperture in ultrasound images. Finally, since glottal ultrasound does not interfere with the acquisition of supra-laryngeal articulatory data, we present the results of preliminary experiments that record laryngeal and supra-laryngeal speech movements simultaneously using ultrasound concurrently with electromagnetic articulometry. This allows the analysis of relative timing between movements in the two systems as well as examination of changes in timing or magnitude due to variables such as prosodic structure or speech rate. [Work supported by NIH.]

4pSCb5. Detailed study of articulatory kinematics of critical articulators and dependent articulators of emotional speech. Jangwon Kim (Univ. of Southern California, 3740 McClintock Ave., EEB 400, Los Angeles, CA 90089, jangwon@usc.edu), Sungbok Lee (Univ. of Southern California Los Angeles, CA 90089, sungbokl@usc.edu), and Shrikanth Narayanan (Univ. of Southern California, Los Angeles, CA 90089)

This study investigates the articulatory kinematics of critical articulators and dependent articulators as a function of emotion. Our hypothesis is that critical articulators and dependent articulators are utilized differently for achieving distinctive emotion goals that overlay linguistic goals. For example, speakers may use variability of dependent articulators distinctively to that of critical articulators in achieving emotion goals. Distinctive articulatory movements for different emotions have been observed (Lee *et al*. 2005). This study uses a database of three speakers (2 female and 1 male) collected with electromagnetic articulography to collect kinematic information. Articulatory trajectories are aligned by dynamic time warping. Linguistically identical syllable-level segments are analyzed based on detailed aspects of articulatory movements (e.g., position, velocity and phase), after sampling with a 20-ms window and 10-ms shifting. The emotion-specific patterns that emerge for critical (i.e., goal-directed) articulators are compared to those for articulators that are not under overt control for a particular phone. [Work supported by NIH and NSF Grants.]

4pSCb6. Statistical estimation of speech kinematics from real-time MRI data. Adam Lammert, Vikram Ramanarayanan (Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089), Louis Goldstein, Khalil Iskarous (Univ. of Southern California, Los Angeles, CA 90089), Elliot Saltzman (Boston Univ., Boston, MA 02215), Hosung Nam (Haskins Labs., New Haven, CT 06511), and Shrikanth Narayanan (Univ. of Southern California, Los Angeles, CA 90089)

The human speech production system can be fundamentally characterized by the kinematic relationships between low-level articulator variables and relatively high-level tasks. Such kinematics can be illuminating about many system aspects from degrees of freedom and redundancy to dynamics and even control. Since these relationships are generally complex and infeasible to express in closed form, recent work has focused on statistical methods for estimating the relevant relationships from data (Saltzman, 2006, in Dynamics of Speech Production and Perception; Lammert, 2010, Proc. INTERSPEECH). Such methods have been applied to synthetic speech data in order to evaluate their effectiveness, but they have not yet been demonstrated on real data. Here, we apply these methods to real speech data acquired from real-time magnetic resonance imaging. We extract articulator variables that are consistent with those chosen by various articulatory models, and we relate them to high-level task variables such as constriction locations and degrees and formant frequencies. This is done to facilitate an analysis of articulatory motor control during speech production. [Work supported by NIH.]

4pSCb7. Semi-automatic modeling of tongue surfaces using volumetric structural MRI. Daniel K. Bone, Michael I. Proctor, Yoon Kim, and Shrikanth S. Narayanan (Univ. of Southern California, Signal Analysis and Interpretation Lab., Los Angeles, CA 90089)

Although volumetric magnetic resonance imaging has proven to be a valuable tool in the study of consonant production (Narayanan *et al*., 1995; Kröger *et al*., 2000), its utility is limited by the difficulty and laboriousness of reliably extracting tissue boundaries from imaging data. Current methods typically involve manual segmentation of air-tissue boundaries (e.g., Birkholz, 2006). Conventional automated (Atkins, 1998) and semi-automated (Ashton *et al*., 1995) methods used for the segmentation of brain MRI datasets may not be directly applicable to lingual segmentation because they are designed to work with different anatomical features. We present a method for extracting tongue surfaces from hi-resolution volumetric MRI data with limited user intervention. For each vocal tract volume to be analyzed, a lingual bounding box and search seed was first specified by an expert user, and voxel intensity was normalized across the region of interest. Lingual surfaces were automatically identified using a multi-pass region-growing algorithm operating over coronal planes. Thresholding was performed asymmetrically to allow for differential detection of air, teeth, and palatal boundaries, in opposition to adjacent lingual tissue. Smoothed tongue surfaces were fit to the resulting volumes by incorporating prior knowledge of intrinsic lingual musculature. [Work supported by NIH.]

4pSCb8. Enhancement of laryngeal features under segmental and prosodic conditioning. Indranil Dutta (Dept. of Computational Linguisitics, EFL Univ., Tarnaka, Hyderabad, Andhra Pradesh, 500007 India)

Evidence in support of enhancement features (Keyser and Stevens, 2006) is presented from an acoustic study of laryngeal contrasts in Hindi. Segmental contrasts where defining features and their corresponding acoustic outcomes are attenuated are said to be made acoustically salient by enhancing gestural features (Stevens and Keyser, 2010). In this study, the four-way laryngeal contrasts in Hindi are examined under varying prosodic and segmental contexts. In segmental contexts where a defining distinctive feature (slack vocal folds) is attenuated in its acoustic manifestation, namely, closure duration (CD), an enhancing gesture (spread glottis) (Avery and Idsardi, 2001) is added to increase the saliency of contrasts between voiced aspirated stops (VAS) and voiced stops (VS). The acoustic consequence of this enhancement is f_0 lowering in the following vowel. Similar results are obtained under weak prosodic conditions where both f_0 lowering and increase in spectral tilt result from the enhancing gesture (spread glottis). In addition, in segmental contexts where gestural overlap compromises the feature (slack vocal folds), (spread glottis) by way of f_0 lowering in the acoustic dimension enhances the laryngeal contrast between VAS and VS. These results lend support to the theory of enhancement as proposed by Stevens and Keyser (2010).

4pSCb9. Statistical analysis of constriction task and articulatory posture variables during speech and pausing intervals using real-time magnetic resonance imaging. Vikram Ramanarayanan, Louis Goldstein, Dani Byrd, and Shrikanth Narayanan (Univ. of Southern California, Electrical Eng., Los Angeles 90089)

We have shown previously using real-time magnetic resonance imaging data (Ramanarayanan *et al*., 2011, ISSP Montreal) that it is more likely that articulatory posture variables (such as jaw angle) are controlled to achieve articulatory settings during pausing intervals in read speech than constriction task variables (such as lip aperture, tongue tip constriction degree, etc.). In this study, we extend this work to examine correlations between constriction task variables and articulatory posture variables during both speech and pause intervals. This work serves to deepen our understanding of the differences in postural motor control of the vocal tract observed during speaking and pausing. [Work supported by NIH.]

4pSCb10. The scope of phrasal lengthening: Articulatory and acoustic evidence. Daylen Riggs and Dani Byrd (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089)

The temporal lengthening that occurs at phrase edges is known as phrasal lengthening. The scope of phrasal lengthening refers to the distance both before and after a phrase edge that phrasal lengthening can occur. This paper examines the influence of prosodic prominence on the scope of phrasal lengthening in articulation. Pitch accent was placed immediately adjacent to the phrase edge and at varying distances before and after the phrase edge. Articulatory durations of gestures were measured in the pitch-accented syllables and in the syllables intervening between the phrase edge and the pitch-accented syllables. Acoustic measurements of consonant, syllable, and vowel duration were also examined. These constriction durations were compared to those for gestures in phonologically parallel control sentences that lacked a phrase boundary. The results indicate that phrasal lengthening is most systematic immediately adjacent to the phrase edge. However, pitch accent can attract phrasal lengthening. One subject showed phrasal lengthening in a pitch-accented syllable, three syllables away from the boundary. Finally, when the remote pitch-accented syllable showed phrasal lengthening, gestures intervening between the phrase edge and the pitch-accented syllable also showed phrasal lengthening. These patterns are evaluated in the context of the prosodic gestural model of Byrd & Saltzman (2003).

4pSCb11. An ultrasound study of Canadian French rhotic vowels. Jeff Mielke (Arts 401, 70 Laurier Ave East, Ottawa, ON K1N6N5, Canada, jmielke@uottawa.ca)

Some speakers of Canadian French produce the vowels /ø/ and /œ/ with a rhotic perceptual quality, leading *pneu*, *docteur*, and *brun* to sound like /pnr/, /dOktarR/, and /bRr/. English /r/ can be produced with a variety of tongue shapes (including bunched and retroflex variants; Delattre and

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Freeman 1968, etc.), raising the question of whether French rhotic vowels are also produced with categorically different tongue shapes. Mid-sagittal ultrasound video was recorded for three native speakers of Canadian French producing words containing /œ/, /œR/, and /ø/ in a carrier phrase. Acoustic analysis of rhotic- and non-rhotic-sounding vowels reveals that the rhotic perceptual quality is associated with a low third formant, which is an important acoustic cue for English /r/. Two of the subjects produced rhotic vowels with a tip-down bunched tongue shape closely resembling Delattre and Freeman's type 4 and one subject produced them with a retroflex tongue shape. The study provides articulatory data on retroflex vowels, which are typologically rare, and on bunched-retroflex variation in a language other than English. The parallel cases of articulatory variability in French and English raise opportunities for investigating articulatory-acoustic mapping in bilinguals.

4pSCb12. Can you say [ṽ] or [x̃]? Aerodynamics of nasalized fricatives in Scots Gaelic. Daniel Brenner, Andrea Davis, Natasha Warner, Andrew Carnie, Muriel Fisher, Jessamyn Schertz, Michael Hammond, and Diana Archangeli (Dept. Ling., Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, dbrenner@u.arizona.edu)

Scots Gaelic is said to have nasalized fricatives ([ṽ, x̃, h̃] etc.). Nasalized fricatives may be aerodynamically impossible (except [h̃]), because opening the velum would vent the high oral pressure required for frication (Ohala 1975). However, phonologically nasalized fricatives which are realized some other way phonetically are possible, and [h̃] is possible because the frication occurs behind the velic opening. The current work presents oral and nasal airflow data from 14 native speakers of Scots Gaelic, primarily from the Skye dialect, to investigate the nasalized fricative distinction. Results indicate that the most common solution to the aerodynamic problem is to neutralize the distinction: most phonological nasalized fricatives, from most speakers, are simply not nasalized at all ([v] for [ṽ]. Some tokens show nasalization during the preceding vowel ([ãv] for [aṽ]). Some tokens in which the expected fricative is pronounced as an approximant (common in Scots Gaelic) show nasalization ([w̃] for [ṽ]). Furthermore, [h̃] occurs. Very rarely, there may be slight nasalization overlapping part of an oral fricative. Thus, the data show that conflicting aerodynamic demands are resolved variably: in this dialect, the distinction is usually neutralized; when it is not, several alternatives that avoid the aerodynamic conflict are produced.

4pSCb13. An articulatory and acoustic investigation of Mandarin apical vowels. SANGIM LEE (Dept. of Linguist., New York Univ., 10 Washington Pl. NY 10003, sangim@nyu.edu)

The present study investigates the articulatory and acoustic properties of apical vowels in Mandarin Chinese using ultrasound and acoustic measures. Instead of the high front vowel, dental and retroflex sibilants are known to be followed by apical vowels made with tongue tip/blade, rather than the tongue body. The ultrasound images of one Beijing speaker confirmed that the tongue tip/blade gesture indeed remains unreleased during the production of these vowels. However, it also revealed that the tongue body for apical vowels is significantly retracted with a relative order of the vowel following dentals being more retracted than the vowel following retroflexes. This gave rise to a consistent acoustic consequence; the second formant of the vowel after dentals of six Mandarin speakers was significantly lower than that of the vowel after retroflexes. In addition, the third formant of the vowel after a retroflex was significantly lowered and nearly merged with the second formant, indicating that the vowel is also strongly retroflexed. The correct analysis is thus claimed to include the apical gestures as secondary features added to the main articulation of the tongue body as in the following notation: a dentalized back vowel and a retroflexed mid vowel.

4pSCb14. Diphthong centralization and reduction in constriction degree. Fang-Ying Hsieh, Louis Goldstein, and Khalil Iskarous (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, fangyinh@usc.edu)

Diphthongs in Mexian Hakka Chinese have been divided into two categories (Cheung 2007): Endpoints of the falling (in vowel space) diphthongs have formant values equivalent to those of the monophthongal vowels, while the endpoints of the rising diphthongs generally have more centralized formants. This categorization provides a chance to test the hypothesis of Iskarous *et al*. (2010) that in speech production the location of constriction (CL) changes discretely in transitions between successive targets, while the degree of constriction (CD) changes continuously. We therefore hypothesize that the centralization in these diphthongs results from a reduction in CD (possibly due to undershoot) rather than CL, thus making it parallel to common consonant reduction changes in languages, such as spirantization of stops. The current study tests this hypothesis using TADA (TAsk-Dynamic) modeling, in which CD and CL in speech production can be manipulated. Diphthongs [ia], [ua], [ai], and [au] were modeled, and the preliminary results indicate that a change in target CD alone yields formant patterns that are more similar to those reported compared with a change in CL. Results of additional studies will be presented that analytically derive CD and CL in these diphthongs from new data collected from speakers of Hakka.

4pSCb15. When inferring lingual gestures from acoustic data goes wrong: The case of high vowels in Canadian French. Will Dalton (Dept. of Linguist., Univ. of Ottawa, 70 Laurier East, Ottawa, ON K1N6N5, wdalt017@uottawa.ca)

Canadian French (CF) is distinguished from other dialects partly by the presence of lax high vowel allophones in closed syllables (Walker, 1984). Acoustically, tense high vowels are characterized by a lower F1 than their lax counterparts, which could be the result of tongue root advancement, tongue body raising, or both (Ladefoged & Maddieson, 1996). High vowel allophony in CF therefore represents a case in which articulatory gestures cannot reliably be inferred from acoustic data alone. Nevertheless, the literature discussing the phonetic properties of high vowels in CF commonly assumes tongue root position to be the parameter that distinguishes between tense and lax vowels, despite an absence of empirical evidence. The purpose of this experiment is to test this assumption and provide articulatory evidence using ultrasound imaging to examine tongue position during speech production by CF speakers. Results indicate that an advanced tongue root gesture is not used to distinguish between high vowels in CF; no significant difference in tongue root position was found between tense and lax allophones. Rather, tongue body height was found to be the distinguishing feature. These findings contribute to our knowledge of the typology of articulatory gestures used to distinguish between so-called tense and lax vowels.

4pSCb16. Identifying relevant analysis parameters for the classification of vocal fold dynamics. Daniel Voigt (Dept. of Linguist., Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, D-04103 Leipzig, Germany, daniel_voigt@eva.mpg.de) and Ulrich Eysholdt (Univ. Hospital Erlangen, Bohlenplatz 21, D-91054 Erlangen, Germany)

In the previous work, a computer-based analysis framework was proposed, which is capable of objectively and automatically classifying vocal fold vibrations as captured by high-speed videoendoscopy during phonation. The method is based on quantitative feature extraction from Phonovibrograms combined with nonlinear machine learning techniques, allowing for the discrimination of normal and pathological laryngeal movement patterns. The diagnostic reliability and potential of this analysis approach were demonstrated. However, the practically relevant question, whether certain control parameters of the procedure can lead to increased classification accuracy, remained partially unanswered. In this study, the following parameter sets of the analysis framework were investigated in a systematic manner: method of feature extraction, type of feature aggregation and normalization, number of considered oscillation cycles, feature laterality, classification task, and employed machine learning algorithm. For this purpose, more than 150 000 experiments were conducted using a data set of 105 laryngeal high-speed video recordings, comprising various clinical cases with non-organic findings and subjects from a healthy control group. The results of this extensive study show the particular suitability of certain parameter combinations, helping to further improve the practical application of the automated classification framework for vocal fold dynamics.

4pSCb17. A two-dimensional/three-dimensional hybrid structural model of the vocal folds. Douglas Cook, Pradeep George, and Margaret Julias (Div. of Eng., New York Univ. Abu Dhabi, Abu Dhabi, United Arab Emirates)

The spatial dimensionality of the vocal fold vibration is a common challenge in creating parsimonious models of vocal fold vibration. The ideal model is one that provides acceptable accurate with the lowest possible

computational expense. Inclusion of full three-dimensional (3D) flow and structural vibration typically requires massive amounts of computation, whereas reduction of either the flow or the structure to two dimensions eliminates certain aspects of physical reality, thus making the resulting models less accurate. Previous two-dimensional (2D) models of the vocal fold structure have utilized a plane strain formulation, which is shown to be an erroneous modeling approach since it ignores influential stress components. We herein present a 2D/3D hybrid vocal fold model that preserves three-dimensional effects of length and longitudinal shear stresses, while taking advantage of a two-dimensional computational domain. The resulting model exhibits static and dynamic responses comparable to a 3D model and retains the computational advantage of a two-dimensional model.

4pSCb18. Nonlinear viscoelastic properties of human vocal fold tissues under large-amplitude oscillatory shear. Elhum Naseri, Mindy Du, and Roger W. Chan (Otolaryngol.—Head and Neck Surgery, Biomedical Eng., Univ. of Texas Southwestern Medical Ctr., Dallas, TX 75390-9035)

Traditionally, viscoelastic shear properties of human vocal fold tissues have been described by the linear viscoelastic moduli G' and G" at small strain amplitudes. However, once the mechanical behavior becomes nonlinear these moduli are no longer sufficient for viscoelastic characterization. MITlaos, a rheological framework developed for better describing such behavior, was used to characterize the nonlinear viscoelastic properties of the vocal fold lamina propria when subjected to large-amplitude oscillatory shear (LAOS). MITlaos involved Fourier transform processing to convert raw stress–strain signals into a filtered/smoothed total stress signal based on the odd-integer harmonic components, with nonlinear viscoelastic parameters simultaneously derived. Rheometric testing of vocal fold cover specimens was performed with increasing strain amplitudes using a controlled-strain, simple-shear rheometer. Smoothed Lissajous–Bowditch curves generated from MITlaos were plotted in Pipkin space, and the nonlinear analysis was summarized by variations of rheological fingerprints. Pipkin diagrams served as a geometric means for identifying the onset of nonlinear behavior, and any distortions due to noise. Results showed that the human vocal fold cover when subjected to LAOS demonstrated intracycle strain stiffening and intercycle strain softening with increasing strain amplitude, as well as shear thinning with increasing strain-rate amplitude. [Work supported by NIH.]

4pSCb19. Recovery of articulatory information from acoustics using linear mixed effects models. Sean Martin (Dept. of Linguist., New York Univ., 10 Washington Pl. New York, NY 10003)

The current study attempts to establish a lower bound on the degree to which articulatory information can be recovered from the speech signal. The recovery of articulatory information is modeled with comparatively simple linear mixed-effects models using synchronized articulatory and acoustic data from the University of Wisconsin X-Ray Microbeam Speech Production Database (Westbury, 1994). The performance of this model is compared with other speech inversion models. While previous work [Mitra *et al*. (2010) and others] demonstrates that a more sophisticated model can perform speech inversion quite accurately, the results here show that even a strongly constrained linear model performs better than might be expected given that the well-known non-linearity of the acoustic-articulatory mapping. More detailed investigation of the results indicates that while non-linearity in the acoustic-articulatory mapping affects the recovery of articulatory information for all tongue regions, it contributes considerably more to error in recovering tongue-tip position. For the tongue body, the correlation between predicted and observed tongue body position is approximately 0.9 while the correlation for tongue tip position was approximately 0.65. A linear approximation seems to be sufficient to recover tongue body position with average error of approximately 12%, even with limited acoustic information.

4pSCb20. Emotion effects on speech articulation: Local or global? Sungbok Lee, Jangwon Kim, and Dany Byrd (Dept. of EE, Univ. of Southern California, Los Angeles, CA 90089, sungbokl@usc.edu)

Although differences in overall articulatory behaviors in emotional speech production have been well acknowledged in the literature, it is not well known whether such articulatory differences are distributed or localized along the time axis. In this study, we examine the lower lip and tongue tip trajectories of a perceptually evaluated EMA (electromagnetic articulography) emotional speech database of one male and two female subjects. The speech material is a set of sentences repeated at least three times, and the emotion types investigated are neutral, angry, and happy. Timing differences between repetitions as well as across emotions are normalized by the FDA (functional data analysis, Ramsay and Silverman, 1997) time- alignment technique so that important kinematic land-marks (e.g., velocity maxima) are aligned with respect to each other. Therefore, not only differences in timing but also differences in movement range and velocity can be investigated along the time line. The normalized quantities are sampled along the time line every 10 ms, with a 20-ms window, and their differences along the time line are compared as a function of emotion. Final results as well as their relation to pitch patterns and other acoustic measures will be reported. [Work supported by NIH and NSF.]

4pSCb21. Suprasegmental features in Northern Paiute. Joseph D. Brooks (josephdbrooks@umail.ucsb.edu)

The relationship between intonation and word-level stress in Northern Paiute (Western Numic; Uto-Aztecan) is described in order to support the level of phonology as a basis for the lexical, syntactic, and ideational integration of connected speech into intonation units (IUs). Word-level stress in Northern Paiute, intrinsic and in most cases peninitially fixed, is manifested by vowel lengthening and a rise in pitch. Intonation depends largely on the word-level pitch rises, minimalizing the frequency range for non-tonic accents and increasing it for most rising tonic accents of most IUs. The opposite pattern also occurs, whereby non-stressed syllables of individual words have an intonation pattern superimposed upon them; most tonic accents with falling pitch and a few with rising pitch over a non-stressed syllable distinguish this second category as independent from word-level stress. Although these two patterns can explain how suprasegmental features integrate words into larger, cohesive units, the deictic word class is an exception. Deictics in Northern Paiute lack intrinsic pitch, have extra-long vowels, and occur overwhelmingly in IU-final position. It is proposed that suprasegmental features serve as a cohesive force between the areas of linguistic structure by varying degrees of phonological integration.

4pSCb22. The prosodic structure of Koasati. Matthew Gordon (Dept. of Linguist., UC Santa Barbara, Santa Barbara, CA 93106, mgordon@linguistics.ucsb.edu), Linda Langley (McNeese State Univ., Lake Charles, LA 70609), and Jack Martin (College of William & Mary, Williamsburg, VA 23187)

This paper presents results of the first systematic acoustic investigation of the prosodic system of Koasati, an endangered Muskogean language of Louisiana. Koasati is prosodically complex, featuring lexically marked pitch accents in nouns and verbs, as well as phonologically predictable pitch accents and boundary tones assigned by the intonational system. While most nouns are characteristically associated with a final F0 rise attributed to a phrase-level tone, some have an additional lexically marked high pitch accent on a non-final syllable. The pitch accent system in verbs is more intricate and consists of different types of aspectually determined accents (termed 'grades' in the Muskogeanist literature) and concomitant segmental changes including lengthening, nasalization, or aspiration. Koasati also has a rich intonation system featuring at least two levels of prosodic constituents, the accentual phrase and the intonational phrase, which are associated with boundary tones and/or pitch accents that often temporally overlap with lexically specified pitch accents. Our results will be compared with those for Creek (Johnson and Martin 2001, Martin and Johnson 2002) and Chickasaw (Gordon 1999, 2003, 2004, 2005, 2007, Gordon *et al*. 2000, Gordon and Munro 2007), Koasati's prosodically best described relatives. [Work supported by NSF.]

4pSCb23. Effects of morpheme boundaries in /n/ palatalization in Korean. Jae-Hyun Sung (Dept. of Linguist., Univ. of Arizona, 1100 E Univ. Blvd, Tucson, AZ 85721, jhsung@u.arizona.edu)

Korean palatalization is known to be one of the rare palatalization phenomena where the morpheme boundary plays a role. For example, /mat+i/ 'the elderly', where a morpheme boundary exists between /t/ and /i/, becomes [madi] by the palatalization process, while a tautomorphemic word /mati/ 'joint' is realized as [madi] (*[madi]), with no palatalization. The present study investigates whether the effect of morpheme boundaries is also at play in allophonic /n/ palatalization in Korean as well as in /t/ palatalization using ultrasonic analysis, and shows that morpheme boundaries contribute to allophonic variation. This study uses ultrasound imaging and audio recordings of 6 native speakers of Korean to examine the difference in palatalization between two morphologically different words in Korean—tautomorphemic words (e.g., [koni] 'swan') and heteromorphemic words (e.g., [muni] (/mun+i/) 'door + nominative marker'). Comparison of the degree of palatalization before [i] shows there is stronger /n/ palatalization in tautomorphemic /n/ + [i] words than that in heteromorphemic /n/ + [i] words, in contrast to /t/ palatalization. The results from this study are in line with the reported language-universal tendency for greater palatalization in tautomorphemic environments.

4pSCb24. Louder is longer: Amplitude conditioned lengthening in diasporic Siraiki. Aditi Arora (Dept. of Phonet. and Spoken English, EFL Univ., Tarnaka, Hyderabad, Andhra Pradesh, India 500007) and Indranil Dutta (EFL Univ., Tarnaka, Hyderabad, India 500007)

Based on a longitudinal study of three generations of diasporic Siraiki speakers in India, it is shown that first generation (Gen1) implosives develop into lengthened plosives, intervocalically, in the subsequent two generations (Gen2/Gen3). Historically, Prakrit geminates developed into Siraiki and Sindhi implosives. Ohala (1992a) observes that maintaining voicing during intervocalic geminates is facilitated by increased oral cavity volume, achieved by lowered larynx (LL). The consequence of this cavity enlargement is listener misperception of the accompanying cues that lead to the genesis of implosives in the intervocalic position in Sindhi (Ohala, 1993). Closure voicing durations, root mean square (rms) amplitude during closure, and f_0 following release are measured in this study. It is observed that in contexts where the Gen1 speakers produce increased rms amplitude at the release of implosives, the subsequent Gen2/Gen3 produce lengthened plosives. Increased peak-amplitude has been shown to be a weak cue for initial voiceless long consonants in Pattani Malay (Abramson, 1998). Based on the evidence from this study and Pattani, it is argued that the acoustic consequence of the feature LL, mainly increased amplitude, allows a Gen2/Gen3 mapping of the Gen1 implosives onto lengthened plosives. This mapping suggests that in addition to listener misperception, sound changes result from articulatory-acoustic mismatch.

4pSCb25. Phonetic vowel hiatus in Spanish. Erika Varis (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089, varis@usc.edu)

Phonologically, Spanish tolerates sequences of adjacent vowels at a word boundary, but coalescence and deletion have been impressionistically identified at this juncture as phonetic resolution strategies (Alba, 2006). In other languages, phonetic measurements of vowels in hiatus show gradient distinctions between different prosodic environments, supporting a gestural overlap analysis that produces the perception of deletion or coalescence (for example, Igbo, Catalan, and Greek). The current experimental study investigates phonetic manifestations of vowel hiatus in four prosodic environments (IP, PP, Pwd, and clitic) in Spanish to determine whether hiatus resolution is accomplished by vowel deletion or gestural overlap. The study was conducted with nine native speakers of Peninsular Spanish. The duration and formant frequencies of vowel sequences was measured in read sentences that manipulated boundary strength separating the two vowels. Higher-level prosodic boundaries IP and PP produced no significant hiatus effects, but vowels separated by lower-level Pwd and clitic boundaries showed formant influences of one vowel on the other. The durations of the two-vowel sequences were longer than singleton controls across all conditions, indicating no vowel deletion. The results support an analysis of vowel gesture overlap and reject deletion, contrary to auditory impressions but consistent with the findings in other languages.

4pSCb26. Acoustic cues to prominence in children's speech. Irina A. Shport and Melissa A. Redford (Dept. of Linguist., Univ. of Oregon, Eugene OR 97403-1290)

Previous work indicates that phrasal prominence patterns are not yet adult-like in 6-year-old children's speech. Perception of prominence in English is influenced by temporal patterning, amplitude changes, and fundamental frequency variations across the phrase. This study examined the relative weightings of these cues to prominence in adult judgments of 6-year-old children and college-aged adults' speech. Eleven adult judges listened to two—word phrases produced by 25 children and 25 adults in a counting task designed to induce stress shift (*thirtéen banana* versus *thirtéen barbeque*). The judges decided which word was most prominent in the phrase: number, noun, or equal prominence. Although the number word was always judged to be prominent regardless of speakers' ages or context, initial results indicate that agreement between judges varied systematically with different cues in children and adults' speech. More judges agreed that the number word was prominent in children's speech when the first syllable of this word was produced with an especially high F0 and long duration, but only the relative intensity of the initial syllable predicted inter-judge agreement for adult speech. The results have implications for understanding the development of cue integration in the prominence production. [Work supported by NIH/R01HD061458.]

4pSCb27. Prosodic pattern detection based on fuzzy set theory. Nicholas Bacuez (Dept. of French and Italian, Univ. of Texas, 1 Univ. Station B7600 Austin, Texas 78712-0224, nicholasbacuez@gmail.com)

This research utilizes fuzzy logic and fuzzy set theory to experimentally abstract the mental representations of prosodic contours otherwise not accessible to speakers or researchers. I have developed a pattern recognition system to extract approximations of these representations and their range of variation from speech datasets. Following the principles of fuzzy set theory, a prosodic contour is defined as a category and the variation range of its components is expressed in terms of degree of membership to the category, from 0 (excluded) to 1 (included). Subsequently, the system analyzes the prosodic contour of new sentences by assessing through fuzzy matching their degree of membership to previously identified categories: 1 is perfect, 0.5 is borderline, and 0.1 is improbable. In the current study, I have used closed questions obtained experimentally from native French speakers. The prosodic contour of closed questions is non problematic and ensures the controllability of the output. The model output is an array storing the ideal curve and its degrees of variation. Fuzzy matching reveals that most closed questions from the dataset are in medium range (0.5). This materializes the idea that vagueness (Russell, Wittgenstein), the linguistic correlate to fuzzy logic, is a natural property of language.

4pSCb28. Pragmatically determined variation in Greek wh-question intonation. Stella Gryllia (Univ. Potsdam, Inst. fuer Linguistik/Allgemeine Sprachwissenschaft, Komplex Golm, Haus 35, 0.02, Karl-Liebknecht-Str. 24-25, D-14476 Golm, Germany, gryllia@uni-potsdam.de), Mary Baltazani (Univ. of Ioannina, Ioannina 45110, Greece), and Amalia Arvaniti (UC San Diego, La Jolla, CA, 92093-0108)

This paper presents production data testing the analysis of Arvaniti and Baltazani (2005) and Arvaniti and Ladd (2005) according to which the default melody used with Greek wh-questions is L*+H L- !H% (showing a delayed accentual peak on the utterance-initial wh-word, a low stretch, and a final curtailed rise), with !H% sometimes being replaced by L%. Here it was hypothesized that the melodies also differ in pitch accent and are used in different contexts. Four speakers, two male and two female, took part in reading a varied corpus of questions in contexts that lead to the use of a wh-question either in order to seek information or in order to politely register disagreement (a function of wh-questions peculiar to Greek). Our results confirmed that there are two different melodies: L*+H L- !H%, with a delayed accentual peak and a final rise, and L+H* L- L%, with an early peak and no final rise. The former is used for requesting information and the latter when questions function as dissenting statements. In addition to leading to a revision of the existing analysis, these results show that distinctions such as statement versus question are too coarse-grained for the analysis of intonational meaning and function.

4pSCb29. Acoustic correlates of narrow focus in Turkish. Canan Ipek (Dept of Linguist., Univ. of Southern California Grace Ford Salvatori 301 Los Angeles, CA 9008, ipek@usc.edu9-1693)

Acoustic correlates of narrow focus in Turkish focus is known to affect the acoustic properties of the lexical element under focus. Its effect on the pre-focus and post-focus domains has attracted much less attention. This study aims to investigate the acoustic changes as a function of focus in the pre-focus and post-focus domains as well as in the on-focus domain in Turkish and listeners' sensitivity to those changes in retrieving information from

the acoustic signal. For this purpose, a production and a perception experiments have been conducted. For the production study, speakers read sentences in which the location of focus was manipulated via wh-questions preceding those sentences. For the perception study, listeners heard selected sentences recorded for the production study and were asked to judge the prominent word in the sentence. Results showed that focused words had increased duration and intensity, and post-focus words had reduced F0, duration and intensity. Interestingly, pre-focus words were found to have increased F0 and duration. Listeners' identification of focus was positively affected by the presence of acoustic changes especially in the post-focus domain. These findings have implications for speech perception and modelling prosody.

4pSCb30. Modeling imperatives in Spanish. Sergio Robles-Puente (Dept. of Linguist., Univ. of Southern California, 3601 Watt Way GFS 301, Los Angeles, CA 90089-1693, roblespu@usc.edu)

The intonation of imperatives in Spanish has traditionally been considered to not differ systematically from that of declaratives. This study shows that given the appropriate contexts, imperatives can exhibit unique phonetic properties. Nine speakers of Peninsular Spanish produced imperatives in response to instructions that elicited different levels of imperativity, along with control declarative items. Results show that while imperatives may fail to differ from declaratives in some conditions, when the context requires a stronger imperative, speakers use intonational configurations not found in declaratives. These include higher F0 values and changes in the overall pitch contour with higher F0 values toward the end of the sentence, different boundary tones and different F0 peak alignments. A perceptual experiment with 13 speakers confirmed the relevance of these intonational modulations by demonstrating that the strategies that were more commonly used in the production experiment were preferred over others to express imperativity. Results can be modeled within the framework of grammar dynamics (Gafos and Benus, 2006. Cognitive Science, **30**, 837–862). [This research was supported by the University of Southern California Del Amo Foundation.]

4pSCb31. Intonation of Mandarin speakers in their English as a Foreign language. Karen Barto-Sisamout (SLAT Program, 1423 E. Univ. Blvd., P.O. Box 210067, Tucson, AZ 85721, kabarto@email.arizona.edu)

Does the prosody of speakers' first language (L1) influence their prosody in their second language (L2)? The current work investigates this for tone languages (Beijing Mandarin and Taiwanese Mandarin) as L1, and an intonation/stress language, English, as L2. English uses F0 contours in the intonation system, to signal syllabic prominence in a word, word prominence in a phrase, and the difference between questions and statements. In English, there is a pitch peak delay, where the F0 peak occurs after the stressed syllable in two-syllable stress-initial words. Conversely, tone languages use F0 contours lexically, and in the case of Mandarin, the F0 peak is on the stressed syllable. Thus, F0 measurements were taken from three subject groups (Beijing Mandarin L1, Taiwanese Mandarin L1 and Native English) who produced narrow and broad focus statements and questions, to learn if the Mandarin speakers lack pitch peak delay in English like in their L1, or delay some of their pitch peaks beyond the corresponding syllable offset, like English speakers. Further, all groups produced contrastive focus statements and questions, to see to what degree differences in the L1 system impact L2 production, as Beijing and Taiwanese Mandarin differ in these structures. Data analysis is ongoing.

4pSCb32. Prosodic characterization of reading styles using audio book corpora. Michael I Proctor and Athanasios Katsamanis (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA 90089)

Although native English speakers have strong intuitions about the felicity of different reading styles, it is unclear which properties of read speech contribute to these reactions. Although the prosodic structures of read speech and spontaneous speech have been shown to differ (Howell and Kadi-Hanifi, 1991; Blaauw, 1994), it is not clear whether similar prosodic factors contribute to the perception of different reading styles as more felicitous nor even whether such differences can be systematically quantified. A large-scale corpus analysis of read speech was conducted to shed more light on the prosodic characteristics of those reading styles preferred and dispreferred by native speakers of American English. Audio book recordings of classic works of English literature by male and female American readers were rated by native speakers. The two most and least preferred renditions were transcribed at lexical and phonemic levels using SailAlign (Katsamanis, 2011). A variety of metrics were calculated to characterize prosodic properties of each of the readers, including %V, VarCoV, VarCoC, and nPVI (Grabe, & Low 2002; Stojanovic, 2009). The results suggest that although listeners exhibit a preference for syllabic regularity, the perceived felicity of reading styles results from a combination of factors. [Work supported by NIH].

4pSCb33. Acoustic correlates of spanish speech rhythms. Michael J. Harris (Dept. of Spanish and Portuguese, UCSB, Phelps Hall 4206, Santa Barbara, CA 93106-4150, michaelharris@umail.ucsb.edu) and Stefan Th. Gries (UCSB, CA 93106-3100, stgries@linguistics.ucsb.edu)

This paper describes a study of the acoustic correlates of speech rhythms of Hispanic Bilinguals living in California and Mexican Monolinguals living in Mexico City in order to study the effect of bilingualism on language, especially on rhythm classes, and the reliability of acoustic correlates in distinguishing these classes. This study addresses stress-timing versus syllable-timing as described in Pike's pioneering work (1945). Ten monolingual speakers from Mexico and ten bilingual Spanish–English speakers born and raised in California to Mexican parents were recorded speaking spontaneously. Fifty vowel durations per speaker were collected from phrases in these recordings and explored statistically and graphically with R.2.12.2 (R Development Core Team, 2011) in order to determine the reliability of various acoustical correlates of language rhythms in differentiating speech rhythms between varieties of Spanish. Specifically the Pairwise Variability Index, introduced by Low and Grabe (1995), and interval measures, such as the standard deviation and normalized standard deviation of vowel durations, were explored. The effects of word frequencies, as determined by relevant files from the Corpus del Espa'ol (Davies, 2002), were also considered in the analysis of the data. [Work supported by a grant for the University of California Institute for Mexico and the United States (UC Mexus).]

4pSCb34. Effect of linguistic background on convergence of prosodic rhythm. Gayatri Rao (Dept. of Psych., University of Texas, 1 University Station A8000, Austin, TX 78712, raog@mail.utexas.edu), Rajka Smiljanic (Univ. of Texas, 1 University Station B5100, Austin, TX 78712), and Randy Diehl (Univ. of Texas, 1 University Station A8000, Austin, TX 78712)

Speech patterns of the interlocutors become more similar to each other over the course of an interaction. These spontaneous speech adaptations, or phonetic convergence (PC), have been demonstrated for segmental features, such as vowels and voice onset times (VOT) and for suprasegmental features, such as stress. In this study, speaker adaptations to speech rhythm are examined before and after an interactive map task. Using American English and Indian English speakers, convergence was measured using the centroid of the envelope modulation spectrum (EMS + centroid, Rao & Smiljanic, 2011). This spectral measure of rhythm goes beyond considering consonantal and vocalic duration variability, as used in the traditional rhythm measures, and includes information about syllable prominence, stressed and unstressed syllable variation and distribution, and pauses and disfluencies. This research will allow us to examine whether language background has an effect on convergence of global speech properties, such as linguistic rhythm. The results of this study add to our current knowledge of features that are subject to imitation in the speech of dialogue partners.

4pSCb35. A biologically inspired neural network for modeling phrase-final lengthening. Erin C. Rusaw (Dept. of Linguist., Univ. of Illinois, 4080 Foreign Lang. Bldg., 707 S Mathews Ave. Urbana, IL 61801, erusaw2@illinois.edu)

This work proposes a central-pattern-generator-inspired neural network model for the interaction between phrase-final lengthening and stress. Recent work in the area of speech prosody has been concerned with the mechanisms involved in phrase-final lengthening, and specifically how phrase-final lengthening interacts with stress or prosodic prominence. The current study investigates the interaction of stress and lengthening at the end of English phrases. Adult American English speakers were recorded reading aloud sentences in which phrase boundaries had been manipulated so that the target words were either phrase-final or phrase-medial, and the durations of syllables in the target words were compared between the two conditions.

Results so far support previous findings that phrase-final lengthening in English affects stressed syllables near phrase boundaries and phrase-final syllables while leaving unstressed syllables between the two unaffected [Turk and Shattuck-Hufnagel, 2007]. Domain-based models of prosodic lengthening have so far been unable to provide a unified account of this phenomenon. A biologically plausible artificial neural network is shown which provides a model of the mechanism behind this interaction using three oscillators with differing periods which input to three interconnected thresholded integrate-and-fire artificial neurons, the output of which determines the timing of the syllables, stress feet, and phrase.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 3, 1:00 TO 5:00 P.M.

Session 4pUWa

Underwater Acoustics: Volume Scattering From Objects, Bubbles, or Internal Waves

R. Lee Culver, Chair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Contributed Papers

1:00

4pUWa1. Boundary enhanced scattering by solid metallic simple geometric shapes: Experiments and modeling. Jon R. La Follett (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL 32407, jon.lafollett@navy.mil) and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

The presence of a flat reflecting boundary can enable target scattering mechanisms that are not possible for objects in the free field. Experimental results demonstrating a strong boundary related backscattering feature for a solid aluminum cylinder near an air–water interface will be presented. This effect has been modeled previously [J. R. La Follett, Ph.D. thesis, WSU (2010)] by treating the volume of water bounded by the top surface of a solid cylinder and the air–water interface as a waveguide; qualitative agreement between the model and experimental results was demonstrated. In the present work that model is extended to incorporate aspects of the scatterer geometry. Monostatic and bistatic experimental measurements were obtained by suspending a solid aluminum cylinder, solid steel sphere, and a rectangular aluminum bar through the air–water interface of a tank. Model predictions for a solid cylinder are in good agreement with the observed dependence of the feature on the distance from the cylinder to the air–water interface. [Research supported by the NSWC PCD In-house Laboratory Independent Research program.]

1:15

4pUWa2. Bistatic scattering by scaled solid metallic objects: Circular line-scan measurements and modeling. Jon R. La Follett, Patrick C. Malvoso, and Raymond Lim (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL 32407, jon.lafollett@navy.mil)

Multistatic sonar systems can be used to obtain target scattering information that cannot be measured monostatically. This information has potential benefits for detection and classification schemes. Bistatic and monostatic scattering measurements have been performed simultaneously on several scaled targets using the circular line-scan system of the small scale test bed facility at NSWC PCD. Water tank measurements were made for targets in the free-field and resting proud on or buried in simulated scaled sediment composed of spherical glass beads. Targets were placed in the center of the circular scan line. Circular synthetic aperture sonar and frequency-aspect target strength (acoustic color) results will be presented for a solid steel sphere, solid aluminum cone, and solid aluminum cylinders. Results for cylinders and the sphere are compared with T-matrix simulations to facilitate interpretation of features observed. Comparisons between back and forward scattering results demonstrate particularly strong features for each target in the forward scattering direction. Forward scattering by the sphere and cylinders also exhibits responses that arrive earlier in time than sound that travels directly from the source to the receiver. In the case of the sphere, this is attributed to elastic target responses involving leaky Rayleigh waves. [Research supported by ONR.]

1:30

4pUWa3. Scattering from highly extended acoustical objects using multiple precision computation. C. Feuillade (Facultad de Física, Pontifícia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago, Chile)

The extended boundary condition (or "T-matrix") technique of Waterman [J. Acoust. Soc. Am., **45**, 1417-1429 (1969)] has recently been used to study scattering from highly non-spherical axisymmetric air-filled acoustical objects in water. Evaluation of the T-matrix for extended objects is frequently problematical, typically requiring the inversion of very large and highly ill-conditioned matrices. A second issue is computation of the matrix elements themselves, requiring accurate numerical quadrature of the surface integrals around the scattering object. In this work, a computational scheme was implemented using MATLAB®, and a freely downloadable software package "mptoolbox," which enables numerical computations to be performed to arbitrary degrees of precision. Here, it was implemented with 250 bits of precision specified for the mantissa part of the number, leading to computations with about 75 decimal places of accuracy. Using a multiple precision method significantly improved the stability and accuracy of both types of operations and indicates a powerful tool for the successful implementation of many scattering and other acoustical problems demanding high precision computational schemes. [Work supported by ONR.]

1:45

4pUWa4. On Rayleigh and Mie scattering. Jerald W. Caruthers (Dept. of Marine Sci., Univ. of Southern Mississippi, 1020 Balch Blvd., Stennis Space Ctr., MS 39560, jerald.caruthers@usm.edu)

Scattering described today as "Rayleigh scattering" represents something that is far short of what Rayleigh actually contributed to the topic in both optics and acoustics. This limited view seems to lie in a few papers in which he truncates series solutions for practical computations, thus leading to scattering of the form $(ka)^4$ for $ka \ll 1$, where k is the wavenumber and a is the radius of the sphere and for selected limitations on index of refraction. These approximations led optical scientists to equating "Rayleigh scattering" to little more than "the blue sky." In 1908, Gustav Mie developed a theory for plane-wave scattering from a sphere to which the names "Mie theory" and "Mie scattering" have been indelibly attached to many applications in optics. It is virtually unknown, especially in optics, that Rayleigh actually developed the full theory of plane-wave scattering from a sphere in 1878 (primarily Section 334, Vol. **2**, *The Theory of Sound*, Macmillan),

including original contributions in the concurrently developing mathematics of Bessel functions. The motivation of this presentation is to establish a means of treating weak scattering from bubbles based on their contribution as a distribution of spheres by combining Rayleigh and Mie.

2:00

4pUWa5. Sound speed, pulse spreading, and reverberation in muddy bubbly sediments. William M. Carey and Allan D. Pierce (Dept. Mech. Eng., Boston Univ., Boston, MA 02215)

The sound speed characteristic of the high-porosity mud has been found to have sonic speeds lower than expected. Since the presence of bubbles is known to be an important factor in decreasing the sound speed, these low sound speeds are attributed to methane microbubbles that result from biological decay. A theoretical treatment of "muddy sediments," the Card House Theory (Pierce and Carey, POMA (5), 7001, 2009), estimated the slow sound speed and frequency dispersion proportional to mud porosity, $C_{mud} \sim (0.91 - 0.97)C_w$. The presence of microbubbles can lower the sound speed consistent with the Mallock-Wood equation [Carey and Pierce, A 1aUW6, J. Acoust. Soc. Am. 129(4,Pt2) April 2011]. The recent Dodge Pond experiment found low sound speed estimates consistent with bubble volume fractions between 10^{-4} to 10^{-5}. The experiment has also produced estimates of pulse time spreading and reverberation. This paper interprets these results in terms of a three-component mixture with the bubbles distributed in a random Poisson process. Since measurement of the bubble size distribution within the mud is difficult, limits on the distribution may be obtained by the frequency dependent nature of the sound speed, pulse spreading, and reverberation characteristics. [Work sponsored by ONR OA and NSWC PCD.]

2:15

4pUWa6. Acoustic scattering by bubbles in naturally occurring mud sediments. Allan D. Pierce and Willam M. Carey (Dept. Mech. Eng., Boston Univ., Boston, MA 02215, adp@bu.edu)

Naturally occurring sediment mud contains bubbles created by decaying vegetable matter. Work reported by Preston Wilson *et al*. (ca. 2007) has determined via x-ray tomography systems that mud bubbles are not spherical in shape, but resemble oblate spheroids and are "inhomogeneously distributed." These features are explained in terms of the card-house structure of mud with an adaptation of the fracture mechanics ideas of Boudreau *et al*. (ca. 2002). The scattering of sound at low frequencies by such nonspherical bubbles has both monopole and dipole components. The scattered wave associated with the monopole term is proportional to the bubble volume. The dipole term involves an effective entrained mass tensor, which is found by a solution of Laplace's equation. All bubbles, regardless of shape, have a smallest resonance frequency, and the scattered radiation near the resonance frequency is monopole in character. Example solutions for the resonance frequencies and the scattering near resonance are given for oblate spheroidal bubbles, and a suggested interpolation from low frequencies to resonance frequencies is given. A discussion is also given of how one can make use of the range-evolving form of compact-source generated pulses to infer information about the bubbles near the propagation path.

2:30

4pUWa7. Affects of nearby bubbles on underwater array gain. R. Lee Culver and J. Daniel Park (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804)

Combining multiple sensor signals coherently (i.e., beamforming) improves spatial or angular resolution and increases signal to noise ratio (SNR). When the array is steered, signals arriving from the steering direction add in phase, while signals arriving from other directions do not (proper choice of signal frequency assumed). Array gain (AG) is a measure of how much the SNR at the array output is increased relative to array input SNR. The degradation in underwater acoustic array AG by scattering from nearby bubbles was measured at the AB Wood tank located at the Institute of Sound and Vibration Research (ISVR), University of Southampton, in June 2008. AG degradation is separate from the effects of bubbles in water to attenuate acoustic signals. Measured statistics of signal phase at the individual sensors show that as bubble density increases, phase differences between the elements increase and AG is degraded. We present a theory and numerical simulation that attributes the phase shifts to scattering from nearby bubbles and provides a way to predict AG degradation from the bubble density. Work sponsored by ONR Undersea Signal Processing.

2:45

4pUWa8. Transport equations for cross-frequency transport equations for acoustic intensity moments. Dennis B. Creamer (P.O. Box 660537, Arcadia, CA 91066)

The second and fourth moment mode-amplitude statistics for ocean sound propagation through random sound-speed perturbations are investigated using exact transport theory for the cross-frequency cross-mode coherence matrix. These exact equations are derived using the method of successive approximations, originally developed by Klyatskin and Tartarskii. These equations allow the determination of the validity of the usual transport equation (involving the Markov approximation), which is the first order approximation (in a infinite sequence of approximations to the exact equations). The range scales for the approach to the asymptotic behavior of the intensity moments, and the decay of the cross-modal coherence is easily determined at all frequencies.

3:00

4pUWa9. Transport equation approach for second order mode statistics in an ocean with random sound speed perturbations: Coherence. John Colosi and Tarun Chandrayadula (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943)

A transport equation has been derived to describe the range evolution of the single frequency cross mode coherence matrix so that acoustic field coherence functions with temporal as well as depth and transverse separations can be easily computed. The theory assumes 2-D propagation in the depth range plane, small angle weak multiple forward scattering, and the Markov approximation, and it has been previously shown to accurately predict the observable of mean intensity for both deep and shallow water environments. This talk will address the issues of: the accuracy of the approximations, relative contributions from coupling and adiabatic effects, scaling with range and frequency, and the functional form of the coherence with regards to lag.

3:15–3:30 Break

3:30

4pUWa10. Investigating sources of variability of the range and structure of the low frequency shallow convergence zone. Stephen D. Lynch, Gerald L. D'Spain (Marine Physical Lab.—SIO, 291 Rosecrans, San Diego, CA 92106), Kevin D. Heaney (OASIS, Lexington, MA 02421), Arthur B. Baggeroer (MIT, Cambridge, MA 02139), Peter Worcester (SIO, La Jolla, CA, 92093), James Mercer (APL-UW, Seattle, WA, 98105), and James Murray (OASIS, Lexington, MA 02421)

During an experiment in the northern Philippine Sea in 2009, a ship towing Penn State's Five-Octave Research Array (FORA) at approximately 120 m depth drove counter-clockwise in an arc, maintaining constant range at one convergence zone (CZ) from a second ship holding station with an acoustic source deployed at 15 and 60 m. In addition, the FORA was towed at various depths in a star pattern about the station-keeping source ship, thereby sampling the first CZ in range, depth, and azimuth. Throughout the experiment, sound speed profiles were measured using expendable bathythermographs, expendable sound velocimeters, and conductivity/temperature versus depth sensors, and detailed bathymetric data were collected using the multibeam systems aboard these and other ships. By incorporating this extensive environmental information into numerical models, variability observed in these measurements of the range and structure and asymmetry of the distribution of received levels of the first CZ resulting from a shallow source to shallow receiver are attributed to variability in the sound speed of the upper-most water column and bathymetry (especially bathymetric features mid-way between the source and receiver). It is found that the sensitivity of the CZ to bathymetry is dependent on the sounds speed in the upper water column.

3:45

4pUWa11. Spectral effects of nonlinear internal waves on narrowband shallow-water signal propagation. Chad M. Smith and David L. Bradley (Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804)

Water column, bathymetry, and acoustical sediment properties collected during the transverse acoustic variability eXperiment (TAVEX) of 2008 are used to create a 30 km computational environment for propagation models. These models are used to display internal-wave-induced acoustic variability which is characteristic of the shallow-water northern East China Sea environment where the experimental work took place. Analyses of computational models are then compared to narrowband acoustic recordings made during the experiment using a spectral and time-of-arrival analysis approach. The expected acoustic effects and computational impact of specific internal wave characteristics on acoustic arrivals will be discussed along with recorded data comparison. [Work supported by the Office of Naval Research.]

4:00

4pUWa12. Variability of horizontal interference structure of the sound field in the presence of moving nonlinear internal waves. Mohsen Badiey (Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716, badiey@udel.edu), Boris Katsnelson, and Andrey Malikhin (Voronezh State Univ., Voronezh 394006, Russia)

Variations of the low-frequency sound field when a train of nonlinear internal wave (NIW) crosses the acoustic track are considered. Three positions of moving NIW train, during 10:30 to 11:37 GMT on 19 August 2006 are examined. During this time, three periods of sound transmission took place, each lasting about 7.5 min. These periods referred to as starting time (10:30–10:37.5 GMT), mid-time (11:00–11:07.5 GMT), and the end-time (11:30–11:37.5 GMT). Low frequency modulated signals centered at 300 Hz, and bandwidth of 60 Hz with duration of about 2 s were transmitted. During this time, NIW consisting of 6-7 separate solitons was shifted in horizontal position by 2.5 to 2.8 km, moving toward the coast at the velocity of about 0.75 m/s. Fluctuations of the sound field in horizontal plane are described in detail for three periods of the sound pulses radiation: near the forward front, in the middle of the train, and near the back front using the fluctuations of an angle of horizontal refraction. Estimation of this angle using experimental data and those of the corresponding theory show the same value. [Work supported by ONR.]

4:15

4pUWa13. Observations of Philippine Sea sound-speed perturbations, and the contributions from internal waves and tides, and spicy thermohaline structure. John Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943), Brian Dushaw (Univ. of Washington, Seattle Washington, 98105), Lora Van Uffelen (Univ. of Hawaii, Honolulu, HI, 96822), Matt Dzieciuch, Bruce Cornuelle, and Peter Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

In the PhilSea09 pilot study two moorings equipped with temperature (T), conductivity (C), and pressure sensors, along with upper ocean ADCP, monitored ocean variability for a month in the Spring. The measurements reveal an energetic and nonlinear mixed diurnal-semidiurnal internal tide, a diffuse Garrett-Munk (GM) type internal wave field at or above the reference GM energy level, and a strong eddy field. One mooring, which was equipped with pumped sensors for enhanced salinity (S) resolution, was able to accurately quantify T and S variability along isopycnals (spice). The spice contribution to sound-speed fluctuation is observed to be strong near the mixed layer but significantly weaker than the other contributions in the main thermocline. Frequency spectra as well as vertical covariance functions will be presented to quantify the temporal and vertical spatial scales of the observed fluctuations.

4:30

4pUWa14. Refraction of horizontal rays and vertical modes from receding solitary internal wave front in shallow water. Mohsen Badiey (Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716, badiey@udel.edu), Valery Grogorev, and Boris Katsnelson (Voronezh State Univ., Voronezh 394006, Russia)

Previously, it was shown that on the approach of an internal wave to an acoustic source-receiver track in shallow water, interference between a direct and a horizontally refracted acoustic path can occur (J. Acoust. Soc. Am. **129**(4), EL141, 2011). This phenomenon that is dependent on the angular geometry between the solitary internal wave (IW) front and the acoustic track is similar to the well known Lloyd mirror interference phenomenon in optics. In particular, for a specific time during the SW06 experiment, it was shown that the refracted pulse propagating along horizontal ray, corresponding to the fourth vertical mode arrives with temporal delay relative to the direct horizontal ray. In this paper, we show this phenomenon occurring on the back front of the solitary IW. On the receding front, the forth mode of the refracted signal occurs with an increasing delay in the modal arrival time as the IW leaves the acoustic track. This separation between the direct and the refracted path continues while it gets larger until the IW is far enough from the track for refraction not to occur. A theoretical description of this phenomenon in support of the experimental observation is also presented. [Work supported by ONR.]

4:45

4pUWa15. Variations in the active and reactive intensity components of the sound field due to nonlinear internal waves. Robert J. Barton (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, Rhode Island 02841), Georges A. Dossot (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882), and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943)

Shallow water acoustic energy propagation influenced by nonlinear internal waves is investigated by examining complex acoustic intensity vector fields. The acoustic field is modeled using the three-dimensional Cartesian version of the Monterey-Miami parabolic equation (MMPE) algorithm, which relies upon a split-step Fourier approach. The modeled internal wave is approximated using environmental mooring data from the Shallow Water ’06 (SW06) field experiment, interpolated into a representative three-dimensional sound speed profile, and incorporated into the PE model. The soliton wavecrests are oriented such that they are parallel to the direction of forward acoustic propagation and variations along their length (such as curvature) are neglected. Both pressure and particle velocity fields are computed in a self-consistent manner, allowing a full description of the three-dimensional acoustic intensity field which describes the flow of energy in the presence of the solitons. The complex intensity field is separated into its active and reactive (real and imaginary) and spatial components and presented in the form of energy plots. Specific modeled examples showing horizontal refraction, focusing, and defocusing effects on the structure of the acoustic intensity field are illustrated.

Session 4pUWb

Underwater Acoustics: Measurement, Characterization, and Mitigation of Underwater Anthropogenic Noise

John A. Hildebrand, Cochair
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Contributed Papers

1:00

4pUWb1. Statistical analysis of northern Philippine Sea underwater sounds. Brianne Moskovitz, Gerald D'Spain (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., Bldg. 4, San Diego, CA 92106, bmoskovi@ucsd.edu), Peter Worcester, Matt Dzieciuch (Scripps Inst. of Oceanogr., San Diego, CA, 92037), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA 22039), Jim Mercer (Univ. of Washington, Seattle, WA 98105), and Art Baggeroer (MIT, Cambridge, MA 02139)

The deep ocean experiment, PhilSea09, was conducted April–May, 2009, in the central part of the northern Philippine Sea. The deep 1000-m section of the distributed vertical line array (DVLA), which was composed of 30 elements (upper 10 at 90 m spacing starting at 4285 m depth, and 5 m spacing for the deepest 20 elements over the depths 5185–5280 m), recorded primarily four types of sounds not created as part of the experiment: wind noise, ship noise, airgun signals, and earthquake T-phases. The statistical properties of these sounds are examined quantitatively using non parametric statistical tests operating on the single element and beam level narrowband envelope time series. These statistical tests include the Wald-Wolfowitz runs test for mutual independence of the data samples, the Kolmogorov-Smirnov two-sample test for stationarity, and the Lillifors test for Gaussianity. One result is that these sounds, except for wind-dominated noise, fail the test for Gaussianity. Higher-order spectral analysis is performed to quantify the degree of non-Gaussianity. In addition, analytical probability density functions are fit to the histograms of the envelope values. Physics-based models are developed to predict the statistical characteristics of some of these sounds. [Work supported by the Office of Naval Research.]

1:15

4pUWb2. Ambient noise bathymetric domains. Donald Ross (Wesley Palms Rm 325, 2404 Loring St., Box 101 San Diego, CA 92109, donaldnmiross@mac.com), Megan F. McKenna, Sean M. Wiggins, and John A. Hildbrand (Univ. of California San Diego, La Jolla, CA 92093)

For the purposes of describing and understanding ambient sea noise for frequencies below about 300 Hz, most of the world's seas can be classified as belonging to one of three bathymetric domains. These domains are distinguished by their proximity to shipping lanes and by the degree to which they are exposed to noises originating at long distances. The three domains display different short-term characteristics as well as different historical patterns. In this paper, the three domains are described and typical ambient noise characteristics for each are shown, including changes which are attributable to increased ocean commerce.

1:30

4pUWb3. Unintended consequences of recent changes in ship traffic. Megan F. McKenna, Sean M. Wiggins, John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Ritter Hall 200E, 8635 Kennel Way, La Jolla, CA 92093-0205, megan.mckenna@gmail.com), and Donald Ross (San Diego, CA 92109, USA)

Underwater ambient noise levels measured off the coast of southern California were correlated with regional changes in commercial shipping trade. Between 2007 and 2010, two events occurred that resulted in a decrease in ship traffic in the Santa Barbara Channel: the economic recession and a coastal air-quality improvement rule. From October 2005 to June 2010, monthly low-frequency ambient noise levels at a site 3 km from a major shipping route were compared to regional traffic levels. Two different metrics of ship traffic showed that on average a 1 dB reduction in low-frequency noise levels resulted from a decrease in traffic by one ship passage per day in a coastal basin.

1:45

4pUWb4. Passive acoustic monitoring of vessel presence at Rose Atoll and the National Park of American Samoa. Pollyanna Fisher-Pool (Joint Inst. for Marine and Atmospheric Res., Univ. of Hawaii, Honolulu, Hawaii, pollyanna.fisher-pool@noaa.gov), Marc O. Lammers, Lisa M. Munger (Univ. of Hawaii, Kaneohe, Hawaii), Kevin Wong, and Russell E. Brainard (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii)

American Samoa is in the process of evaluating the development of a network of marine protected areas (MPAs) to preserve coral reef environments and to prevent the decline of fish populations. Two long-standing MPAs in American Samoa are Rose Atoll Marine National Monument (RAMNM) and coastal marine regions within the National Park of American Samoa (NPSA) system. NPSA includes areas on the populated island of Tutuila, while RAMNM is approximately 130 miles away from the nearest population. Both are protected marine reserves where commercial and public recreational fishing are restricted, although the size and remoteness of the locations create a challenging task for observation and enforcement. A growing management concern over the decline of large fishes and possible illegal fishing has prompted interest in vessel incidence within the two MPAs. We gathered evidence of vessel presence with the use of long-term, autonomous passive acoustic monitoring within the two MPAs. Here we present results of vessel detection within acoustic recordings collected 2009–2010 at RAMNM and 2006–2007 and 2008–2009 at the NPSA Tutuila location. Results from this study highlight the patterns and seasonality of vessel incidence and provide managers with information to assist enforcement.

2:00

4pUWb5. A high-fidelity model for mitigating underwater pile driving noise in a shallow ocean waveguide. Kevin L. Cockrell, Ann Stokes, and Dwight Davis (Appl. Physical Sci. Corp., 2445 Truxtun Rd. #200, San Diego, CA 92106)

Driving large piles into the seafloor, as is done when constructing off-shore wind farms, produces high level underwater noise that can have an adverse effect on local marine life. This talk reviews an investigation of methods to mitigate this pile driving noise. A numerical model was used to simulate the structural vibrations in the pile and its coupling to the acoustic field in surrounding air, water and sediment. The simulated acoustic field in the immediate vicinity of the pile was then coupled into an ocean waveguide propagation model using a virtual-source technique to match the boundary conditions. These numerical models were used to assess the relative contribution of the air-borne, water-borne, and sediment-borne acoustic radiation to the noise level in the water-column at ranges up to several hundred meters from the pile. Various noise mitigation methods were simulated and compared. It was determined that a dewatered cofferdam, which places a layer of air between the vibrating pile and the seawater, has the potential to reduce the far-field noise level by approximately 20 dB. A practical method for creating a dewatered cofferdam during construction is the subject of ongoing work.

2:15

4pUWb6. A model for noise radiated by submerged piles and towers in littoral environments. Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, Preston S. Wilson, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Pile driving in shallow water during the construction of bridges and other structures can produce transient broadband noise of sufficient intensity to kill fish and disturb marine mammals. Sustained tonal noise radiated by towers supporting offshore wind turbines contains energy in frequency bands that may inhibit detection of coastal activities via passive sonar and seismic sensors. Understanding the generation and propagation of underwater noise due to pile driving and wind farms is important for determining the best strategies for mitigating the environmental impact of these noise sources. An analytic model, based on a Green's function approach, is presented for the sound radiated in the water column by a submerged cylindrical structure embedded in horizontally stratified layers of sediment. The sediment layers are modeled as viscoelastic media and the Green's function is derived via angular spectrum decomposition. Noise radiation due to both vibration of the structure and impulses delivered to the sediment is considered. Contributions to the pressure field in the water column due to radiation directly into the water, radiation from the sediment into the water, and Scholte waves propagating along the sediment-water interface will be discussed. [Work supported by the ARL:UT IR&D program.]

2:30

4pUWb7. Numerical analysis of sound radiation underwater from a fully submerged pile. Shima Shahab, Katherine F. Woolfe, and Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The study of acoustic radiation from pile driving is essential because the impact between the hammer and pile causes extremely high underwater sound pressure levels that are potentially harmful to the marine environment. When the hammer strikes a cylindrical steel pile, movement of the pile wall in circumferential, longitudinal, and radial directions can excite many modes of vibration. The radial expansions of the pile that propagate along the pile after impact create sound waves in the surrounding water. A finite-difference time-domain (FDTD) formulation is presented to analyze the sound produced from a fully submerged pile with simply supported ends. The pile considered is a cylindrical shell of finite length, bounded by sediment at the bottom end, and surrounded by and filled with water. Three coupled partial differential equations govern the vibration of the shell, and the effects of acoustic media including water and sediment are added in the radial direction. Results of a correlation between the radiated sound field predicted by the FDTD model and acoustic data from a scaled physical model of a fully submerged pile hit with an impact hammer are presented. [Work supported by the Oregon Department of Transportation and Georgia Institute of Technology.]

2:45

4pUWb8. A scaled physical model to study underwater noise from impact pile driving. Katherine F. Woolfe, Shima Shahab, Juan Morales, and Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Development of computational models to predict underwater noise from impact pile driving is limited because of the difficulty and cost involved in collecting acoustic field data for model verification during construction activities. To alleviate this situation, a scaled physical model for marine pile driving was designed and implemented in a 500-gallon shallow water tank, 3.5 m long and 0.85 m wide. The scaled piles are steel pipes having lengths up to 1 m, and radius-to-wall thickness and length-to-radius ratios similar to large cast-in-shell steel (CISS) piles. The wavelength-to-depth ratio for the primary breathing mode of the fully submerged scaled piles and a fully submerged CISS pile of diameter 2.4 m and length 30 m is between 2.0 and 2.5. The impact force is generated and measured with an impulse hammer, and sound field data are collected using a small 2-D hydrophone array. Data are correlated with the results of numerical and analytical models developed to predict sound radiation from CISS piles to verify that the scaled physical model accurately represents their structural acoustics behavior. [Work supported by the Georgia Institute of Technology and the Oregon Department of Transportation.]

3:00

4pUWb9. Underwater structure-borne noise analysis using a finite element/boundary element coupled approach. Dooho Lee (Dept. of Mech. Eng., Dongeui Univ., Busan, 614-714, South Korea, dooho@deu.ac.kr), Hyun-Sil Kim, Bong-Ki Kim, and Seong-Hyun Lee (Acoust. Team, KIMM, Daejeon, South Korea)

Radiated noise analysis from ship structure is a challenging topic due to difficulties in accurate calculation of fluid-structure interaction as well as massive degree of freedoms of the problem even in the case of finite element/boundary element coupled formulations. To reduce the severity of the problem, a new added mass and damping approach is proposed in this paper. The complex frequency-dependent added mass and damping matrices are calculated by using the high-order Burton-Miller boundary element formulation in order to obtain accurate values over all frequency bands. The calculated fluid-structure interaction effects are added to the structural matrix calculated by commercial finite element software, MSC/NASTRAN. An iterative solver is introduced to solve the eigenvalue problem because the combined system matrices are complex and frequency-dependent. The calculated eigenfrequencies for a submerged cylindrical shell show good agreement on reference data. The accuracy and efficiency of the present formulation is compared with those from conventional finite element/boundary element formulation. The comparison results show that the present formulation has better accuracy than the conventional one because of accurate added mass and damping calculation.

3:15

4pUWb10. Non-cavitating propeller noise source inversion. Woojae Seong, Jaehyuk Lee, Ara Hyun (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul 151-742, Korea, wseong@snu.ac.kr), and Junghoon Lee (Samsung Heavy Industry, Taejon, Korea)

Ship's propeller cavitation generates major inboard noise and vibration over the aft body surface of a ship. During the last decade, cavitation-induced hull pressure forces have been reduced considerably leading to reduced noise and vibration levels owing to cavitation, in which case the non-cavitating components of propeller excitations have to be considered: pressure fluctuations due to blade loading and blade thickness. In this work, an algorithm to invert for the non-cavitating propeller noise source parameters based on the experimental data in a cavitation tunnel is shown. The fluctuating hull pressure is estimated by forward modeling based on acoustic boundary element method (BEM). The propeller blade loading and thickness noise sources are modeled using rings of dipoles and quadrupoles, respectively. The inversion

for these pseudo sources are carried out by matching the data obtained from a cavitation tunnel experiment, where several hydrophone are flush mounted on the ship's surface near the propeller. Proper inversion results are obtained when six or more dipole and quadrupole rings are placed at each blade. Using the inverted source parameters, hull pressures were estimated, which show good agreement with the experiment data.

THURSDAY EVENING, 3 NOVEMBER 2011 | 7:30 TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings beginning at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

Animal Bioacoustics	Pacific Salon 1
Noise	Royal Palm 1/2
Speech Communication	Royal Palm 5/6
Underwater Acoustics	Pacific Salon 3

4p THU. PM

Session 5aAB

Animal Bioacoustics: General Topics in Animal Bioacoustics II

Ted W. Cranford, Chair
Dept. of Biology, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182

Contributed Papers

8:15

5aAB1. Hearing threshold measurements using auditory evoked potentials of four stranded short-finned pilot whales (*Globicephala macrorhynchus*) in Key Largo, FL. Danielle R. Greenhow (College of Marine Sci. Univ. of South Florida, 140 7th Ave. S, Saint Petersburg, FL 33701), Micah Brodsky (Dolphins Plus, Key Largo, FL 33037), Robert Lingenfelser (Marine Mammal Conservancy, Key Largo, FL 33037), and David A. Mann (College of Marine Sci., Univ. of South Florida, St. Petersburg, FL 33701)

Approximately 26 short-finned pilot whales, *Globicephala macrorhynchus*, stranded in Cudjoe Key, FL, on May 5, 2011. Four animals, two adult and two juvenile females, were transported to a rehabilitation facility in Tarpon Basin, FL. Auditory evoked potentials (AEPs) were recorded in response to amplitude modulated tone pips modulated at 1000 Hz. AEP thresholds were determined at 10, 20, 40, 80, and 120 kHz for all the four animals. Audiograms were similar to previous findings in pilot whale hearing tests. Short-finned pilot whales have a lower peak sensitivity than other odontocetes such as bottlenose dolphins. Greatest sensitivity was 40 kHz for all whales, while thresholds for the two adults were 25–61 dB higher at 80 kHz than those for the juveniles tested. Click evoked potentials were similar between the four whales and comparable to other echolocating odontocetes. Five total pilot whales have been tested during two separate stranding events; the previously tested juvenile male was found to have profound hearing loss (Mann *et al*., 2010; Schlundt *et al*., 2011). These findings add to the limited database of pilot whale (short- and long-finned) hearing studies, of which there are only two others (Schlundt *et al*., 2011; Pacini *et al*., 2010).

8:30

5aAB2. Monaural and binaural aerial hearing thresholds in a California sea lion (*Zalophus californianus*). Arial M. Brewer (G2 Software Systems Inc., 4250 Pacific Hwy., Ste. 125, San Diego, CA 92110), Jason Mulsow, and James J. Finneran (Space and Naval Warfare Systems Ctr. Pacific, Code 71510, San Diego, CA 92152)

Though studies have focused on some aspects of binaural hearing in otariid pinnipeds, none have directly measured monaural hearing thresholds. Recent auditory evoked potential (AEP) testing with a young male California sea lion suggested that hearing threshold differences potentially existed between the subject's two ears. To further investigate these findings, aerial psychophysical audiograms were collected for this sea lion. Using headphones to deliver stimuli, monaural and binaural thresholds were measured at seven frequencies from 0.5 to 32 kHz in one-octave steps. The binaural and two monaural (left and right ears) psychophysical audiograms were all similar in that they had the typical mammalian U-shape, with peak sensitivity from 8 to 16 kHz. The right ear showed generally elevated psychophysical thresholds when compared to the left, although all differences were less than 7 dB. The psychophysical and AEP audiograms were similar in terms of shape; however, the directions of monaural threshold differences determined with AEP methods were not always consistent with those obtained using psychophysical methods. [Work supported by ONR.]

8:45

5aAB3. Hyperoodon echo-morphology: Biosonar anatomy in the northern bottlenose whale. Ted W. Cranford (Biology Dept., San Diego State Univ., 2674 Russmar Dr., San Diego, CA 92123-3422), Marianne H. Rasmussen (Univ. of Iceland, 640 Hsavik, Iceland), Charles W. Potter (Natl. Museum of Natural History, Washington, DC, 20560), and Petr Krysl (Univ. of California at San Diego, La Jolla, CA, 92093)

The northern bottlenose whale (*Hyperoodon ampullatus*) is one of the largest toothed whales, marked by an outsized bulbous forehead. Like other odontocetes, the head contains a sophisticated biosonar system. The *in situ* anatomic geometry of the biosonar components is normally inaccessible by the large size of these animals. We acquired a 400 kg head from a 6.18 m female Icelandic bottlenose whale. The head was encased in a special sarcophagus. An industrial CT scanner generated more than 700 scans at 2 mm thick, from which we reconstructed the detailed anatomy of the head. The acoustic fats are ensheathed in higher density (acoustically reflective) tissues that act to channel sounds into and out of the head. Lipid channels begin at both sets of phonic lips, coalesce into a single S-shaped channel that eventually passes between two large maxillary crests, and then forms a horn shaped melon that projects sound into the environment. Sounds are apparently received over the jaws and throat and travel back to the bony ear complexes through lipid pathways. These lipid channels bifurcate and insert onto the ear complex where the bone is thin. The finite element simulations for biosonar sound transmission and will be reported.

9:00

5aAB4. Anatomy of a Northeastern Pacific Ocean Sciaenid Fish, the White Weakfish (*Atractoscion nobilis*). Carl R. Schilt (Bigleaf Sci. Services, P.O. Box 225, North Bonneville, WA 98639) and Ted W. Cranford (San Diego State Univ.)

Anatomy of a Northeastern Pacific Ocean Sciaenid Fish, the White Weakfish (*Atractoscion nobilis*) Carl Schilt, Bigleaf Science Services, North Bonneville, WA and Ted Cranford, San Diego State University, San Diego, CA. The bony fish family Sciaenidae, commonly called the croakers, grunts, and drums, are as bioacoustically interesting as their names suggest. Although the anatomy and bioacoustics of some North Atlantic Ocean forms have been investigated, those of the Pacific Ocean are less-well studied. The white weakfish (or white seabass, *Atractoscion nobilis*) is a near-shore marine sciaenid of the West Coast of North America and is the target of sport and commercial fisheries and the subject of a substantial stock enhancement program. Two fresh post-mortem specimens of the white weakfish were subjected to CT scanning on two different spatial scales. The full body of the larger animal, about 1 m long total length (TL), was scanned at a 0.625 mm voxel size whereas the head of a much smaller specimen (about 20 cm TL) was scanned in a "micro-CT scanner with a voxel size of 90 μm. Images and measurements of both specimens, especially bioacoustically relevant anatomy and comparisons with other sciaenid species, will be presented.

9:15

5aAB5. Performance of an echolocating bottlenose dolphin in the presence of anthropogenic masking noise. Eryn M. Wezensky (Sensors and SONAR Dept., Naval Undersea Warfare Ctr., Code 1554, 1176 Howell St., Newport, RI 02841, emwezens@nps.edu), James J. Finneran, Jason Mulsow (US Navy Marine Mammal Program, SSC Pacific, San Diego, CA 92152), Cameron R. Martin (Naval Res. Enterprise Internship Program, Ste. 600, Washington, DC 20036), Brian K. Branstetter, Patrick W. Moore, and Dorian S. Houser (Natl. Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106)

To study the effects of acoustic masking from anthropogenic noise, bottlenose dolphin (*Tursiops truncatus*) echolocation performance was assessed in the presence of different masking noise types using Navy relevant source transmissions. Echolocation clicks produced by the dolphin were detected with a hydrophone, then digitized within a phantom echo generator (PEG). The PEG converted the received clicks into echoes, delayed appropriately for the simulated target range. The echoes were then broadcast to the dolphin via a sound projector while masking noise transmissions were held constant. Using an acoustic response and a modified method of constants procedure, the echolocation performance of the dolphin was computed as a function of range between 3 and 17 m. Comparative echolocation performance to different masking noise type categories was analyzed between intermittent and continuous noise, direct path transmissions and multipath exposure, and mid frequency versus high frequency bands. These results expand the limited understanding of biosonar processing capability and signal characteristic alterations used to discriminate and resolve changes in small scale features while exposed to potential noise interference types. [Work supported through the Office of Naval Research.]

9:30

5aAB6. Dolphins (*Tursiops truncatus*) use multiple auditory mechanisms for detecting tonal signals in natural and anthropogenic noise. Brian K. Branstetter (Natl. Marine Mammal Foundation, 2240 Shelter Island Dr., #204), Jennifer S. Trickey, Kimberly L. Bakhtiari (G2 Software Systems Inc., San Diego, CA 92110), and James J. Finneran (US Navy Marine Mammal Program)

The dolphin's ability to detect a 10 kHz tone masked by a variety of noise types was measured using a standard band-widening paradigm. Maskers included natural noise (rain, snapping shrimp, and ice squeaks), anthropogenic noise (pile saw and boat propeller cavitation) and statistical noise (Gaussian and comodulated noise). For most noise types, detection thresholds increased as noise bandwidth increased up to 1 kHz (the dolphin's critical bandwidth at 10 kHz). Masking patterns for narrow-band maskers were similar regardless of the masker type. However, for noise bandwidths greater than 1 kHz, masking patterns diverged by as much as 23 dB depending on the noise type. The power spectrum model provided reasonable predictions for Gaussian, rain, pile saw, and boat noise masking patterns. Additional experiments suggested that mechanisms related to temporal envelope processing (across-channel envelope comparison and within-valley listening) determined masking patterns for snapping shrimp and comodulated noise. Thresholds in ice squeak noise, which proved to be the most effective masker, were related to the dolphin's inability to discriminate the signal from the background noise rather than inability to detect the signal. These results suggest that the dolphin auditory system uses multiple mechanisms for signal detection in complex noise [Work supported by the ONR.]

9:45

5aAB7. Auditory effects of multiple underwater impulses on bottlenose dolphins (*Tursiops truncatus*). James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, San Diego, CA 92152, james.finneran@navy.mil), Jennifer S. Trickey (G2 Software Systems Inc., San Diego, CA 92110), Brian K. Branstetter (Natl. Marine Mammal Foundation, 2240 Shelter Island Dr., #204 San Diego, CA 92106), Carolyn E. Schlundt (ITT Corp., San Diego, CA 92110), and Keith Jenkins (US Navy Marine Mammal Program, San Diego, CA 92152)

It is becoming increasingly clear that noise may adversely affect the hearing and behavior of many marine mammals, including dolphins and whales. One of the possible consequences of noise exposure is a temporary loss of hearing sensitivity, called a temporary threshold shift (TTS). Previous studies of TTS in marine mammals have used only steady-state and single impulsive sounds; however, many naval, seismic survey, and construction activities generate multiple underwater impulsive sounds. To investigate the auditory effects of multiple underwater impulses, auditory thresholds were measured in bottlenose dolphins before and after exposure to a series of impulses produced by a seismic airgun. Hearing thresholds were compared to determine the amount of TTS as a function of exposure level and the number of impulses. Thresholds were measured by using both psychophysical and auditory evoked potential methods. Preliminary results reveal no measurable TTS after exposure to 10 impulses, with a total cumulative sound exposure level of approximately 195 dB re 1 μPa^2 s. [Work supported by the EP Sound and Marine Life Joint Industry Programme.]

10:00

5aAB8. Subject-based calibration of a jawphone transducer with a bottlenose dolphin (*Tursiops truncatus*). Jason Mulsow (U.S. Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr. Pacific, Code 71510, 53560 Hull St., San Diego, CA 92152), Arial M. Brewer (G2 Software Systems Inc., 4250 Pacific Hwy., Ste. 125, San Diego, CA 92110), and James J. Finneran (U.S. Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA 92152)

Hearing studies with odontocete cetaceans often use suction-cup transducers known as''jawphones" to imitate underwater stimulus transmission. Jawphones are typically calibrated by measuring a frequency-response curve with a receiving hydrophone placed at a controlled distance in an underwater direct field. This procedure is somewhat controversial, as it may not sufficiently reproduce the odontocete sound reception pathway. This study calibrated a jawphone by comparing two behavioral audiograms for a single bottlenose dolphin. The first audiogram comprised underwater hearing thresholds (in dB re 1 μPa) measured using direct-field stimulation, the second comprised thresholds (in dBV) measured using a jawphone. All thresholds were measured using a psychophysical staircase procedure at frequencies from 14.1 to 150 kHz. The calibration curve was calculated by subtracting, at matched frequencies, jawphone thresholds from underwater direct-field thresholds. The resulting curve had the shape of a bandpass filter, with highest levels at frequencies from 56.6 to 130 kHz. This subject-based curve was similar in shape to a previously obtained direct-field calibration curve, although the subject-based values were higher above 20 kHz. These results are especially relevant to auditory evoked potential hearing studies that measure thresholds in odontocetes that are untrained for psychophysical procedures. [Work supported by ONR.]

10:15–10:30 Break

10:30

5aAB9. Migrating gray whale vocalizations and concurrent visual observations near Santa Barbara, California. Lauren K. Roche (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., Mail Code 0205, La Jolla, CA 92093-0205, lroche@ucsd.edu), Michael H. Smith (Gray Whales Count, CA, 93101), Ana Širović (Scripps Inst. of Oceanogr., CA 92093-0205), Allison M. Cope (Gray Whales Count, CA 93101), Jasmine S. Buccowich, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., CA 92093-0205)

The vocalizations of migrating eastern North Pacific gray whales *(Eschrichtius robustus)* are not well understood, but studying these sounds may provide insights to their behaviors and encounters along their migratory route. To record migratory gray whale sounds over long periods, a high frequency acoustic recording package (HARP) was deployed from March to May 2010 off the north coast of the Santa Barbara Channel, an area where the gray whale northbound migration is close to shore. The HARP recorded continuously and concurrently with a shore-based visual survey of marine mammals. Comparisons of calls and sightings were conducted to relate calling repertoires with various types of migrating whale behaviors and groups (e.g., cow/calf pairs, juveniles, etc.). The spatial and temporal overlap of the visual and acoustic data enabled the estimation of call detection ranges and source levels. Characterization of gray whale vocalizations also will help with the development of automatic detectors which will aid in future the investigations of long-term gray whale recordings, including potential responses to anthropogenic activity.

10:45

5aAB10. Characterization of shape diversity in bat biosonar beampatterns. Mohammad Motamedi, Lvyin Cai (Dept. of Mech. Eng., Virginia Tech, IALR, Danville, VA 24540), Qiuping Xu, Washington Mio (Dept. of Mathematics, Florida State Univ., Tallahassee, FL 32306), and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, IALR, Danville, VA 24540)

Numerical beampattern estimates for a large digital shape data set with 361 three-dimensional models of the pinna and noseleaf shapes from 106 different bat species are currently in preparation. Shape samples in this data set have been selected to represent taxonomic diversity and balance noseleaves versus pinnae. The data set will be used to analyze the biodiversity in bat beampatterns. In order to facilitate the extraction of interspecific patterns, spherical harmonics are being evaluated as an analysis tool to describe the spatial frequency content of the beampatterns. Spherical harmonics are the angular portion of a set of solutions to Laplace's equation represented in a system of spherical coordinates. The higher the harmonic, the more solutions are in the respective set. Regardless of the number of solutions, each set forms an orthogonal system. For a compressed characterization of the beampatterns, the representation of the beampattern shape at a given frequency is expressed as a power spectrum; the norm of the vector of the weighting coefficients for all basis functions in the set of a harmonic is taken to represent the portion of the total power explained by a particular harmonic. This power spectrum was found to reflect several beampattern properties well.

11:00

5aAB11. Non-rigid pinna deformations in bats—Behavioral adaptation or dynamic sensing?. Rolf Müller (Dept. of Mech. Eng., Virginia Tech, IALR, 150 Slayton Ave., Danville, VA 24540, rolf.mueller@vt.edu), Li Gao, and Lin Feng (School of Phys., Shandong Univ. 250100 Jinan, China)

Mammal groups with exceptional spatial hearing abilities, bats and primates in particular, are known to have intricate pinna shapes. These shapes determine the monaural spatial hearing sensitivity as a function of direction and frequency, i.e., the reception beampattern. Since even comparatively small, local pinna features can have profound effects on the beampattern, it is possible that some of the interspecific diversity in these pinnae represents spatial hearing adaptations on an evolutionary time scale. In addition, some bat groups, such as horseshoe bats, have elaborate muscular actuation mechanisms that can produce non-rigid pinna deformations. These mechanisms could enable beampattern control on a much shorter, behavioral time scale. This could add considerably to the capabilities of the biosonar systems in these species, because by switching between different pinna shapes—and hence reception beampatterns—could meet even conflicting demands of different biosonar tasks. However, since pinna deformations in horseshoe bats can happen on comparatively short time scales (i.e., one tenth of a second), it is also conceivable that the bats' pinnae are in motion during the reception of an echo. It is not clear at present if and how such a dynamic receiver paradigm could be utilized to enhance biosonar performance.

11:15

5aAB12. Reception beamforming in horseshoe bats is a dynamic process. Lin Feng (School of Phys., Shandong Univ., Shanda South Rd. 27, Jinan 250100, China, lfeng@sdu.edu.cn) and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, IALR, Danville, VA 24540)

Horseshoe bats execute conspicuous pinna movements as part of their biosonar behaviors. These movements contain rigid rotations as well as deformation components that can result in considerable changes to the pinna shapes. Since pinna shape determines the biosonar system's sensitivity as a function of direction and frequency, these non-rigid deformations could be functionally relevant. In the work reported here, it has been investigated how the non-rigid deformations of pinna shape relate to the arrival of echoes. For this purpose, multi-camera high-speed video recordings of both pinna of a bat were synchronized with ultrasonic recordings of the emitted pulses and the returning echoes from a microphone placed next to the head of the animals. Landmark dots were painted on the pinna to enhance the determination of position and velocity from stereo pairs of video recordings. It was found that the pinnae were deformed between as well as during echo receptions. Hence, the echoes were received by different static each pinna shapes (e.g., bent or upright) as well as by pinna shapes that underwent non-rigid deformation within the duration of the echo. As a consequence, echo reception by horseshoe bats involves dynamic beampatterns that can change between as well as during echoes.

11:30

5aAB13. Characterization of dynamic baffles in biosonar and biomimetic devices. Sajjad Z. Meymand, Mittu Pannala, and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, IALR, 150 Slayton Ave., Danville, VA 24540, sajjadzm@vt.edu)

The pinna shapes of horseshoe bats are actively actuated to undergo non-rigid deformations on a time scale of one tenth of a second. These cycle times are similar to the durations of the echoes that the animals receive. Hence, it is possible that the bat pinnae are undergoing a change in shape while an individual echo is impinging on them. Such a dynamic reception paradigm based on a continuously deforming baffle are an interesting concept for explaining the biosonar as well as the design of bioinspired devices. Beampatterns are the standard characterization of time-invariant reception devices. In a beampattern, device gain is given as a function of direction and frequency. The Fourier transform that underlies this representation does not represent time variant behavior. Hence, new ways to characterize the time-variant behavior of a deforming baffle have to be found. This is further complicated because a complete description of system behavior requires four independent variables: azimuth, elevation, and either two time variables or time and frequency. Representations that are either time-frequency or entirely time based and use different ways to visualize the overall system behavior have been implemented and tested on the responses of biomimetic pinna shapes.

11:45

5aAB14. Time-variant biomimetic beamforming baffle prototypes augmented with local shape features. Mittu Pannala, Ojili Praveen Kumar Reddy, Sajjad Z. Meymand, and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, IALR, 150 Slayton Ave., Danville, VA 24540, mpannala@vt.edu)

Horseshoe bats are known to exhibit non-rigid pinna deformations as part of their biosonar behaviors. During these deformation cycles, the pinnae of the animals alternate between upright and bent shape configurations that differ in their overall geometry as well as in the shapes and relative positions of local shape features. Artificial baffle prototypes have been designed to emulate these deforming bat pinnae. The prototype shapes were designed by augmenting an obliquely truncated cone with biomimetic local shape features. The prototype was implemented using elastic (synthetic rubber) materials. The deformation of the structure was accomplished through a simple actuation mechanism that was inserted on the back side of the artificial pinna. Features of the horseshoe bat pinna that have been evaluated for this study include a prominent vertical ridge that runs along the entire inner pinna surface, the antitragus along with the lateral incisions that separate it from the back portion of the pinna, and a pinna-edge incision near the tip of the pinna. An experimental analysis of deforming prototypes has demonstrated that such biomimetic local shape features affect the beampattern of the static artificial pinna shape as well as the time-varying properties of the deforming prototype.

Session 5aNSa

Noise and Physical Acoustics: Launch and Aircraft Noise

Victor W. Sparrow, Chair
Graduate Program in Acoustics, Pennsylvania State Univ., 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—8:00

Contributed Papers

8:05

5aNSa1. A semi-analytical ray method to predict the propagation of long-range vertical noise: Application of NORD 2000 to the prediction of aircraft en-route noise. Kieran Poulain and Victor Sparrow (Penn State, Grad. Program in Acoust., 201 Appl. Sci. Bldg., University Park, PA 16802)

In the past, considerable efforts were made into developing models for the prediction of short-range propagation, in the context of airport noise (Integrated Noise Model, FAA) or more generally for short-range horizontal community noise propagation (NORD 2000, Delta Inc.). Recent research efforts have focused on the prediction of aircraft en-route noise for flights above 18 000 feet above ground level (AGL) (5.49 km AGL), in order to estimate the noise impact in U.S. National Parks and other quiet areas. Unlike other community noise issues, long-range vertical propagation requires an altitude-stratified and realistic atmosphere, which directly impacts the geometrical and absorption losses as well as the ground effects. Given the large distances involved, the use of a semi-analytical propagation model based on the ray theory could prove to be useful in reducing the computation time. NORD 2000, based on semi-analytical ray theory, was modified for use with aircraft en-route noise. A comparative study against AERNOM (Advanced En Route NOise Module, based on the numerical ray method) from Penn State is presented. [Work supported by VOLPE National Transportation Systems Center. The findings are the views of the authors and do not necessarily reflect the views of VOLPE, the FAA, NASA, or Transport Canada.]

8:20

5aNSa2. Evaluation of procedural changes for minimizing noise impact near military airfields. Andrew Christian, Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, azc144@psu.edu), Micah Downing, and Kevin Bradley (Blue Ridge Res. and Consulting, Asheville, NC 28801)

Over time, communities surrounding military airfields tend to grow, while the noise generated by subsequent generations of military aircraft tends to increase. This creates an accelerating problem of human exposure to noise. One technique to alleviate this disparity is to modify attributes of flight trajectories in a way that reduces the sound exposure of the population. This involves the changing of flight ground tracks, profiles, and power settings within linear and non-linear boundaries which can include exceptions (such as no-fly-zones.) To demonstrate the effectiveness of these changes, DNL levels are calculated for real world population distributions and existing Navy departure procedures. These values are generated computationally using the widely accepted NoiseMap model. It is shown that significant reductions of exposure averaged over the population can be achieved in this way. Possible methods for automating this procedure are discussed. Properties of the objective function spaces are explored (e.g., convexity) and the appropriateness of different optimization methods is discussed. [Work supported by Naval Air Warfare Center.]

8:35

5aNSa3. Similarity spectra and the effect of propagation on F-22 spectra at afterburner. Tracianne B. Neilsen and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602)

The afterburner spectra of a static F-22 Raptor have been compared to the similarity spectra from the two-source theory for jet noise. In the two-source theory, fine-scale turbulent structure (FSS) account for sideline radiation and large-scale structures (LSS) dominate the radiation for inlet angles aft of 125 deg for model-scale jets [Tam and Zaman, AIAA J. **38**(4), 592–599 (2000)]. A comparison of the similarity spectra with F-22 afterburner data, recorded at ground-based microphones along the 40-ft foul line, shows good agreement between the FSS spectrum and the sideline radiation and the LSS spectrum and the spectrum measured at 150 deg. However, discrepancies exist between the measured spectrum in the direction of maximum radiation and the LSS spectrum. Most significantly, the high frequency roll-off in the data is shallower (approx. 20 dB/decade) than predicted by the LSS (29.8 dB/decade). Most likely this difference comes from nonlinear propagation of the noise, which could then cause the agreement with the similarity spectra to be dependent on distance. Results from the ongoing investigation into how the measured spectra evolve relative to the FSS and LSS spectra as the noise propagates are presented. [Work sponsored by the Air Force Research Laboratory.]

8:50

5aNSa4. A parabolic equation method for predicting sound propagation above and below a layered ground. Hongdan Tao and Kai MIng Li (Dept. of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031)

A numerical model based on a wide-angle parabolic equation (PE) technique has been developed for calculating the sound fields above and below a layered ground. A finite element discretization was applied along the vertical direction instead of the classical finite difference scheme. By using the finite element approach, the boundary conditions, i.e., the continuity of pressure and velocity, can be incorporated directly at the air/ground and ground/ground interfaces. The range-dependent sound fields were obtained by marching the finite-element solutions (above and below the layered ground) in the radial direction. This paper reports a preliminary formulation and presents some initial computational results for the sound fields above and below a porous ground. The numerical results from the PE calculations were compared with the results obtained from analytical solutions for and other benchmark results. A linear and cubic Hermite interpolate function was used in the numerical formulations for the finite element model. It has been shown that use of the cubic Hermite interpolation function generally leads to more accurate numerical solutions with fewer elements. [Work partially funded by Federal Aviation Administration and the China Research Scholarship Council.]

9:05

5aNSa5. Shock associated noise generation in curved turbulent Coanda wall jets. Caroline Lubert (Dept. of Mathematics & Statistics, James Madison Univ., Harrisonburg, VA 22807)

Curved three-dimensional turbulent Coanda wall jets are present in a multitude of natural and engineering applications. The mechanism by which they form a shock-cell structure is poorly understood, as is the accompanying shock-associated noise (SAN) generation. This paper will discuss these phenomena from both a modeling and experimental perspective. The Method of Characteristics will be used to rewrite the governing hyperbolic partial differential equations as ordinary differential equations, which will then be solved numerically using the Euler predictor-corrector method. The effect of complicating factors such as radial expansion and streamline curvature on the shock-cell location predictions will be discussed. Comparison of the theoretical calculations of the shock wave structure with associated schlieren flow visualization results will be presented. Related acoustical measurements will also be communicated. In this way, critical flow characteristics for shock-cell formation will be identified, and their influence on SAN discussed.

FRIDAY MORNING, 4 NOVEMBER 2011 ROYAL PALM 1/2, 9:45 TO 11:50 A.M.

Session 5aNSb

Noise: General Topics in Noise

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

Chair's Introduction—9:45

Contributed Papers

9:50

5aNSb1. Vuvuzelas and their impact. Kenneth A. Cunefare (The Georgia Inst. of Technol., School of Mech. Eng., Atlanta, GA 30332, ken.cunefare@me.gatech.edu), Richard Ruhala, Laura Ruhala (Div. of Eng., Southern Polytechnic St. Univ., Marietta, GA 30060), and Tina Ortkiese (Arpeggio Acoust. Consulting, LLC, Hampton, GA 30228)

Vuvuzelas are inexpensive plastic horns that became a part of the American vocabulary during the 2010 World Cup held in South Africa, though horns of the style have been employed for decades in the States. During the World Cup, spectators played these horns through the games, causing noise complaints from athletes and spectators alike, including television listeners. The vuvuzelas and similar small, plastic horns are sold in the U.S. and other countries, and there is concern that they will create a related noise problem at future sporting events. This work presents the sound pressure levels, spectra, and directivity measured in a hemianechoic chamber produced by several different vuvuzelas, one at a time. Furthermore, a model is developed to predict noise levels within the stands and on the playing field assuming type- and temporal-mixes of the horns.

10:05

5aNSb2. Theoretical and numerical modeling of a parallel-baffle rectangular duct. Davide Borelli, Corrado Schenone, and Ilaria Pittaluga (DIPTEM, Univ. of Genova, Via all'Opera Pia 15/A, I 16145, Genova, Italy)

The paper describes the theoretical and numerical analysis of sound attenuation in a parallel-baffle rectangular ducts. Insertion losses in a frequency range up to 8000 Hz were predicted by means of a FEM numerical model and by means of analytical models from Sabine and Kurze. The models were then validated in the frequency range from 125 to 8000 Hz by comparing theoretical and numerical results with experimental data obtained in accordance to EN ISO 11691 and EN ISO 7235 standards. The results of the comparison indicate that the behavior of such a dissipative/reactive silences, with its internal-reflections and energy dissipation phenomena, can be predicted quite well by the FEM model on the whole frequency range. On the contrary, analytical models show little accuracy and such predictions are not always so accurate as design requires; besides, the complexity of the analytical approach tends to limit its application to the common design practice. Overall comparisons suggest that FEM modeling can be an accurate and inexpensive way to predict sound attenuation in parallel-baffle mufflers and fulfile the ever rising needs of proper methods in acoustic design of AC and ventilation plants.

10:20

5aNSb3. Active control of a centrifugal fan in a mock laptop enclosure. J. James Esplin, John K. Boyle, Scott D. Sommerfeldt, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, jamesesplin@hotmail.com)

Active noise control (ANC) has shown promise in minimizing the effect of fan noise on users. Recent research by the authors has developed a model which is used to implement ANC on the inlets of centrifugal cooling fans. This model is based on minimizing radiated acoustic power in a model of the fan radiation and using those results to determine appropriate nearfield locations for the error sensor(s). Though this approach has been experimentally verified in an idealized setting, it was not verified in a more realistic situation. This paper describes how this model was expanded from its idealized setting to a mock laptop enclosure. When necessary modifications to the model were made, tonal noise can be predicted in the nearfield of the fan inlets, which allows one to develop an effective compact, realistic ANC setup for use in the mock laptop enclosure. With this ANC setup, significant global reduction of the inlet tonal noise can be achieved.

10:35

5aNSb4. A two-dimensional model for control of centrifugal fan inlet noise in a notebook computer. John K. Boyle, J. James Esplin, Scott D. Sommerfeldt, and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, jkb321@gmail.com)

Previous work on active control of exhaust noise from small centrifugal fans demonstrated significant reductions of the blade passage frequency (BPF) tone. A fan and heat-sink were placed within a mock-up notebook computer case, and control of the fan exhaust noise was measured. It was found that control of the BPF in the exhaust did not significantly affect noise radiated from the fan inlets into the notebook casing, suggesting that exhaust noise and inlet noise may be controlled separately without one adversely affecting the other. In the current work, a two dimensional half-space, source coupling model has been developed to calculate the field within the notebook casing caused by the inlet noise. As a first approximation, free-space boundary conditions were used. A two-dimensional space was constructed to test the model, and error sensor placement was predicted. Measurements of radiated sound power show significant reduction of the blade passage frequency tone. Factors influencing experimental agreement with the model are discussed, such as modal effects and primary source location.

10:50

5aNSb5. Noise reduction in pipelines using Helmholtz resonators. Maaz Farooqui, Samir Mekid, Muhammad Hawwa, Atif Aliuddin, and AbduAziz Al-Hamoud (Mech. Eng., King Fahd Univ. of Petroleum and Minerals, Dhahran 31261, Saudi Arabia, maazfarooqui@kfupm.edu.sa)

Compressors are at the heart of most petrochemical and industrial power plants. They are usually the source of noise generation transmitted to pipelines. This noise is undesirable for people living close to installation and can also potentially cause structural failures in the piping. Particular attention is given to the modified solution using the Helmholtz resonator concept to reduce noise. An assessment of the noise reduction results using various types of resonators is carried out. A comparison between analytical, numerical simulation using COMSOL and experimental test is presented and discussed using a known patch-test.

11:05

5aNSb6. Compliant-wall methods for fluid-borne noise reduction. Kenneth A. Marek, Nicholas E. Earnhart, and Kenneth A. Cunefare (Georgia Tech Dept. of Mech. Eng., 771 Ferst Dr. Grad Box 334, Atlanta, GA 30332, ken.marek@gatech.edu)

One means of addressing fluid-borne noise in hydraulic systems is to add a compliant-walled section to the flow path. The impedance mismatch at the section boundary reflects a substantial amount of acoustic energy back to the source, and additional damping may be present due to the compliant material. While hoses can be used to add this compliance, the hose wall vibration produces undesirable breakout noise from the system. Additionally, the requirement that the hoses contain system pressure determines the minimum practical hose stiffness. An alternative approach is to use a silencer composed of a rigid outer housing, an internal compliant liner, and a central flow path through the liner. Such a configuration would avoid the pressure bearing and breakout noise complications while still adding compliance to the system. These two approaches are compared in this study, with simulated transmission loss presented for various lengths and material properties of the compliant sections.

11:20

5aNSb7. Sound quality characteristics for transient noise of high speed railway interior. Buhm Park (Dept. of Mech. Eng., Hanyang Univ., Haengdang-dong, Seongdong-gu, Seoul, Korea, parkbuhm@gmail.com), Sinyeob Lee (Hanyang Univ., Seoul, Korea), Sunghoon Choi (High-Speed Rail Div. Korea Railroad Res. Inst., Uiwang-si, Kyungki-do, Korea), and Junhong Park (Hanyang Univ., Seoul, Korea)

This study presents interior noise characteristics of Korean high-speed train under different operation conditions. The interior noise was measured for various operating conditions such as different speeds, concrete and ballast track, and open and tunneled track, passing by another moving train. The transient variation of the sound pressure was recorded. The objective and subjective sound quality evaluation of the interior noise was performed. For quasi-static sounds measured for open lands and in tunnel, Zwicker's loudness has dominant impact on the annoyance. When there are transient variations in the sound pressure encountered during entrance, exit, and passing by, the sharpness and roughness also have significant impact on the perceived annoyance. The obtained data are useful for the design of interior of the train since the transient variation of the sound pressure level is influenced also by the sound environment in the room.

11:35

5aNSb8. Measurement and analysis of squeal noise of vehicle wiper system. Dongki Min (Dept. of Mech. Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul 133-791, Korea, dkmin@hanyang.ac.kr), Seongbin Jeong (Hanyang Univ., Seoul 133-791, Korea), Heewon Kang (Res. & Development Div. Hyundai Motor Co.), Honghee Yoo (Hanyang Univ., Seoul 133-791, Korea), and Junhong Park (Hanyang Univ., Seoul 133-791, Korea)

The squeal noise occurring from rubber and hard surface is an important issue, especially for vehicle wiper system. The noise is generated at relatively high frequency ranges, and its generation mechanism during the wiper operation is not straight forward to understand. In this study, experimental setup to generate the squeal noise in a consistent manner is designed using a rotating glass table. The wiper is located on the glass table in compressed status so that to simulate the wiper in the actual operating system. While the glass was rotating with constant velocity, the windshield washer fluid was sprayed on the glass. Under the normal constant velocity of the wiper operation, the condition (temperature, moisture contents, velocity) of continuous occurrence of the squeal noise was identified. The friction coefficient was measured, and its impact on the squeal noise generation was identified. The influence of various parameters such as the compression ratio, temperature, types of the wiper, angle of wiper installation, kinds of washer liquids was investigated using the suggested experimental setup. This provide the required information to design the vehicle wiper system that does not induce the squeal noise during the normal operation.

Session 5aSCa

Speech Communication: Speech Rhythm in Production, Perception and Acquisition II

Amalia Arvaniti, Chair
Dept. of Linguistics, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108

Chair's Introduction—8:00

Invited Papers

8:05

5aSCa1. Is infants' native language preference in the visual modality guided by speech rhythm?. Nancy Ward and Megha Sundara (Dept. of Linguist., UCLA, 3125 Campbell Hall, Box 951543, Los Angeles, CA 90069-1543, nancyward@ucla.edu)

Newborns prefer to listen to their native language. We have shown that this preference for the native language also exists with visual cues alone (Ward and Sundara, 2010). Four and 8-month-old monolingual English-learning infants looked longer at a face speaking English compared to Spanish. What information in the talking faces facilitates language identification, and thus language preference in infants? Studies on visual language discrimination (Soto-Faraco *et al*., 2007; Ronquest *et al*., 2010) suggest three possibilities: (1) lexical information accessed through lip-reading, (2) language-specific movement of the articulators, and (3) rhythmic information. Infants in their first year of life cannot rely on their developing lexicon for visual language preference. Here, we investigate whether rhythmic information is guiding infants' attentional preferences. We use the preferential looking procedure to test whether monolingual English and monolingual Spanish-learning 4-month-olds show a preference for looking at a face speaking Dutch over Spanish. Monolingual Spanish-learning 4-month-olds are expected to show a native language preference for the face talking in Spanish, replicating our previous findings. Crucially, if familiarity with language rhythm is sufficient to guide preferences for talking faces, we expect monolingual English-learning infants to demonstrate a preference for rhythmically similar Dutch, when presented opposite a rhythmically different Spanish.

8:25

5aSCa2. A cross-linguistic investigation of word segmentation by monolingual infants learning Spanish and English. Megha Sundara and Victoria Mateu (UCLA, Dept. of Linguist., 3125 Campbell Hall, Los Angeles, CA 90095)

Infants' ability to find words in fluent speech is affected by their language experience. The focus of this study was the extent to which monolingual Spanish- and monolingual English-learning 8-month-old infants' segmentation abilities are tied to their familiarity with the rhythmic properties of their native language. Spanish and English are considered to belong to different rhythm classes; the former is syllable-timed, whereas the latter is stress-timed. However, in both languages CVC.CV words are predominantly trochaic—i.e., words with stress on the initial syllable (Cutler Carter, 1984; Pons Bosch, 2010). We tested segmentation of trochaic words of the form CVC.CV in Spanish and English. Spanish word segmentation was tested using the Headturn Preference Procedure and English word segmentation using the visual fixation procedure. According to Jusczyk *et al*.'s rhythm hypothesis, infants should fail to segment words in a rhythmically different language. Contrary to the rhythm hypothesis, preliminary data indicate that Spanish-learning infants are successful in segmenting trochees in both Spanish and English. Thus, successful transfer of word segmentation abilities by infants seems to depend upon a match in the prosodic shape of the target word across the two languages.

8:45

5aSCa3. The learnability of language-specific fundamental frequency-based rhythmic patterns. Kristine M. Yu (Dept. of Linguist., Univ. of Mass. Amherst, 226 South College Bldg., Amherst, MA 01003, krisyu@linguist.umass.edu)

In addition to durational patterns, melodic patterns from variations in f0 contribute to percepts of speech rhythm, too (June 2005, Niebuhr 2009). The overall regularity of these melodic patterns is language-specific. In lexical tonal languages like Mandarin, f0 patterns may be quite variable, as many or all syllables may be associated with a tone; in stress-accent languages like English, f0 variation may be less variable as it is driven by stress and prosodic boundaries (Eady, 1982). In accentual phrase languages like Bengali and French, f0 variation is typically quite regular as it is driven almost entirely by prosodic boundaries in a very constrained way. With only acoustic parameters from the input available, are melodic patterns from these different kinds of languages learnable? And how is learnability affected if speech is directed to young infants? To study this, we recorded parents reading language samples in laboratory speech and in simulated infant directed speech in the languages mentioned above, and we are analyzing the speech with durational and f0 variability measures and prosodic (ToBI) labels. We will present results on learnability of melodic patterns using computational methods for pattern classification from a selection of the languages.

9:05

5aSCa4. The acquisition of English rhythm as a function of changes in phrase-level prosody. Melissa A. Redford, Hema Sirsa, and Irina A. Shport (Dept. of Linguist., Univ. of Oregon, Eugene, OR 97403-1290, redford@uoregon.edu)

Language rhythm emerges from a combination of lexical and phrase-level prosody. This talk will review evidence from on-going and completed studies on the acquisition of English prosody in typically developing school-age children to show that the surprisingly protracted acquisition of English rhythm is due to the extended acquisition of phrase-level prosody rather than to immature lexical stress production. Cross-sectional and longitudinal data from 20 native American-English speaking children indicate that whereas adult-like rhythm may not be fully acquired until age 8, even the youngest children have mastered lexical stress. Unlike eight-year-olds, though, younger children show less reduced vowels in function words and a less robust pattern of phrase-final lengthening. We argue, based on evidence from a separate study on stress-shifting in 25 six-year-olds and 25 adults, that younger children have yet to fully integrate lexical patterns into phrase-level prosodic structures. A lack of complete integration between lexical and phrasal levels may account for the extended acquisition of prosodically conditioned vowel reduction, which in turn could account for the prolonged acquisition of English rhythm. [Work supported by NIH Grant R01 HD061458.]

9:25

5aSCa5. Weighting of prosodic cues in language discrimination by infants and adults. Chad J. Vicenik and Megha Sundara (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095)

Previous research has shown that infants and adults can discriminate between prosodically similar languages using only prosodic cues. These experiments were designed to determine whether listeners use pitch cues or segmental duration and timing cues (i.e., rhythm cues) in language discrimination. We tested American English learning 7-month-old infants and adults on their ability to discriminate between sentences in American English and German that had been re-synthesized to isolate different cues. Infants were presented with low-pass filtered speech and speech with re-synthesized sinusoidal pitch. Adults were presented with low-pass filtered speech and two conditions where sonorants were replaced with /a/ and obstruents were replaced with silence: one contained both pitch and durational cues and the other contained only durational cues. Pitch cues were important for infants as well as adults. For infants, pitch cues were necessary for successful language discrimination; neither segmental nor durational cues were sufficient. For adults, the presence of pitch prevented listeners from discriminating English and German. Only when pitch cues were removed did adults rely on durational differences to distinguish languages. These results demonstrate that to distinguish prosodically similar languages, specifically English and German, American English listeners' weight pitch cues over durational cues.

9:45

5aSCa6. Rhythm, tempo, and F0 in language discrimination. Tara Rodriquez and Amalia Arvaniti (Dept. of Linguist., UC San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, aarvaniti@ucsd.edu)

Existing literature on language rhythm suggests that languages can be discriminated in AAX "oddball" experiments only if they belong to different rhythm classes. Since exceptions to this pattern have been noted (e.g., the discrimination of Polish from English though both are said to be stress-timed), a series of five AAX experiments was run to test whether successful discrimination depends on factors other than rhythm class. English utterances were used as the base (AA) with Polish, Greek, Korean, Spanish, and Danish as the target languages. The stimuli were converted to sasasa while either retaining their original F0 or with flat F0; further, either the original tempo of each stimulus was retained or the stimuli (within each experiment) were manipulated to all have the same tempo (2 * 2 overall design, tempo * F0). The results show that tempo played the major role in discrimination with F0 cues being used when tempo differences between the two languages were eliminated. Rhythm class, on the other hand, played no significant part in discrimination suggesting that previous experiments may have interpreted as an effect of rhythm class the fact that tempo is often (though not always) slower in languages classified as stress-timed.

10:05–10:20 Break

Contributed Papers

10:20

5aSCa7. Quantifying rhythm: Interspeaker variation in %V. Margaret E L Renwick (Dept. of Linguist., Cornell Univ., Ithaca, NY 14850, mer56@cornell.edu)

The long-standing intuition that languages differ systematically in speech rhythm received support from an acoustic measure, %V (the ratio of vocalic material to the total duration of an utterance), which seemed to separate languages according to their perceived rhythms. Across studies, however, computations of %V values for target languages differ, raising the possibility that %V reflects not language-specific rhythmic habits, but the syllable structure of the particular utterances selected for analysis, and individual speaker differences. This paper replicates the %V methodology (using new recordings), and seeks to explain the high variability in %V by explicitly testing the link between %V, phonotactics, and individual speaker behavior. A high positive correlation is found between %V and an utterance's proportion of open syllables, while %V correlates negatively with the number of complex onsets in an utterance; these results support the hypothesis that utterance-level variation in syllable structure contributes substantially to the computed value of %V. However, the strength of this relationship varies across speakers. The goal of this paper is to determine the relative effects of utterance-level phonotactics, and individual speaker variation, on %V results, e.g., by comparing results when the same sentence is spoken by different speakers of a single language.

10:35

5aSCa8. Comparative electromagnetic articulographic study of English rhythm as produced by native and non-native speakers. Donna Erickson (1-11-1 Kamiasao Asao-ku, Kawasaki City, Kanagawa Prefecture, 215-8558 Japan), Atsuo Suemitsu (Japan Adv. Inst. of Sci. and Technol., Ishikawa, Japan), Yoshiho Shibuya (Kanazawa Medical Univ., Kanazawa, Japan), Sungbok Lee (USC, Los Angeles, CA), and Yu Tanaka (UCLA, Los Angeles, CA)

Previous work suggests that concurrent changes in jaw displacement and formant frequencies may manifest rhythm of spoken American English utterances [e.g., Erickson, in *Proceedings of the Speech Prosody 2010*, Chicago (May 2010), p. 1]. This paper compares jaw displacement and

corresponding formant frequencies of monosyllabic American English words produced on low vowels in four word phrases with varying positions of emphasis as spoken by three native speakers and three Japanese speakers of English. Preliminary findings suggest an alternating pattern of strong-weak jaw displacement along with corresponding formant changes for the native speakers. This pattern was not consistently seen for the non-native speakers and seemed to vary as a function of their skill-level in spoken English.

10:50

5aSCa9. Rhythmic conversion between speakers of different dialects. Jelena Krivokapic (Dept. of Linguist., Yale Univ., 370 Temple St., 302, New Haven, CT 06520-8366, jelena.krivokapic@yale.edu)

Speakers' convergence to each other's production of segments has been well established [Sancier and Fowler, J. Phon. **25**, 421–426 (1995); Nielsen, J. Phon. **39**, 132–142 (2011)]. Less is known about prosodic convergence, but it has been identified for stress and pitch accent [Krivokapic, JASA, **127**, 1851 (2010); Krivokapic, JASA **129**, 2658 (2011)]. In an acoustic experiment, rhythmic conversion between speakers of American and Indian English is examined. The two languages differ both in the stress patterns of individual lexical items and in their global rhythmic properties, with Indian English being syllable-timed and American English stress-timed. Changes in speakers' productions are examined using a synchronous speech task [Cummins, ARLO **3**, 7–11 (2002), Zvonik and Cummins, Proc. Eurospeech **2003**, 777–780 (2003)], where two speakers read sentences at the same time. Eight subjects (four dyads), each consisting of one Indian and one American speaker, read a short story that contained 12 words in which the two dialects differ in stress pattern and four sentences in which the global rhythmic properties are tested. The data of the dyad examined to date indicate convergence to a more stress-timed pattern for the Indian speaker.

11:05

5aSCa10. Analysis of rhythmic entrainment in speech production using real-time magnetic resonance imaging. Louis Goldstein, Michael I. Proctor, and Adam Lammert (Univ. of Southern California, Dept. of Linguist., Los Angeles, CA 90089)

It has been shown that subjects may recruit extra-linguistic articulators during the rhythmic production of speech (Tiede, 2010), that sympathetic head movement is associated with dysfluencies in running speech (Hadar, 1984), and that the magnitude of this activity positively correlates with both speaking rate and increased effort in accelerating speech production tasks (Hadar, 1991; Tiede, 2011). It is unclear at what level of linguistic organization this entrainment occurs. We examined patterns of articulation and head movement during an accelerated repetitive speech task involving multi-gestural segments, in order to gain more insights into the coordinative bases of this behavior. Real-time magnetic resonance imaging was used to examine the production of paired English words, contrasting along various articulations, e.g., "cop-top," "kid-kim," "flee-free," "muck-duck." Subjects' upper airways were imaged in the midsagittal plane while producing trochaic repetitions of each word pair in time to an accelerating metronome. Labial, tongue tip, tongue body and velic activity, as well as gross head movement, were tracked with direct image analysis (Lammert, 2010). Spontaneous head nodding was frequently observed in coordination with the prosodic foot. Nodding was observed to increase in amplitude with speech rate and during production of stimuli containing complex segments. [Work supported by NIH].

11:20

5aSCa11. Discriminating language and talker using non-linguistic measures of rhythm, spectral energy and f0. Kathy M. Carbonell (Dept. of Speech, Lang. & Hearing Sci., Univ. of Arizona, 1131 E 2nd St., Tucson, AZ 85721, kathyc@email.arizona.edu), Kaitlin Lansford, Rene Utianski, Julie Liss (Arizona state Univ., Tempe, AZ 85287-0102), and Andrew Lotto (Univ. of Arizona, Tucson, AZ 85721)

Recent studies have shown that rhythm metrics calculated from amplitude envelopes extracted from octave bands across the spectrum (the envelope modulation spectrum or EMS) can reliably discriminate between spoken Spanish and English even when produced by the same speakers [Carbonell *et al*., J. Acoust. Soc. Am. **129**, 2680]. Additionally, bilingual speakers could be discriminated fairly *seven females and five males* well on EMS variables even across sentences spoken in the different languages. In the current study, EMS, a general acoustic measure with no reference to phoneme/linguistic entities, was supplemented with measures of the mean and variance of spectral energy in each octave band as well as the mean and variance of fundamental frequency. Using stepwise discriminant analysis and the set of bilingual productions of Spanish and English, it was determined that language discrimination was excellent using both EMS and spectral measures, whereas spectral and f0 measures were most informative for speaker discrimination. The results demonstrate that this suite of easily-calculated acoustic measures provides abundant information about differences between languages as well as inherent differences in speakers that are independent of language spoken. [Work supported by NIH-NIDCD.]

11:35

5aSCa12. Temporal regularity in speech perception: Is regularity beneficial or deleterious?. Eveline Geiser (Dept. of Brain and Cognitive Sci., Massachusetts Institut of Technoloy, 43 Vassar St. 46-4033, Cambridge, MA 02139, egeiser@mit.edu), Stefanie Shattuck-Hufnagel (Res. Lab. of Electrons, MIT, Cambridge, MA 02139), and John D. E. Gabrieli (Dept. of Brain and Cognitive Sci., Massachusetts Institute of Technology, Cambridge, MA 02139)

Speech rhythm is of crucial importance for correct speech perception and language learning; for example, the specific rhythm is among the first things infants learn about their native language (Ramus, 2000). This study aimed to investigate the importance of speech rhythm in second language learning. German pseudosentences were presented to subjects in two conditions: spoken with a normal conversational speech rhythm versus with a speech rhythm containing a temporally regular beat. Nine native English speakers with 3.5 ± 1.6 years of German training repeated each sentence after hearing it once over headphones. Responses were transcribed using to the IPA and analyzed for the number of correct, false, and missing consonants, as well as for consonant intrusions. The over-all number of correct reproductions of consonants did not differ between the two experimental conditions. However, the experimental condition comprising a normal conversational speech rhythm resulted in significantly fewer consonant intrusions than the condition containing a beat. These results highlight the importance of speech rhythm in language perception/production and suggest, in particular, that second language learning may be facilitated by language-specific temporal grouping. [During the preparation of this work E. G. was supported by the Swiss National Science Foundation (SNF;PBZHP1-123304).]

11:50–12:10 Panel Discussion

Session 5aSCb

Speech Communication: Speech Perception by Native and Non-Native Speakers (Poster Session)

Grant McGuire, Chair
Dept. of Linguistics, Univ. of California, Santa Cruz, Santa Cruz, CA 95064

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

5aSCb1. Stimulus direction predictability dampens auditory neural responses to pitch-shifted voice feedback. Oleg Korzyukov, Lindsey Sattler, Roozbeh Behroozmand, and Charles Larson (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208)

Models of neural mechanisms of voice control propose that voice auditory feedback is compared with an internal representation of the predicted voice output that is based on efference copies of motor commands. In addition, sensory memory from previous productions can also help the brain form predictions about incoming auditory feedback during vocal production. Previous studies have shown that the auditory evoked neural responses are maximally dampened in conditions where the incoming feedback closely matches the internal predictions. The present study aimed to determine if the predictability of stimulus direction influences the ERP responses to pitch shifts in voice auditory feedback. Subjects were tested with all upward stimuli, all downward stimuli, or randomized stimulus directions in separate blocks of trials. The N100 ERP response had a greater amplitude in conditions where the stimulus direction was randomized (unpredictable) compared to constant direction stimulus conditions (predictable), regardless of whether the stimuli were upwards or downwards. These findings suggest that auditory neural responses to predictable stimulus direction are dampened compared with unpredictable stimuli possibly because the predictable stimuli are suppressed by the internal predictions formed by the efference copy and/or sensory memory mechanisms.

5aSCb2. Duration of American English vowels and its effects on intelligibility for native and non-native speakers. Chia-Tsen Chen, Chang Liu, and Su-Hyun Jin (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX 78712)

Durations of 12 American English vowels were measured for English-, Chinese-, and Korean-native speakers. Results showed that vowel duration was significantly affected by speaker groups. That is, vowels produced by English- and Korean-native speakers were significantly longer than vowels produced by Chinese-native speakers. The patterns of vowel duration as a function of vowel category were quite similar between native and non-native speakers. However, Mandarin Chinese vowels did not differ in duration across Chinese-native speakers, while durations of Korean vowels were significantly different across Korean-native speakers. These results suggest that non-native speakers were able to follow the duration pattern of native speakers in their vowel production. When vowel duration was equalized across vowel category with the central vowel nucleus remained, vowel intelligibility significantly dropped about 5% compared to vowels presented in the syllabic context for native and non-native speakers. The perceptual effects of vowel duration on vowel intelligibility appeared to be the same for native and non-native speakers, independent of speaker language background and the second language proficiency level.

5aSCb3. Perceptually relevant information in energy above 5 kHz for speech and singing. Brian B. Monson, A. Davi Vitela, Brad H. Story, and Andrew J. Lotto (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, bbmonson@email.arizona.edu)

Although high-frequency energy (HFE) is typically ignored in the speech perception and music literature, previous work has demonstrated that listeners are able to extract information from the structure of energy above 5 kHz. When exposed to HFE stimuli, listeners could accurately determine the mode of production (speech versus singing) and the gender of the speaker/singer. However, HFE in these previous experiments was presented in isolation at levels higher than what is typical of normal speech and singing. In the series of experiments presented here, the stimuli were presented at more appropriate levels. Additionally, speech-shaped masking noise from 0 to 5.6 kHz (set at the mean levels for an average speaker/singer) was used to determine if HFE is usable by listeners even in the presence of masking by the more intense energy below 5 kHz in typical speech/voice listening conditions. Results indicate that even when the HFE stimuli are presented at typical levels and in the presence of a masker, listeners are still able to discriminate production modes and gender. These results suggest that HFE contains information that normal-hearing listeners can potentially access during the perception of speech and singing. [Work supported by NIH-NIDCD.]

5aSCb4. Stability of compensatory behavior for real-time perturbations of vowel formants. Takashi Mitsuya (Dept. of Psych., Queen's Univ., 62 Arch St., Kingston, ON K7L3N6, Canada, takashi.mitsuya@queensu.ca), Ewen N. MacDonald (Dept. of Psych. & Elec. and Comput. Eng., Queen's Univ.), and Kevin G. Munhall (Dept. of Psych. & Otolaryngol., Queen's Univ.)

While we talk, not only do we listen to the speech sounds other people make but also we listen to our own voice in order to control the phonetic/phonological details of the sounds. One example of this relationship is demonstrated by real-time formant perturbation studies which show that talkers automatically change their formant production when auditory feedback does not match with the vowel they intend to produce. The reported results are consistent across studies, yet the variability between talkers is usually quite large, with some talkers showing a large magnitude compensation while others compensate modestly. To date, the degree to which talkers compensate has been assumed to be stable, but this has never been directly examined. The current study tested the stability of compensatory behavior for perturbed formant shifted feedback by repeatedly testing a group of talkers over the course of a few weeks to measure the variance in formant values and compensation across experimental sessions. The results will be discussed in terms of sensorimotor adjustment and speech production models.

5aSCb5. Overall pitch height as a cue for lexical tone perception. Jing Shen, Diana Deutsch (Dept. of Psych., Univ. of California San Diego, La Jolla, CA 92093, jshen@psy.ucsd.edu), and Jinghong Le (School of Psych., East China Normal Univ., Shanghai 200062, China)

Absolute pitch has been hypothesized to be involved in processing lexical tones in tone languages, which associate pitch information with verbal labels. Since possessors of absolute pitch in music utilize the overall pitch of a tone in making identification judgments, the hypothesis was tested that native speakers of Mandarin utilize the overall pitch of a lexical tone as a cue to retrieve its tone label. In a reaction time task, Mandarin syllables in all four tones were presented both in their original forms and also transposed to four different levels of pitch height; subjects listened to each token and judged whether or not its meaning corresponded to the original. It was found that although Mandarin syllables with transposed overall pitch heights were judged to be the same tones as the original tokens, subjects were significantly slower in making judgments for those syllables that were transposed to different levels of pitch height, compared to those that were presented at their original pitch heights. This effect was most extreme for tones 1 and 3. These findings suggest that overall pitch height serves as a cue for identifying lexical tones and further strengthens the link between absolute pitch and lexical tone perception.

5aSCb6. Perceptual importance of the voice source spectrum from H2 to 2 kHz. Jody Kreiman (Head and Neck Surgery, UCLA School of Medicine, 31-24 Rehab. Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu) and Marc Garellek (UCLA, Los Angeles, CA 90095)

Modeling the source spectrum requires understanding of the perceptual importance of different spectral-domain attributes of the voice source. Although the roles of H1-H2 and high-frequency harmonics in quality perception are somewhat understood, the extent of spectral detail that is perceptually significant in other frequency ranges is not known. This experiment examined the perceptual importance of amplitude difference between the second and fourth harmonics (H2-H4), which varies with linguistically significant changes in mode of phonation and pitch. To further determine if sensitivity depends on H1-H2 or on F0, we synthesized non-pathological male (F0=100 Hz) and female voices (F0 = 200 Hz) and manipulated the values of H2-H4 by 1 dB increments over a 20 dB range, for three different H1-H2 levels. Listeners heard pairs of stimuli in a 1-up, 2-down adaptive paradigm, and variations in just-noticeable-difference for H2-H4 as a function of F0 and H1-H2 were assessed with analysis of variance (ANOVA). Discussion will focus on the extent to which H2-H4 sensitivity covaries with F0, H1-H2, and the shape of the spectrum above H4. [Work supported by NSF and NIH.]

5aSCb7. Perceptual benefits of English coda-stop glottalization. Marc Garellek (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, marcgarellek@ucla.edu)

Glottalized stops in codas (e.g., [t]) are commonly found in English, often when the coda-stop is unreleased. Glottalized phonation is known to increase the amplitudes of higher harmonics in the spectrum, possibly due to faster closing velocity of the vocal folds (Stevens, 1977). This suggests that glottalized phonation occurring before an unreleased stop may be useful to the listener for perceiving the formant transitions into the stop, because the higher frequencies will be amplified during glottalized phonation leading into the glottalized stop. To test this hypothesis, a phoneme monitoring task was conducted with native English speakers, who were asked to monitor for /t/ in onsets and codas of English words. The target stimuli were monosyllables ending in a coda-/t/, e.g., /bit/. All the stimuli were uttered by a phonetically trained speaker and varied according to whether the word was glottalized and/or ended in an unreleased stop. Preliminary results from the listeners' data suggest that their reaction time is faster for glottalized coda-/t/ (whether released or not), and that their detection accuracy improves when the coda-stop is both released and glottalized. Discussion will involve whether glottalization is a cue to formant transitions in general or to the presence of /t/ specifically.

5aSCb8. The role of vowel transitions in the perception of Mandarin sibilants. Yung-hsiang Shawn Chang (Dept. of Linguist., Univ. of Illinois, 707 S. Mathews, Ave., 4080E FLB, Urbana, IL 61801, yhchang@illinois.edu)

Vocalic formant transitions have been reported to play a role in identification of sibilants in English (e.g., Whalen, 1981), Shona (Bladon *et al*., 1987), and other languages. In distinguishing the Mandarin /s/-/ʂ/ contrast, while F2 at vowel onset is suggested to reflect the vocal tract configuration at the moment of fricative release into the vowel (Stevens *et al*., 2004), it has never been studied whether this acoustic cue contributes to phoneme perception. This study used a F2 continuum spliced between a fricative highly confusable between /s/ and /ʂ/ and the vowel /a/ for an identification task on 16 native listeners. The results did not show a categorical perception effect on /sa/ and /ʂa/ distinction along the F2 continuum. The second experiment used the naturally produced /sa/ and /ʂa/ whose vocalic transitions were cross-spliced to investigate if the listeners were affected by misleading formant transitions. Listeners' goodness ratings revealed no sensitivity to the acoustic manipulation. It was concluded that F2 is not a primary cue for the place distinction of Mandarin /s/ and /ʂ/. Our findings support Wagner *et al*.'s (2006) observation that the contribution of formant transitions to fricative perception is language-specific and depends on spectral similarity among target fricatives.

5aSCb9. Vowel nasality as a perceptual cue to pharyngeal consonants in moroccan arabic. Georgia, E. Zellou (Dept. of Linguist., 295 UCB, Hellems 290, Boulder, CO 80309-0295, georgia.weissman@colorado.edu)

Pharyngeal consonants have been shown cross-linguistically to be coarticulated with velopharyngeal port opening (Bladon and Al-Bamerni, 1982, Elgendy, 2001). We examined whether coarticulatory vowel nasality is a perceptual cue to an adjacent pharyngeal, and/or nasal, consonant in Moroccan Arabic (MA). Monosyllabic MA words spoken by a native speaker containing a pharyngeal, voiced /ʕ/ or voiceless /ħ/, or a nasal /n,m/ were cross-spliced with vowels from other contexts creating two conditions per word: "Oral" stimuli were pharyngeals cross-spliced with pharyngealized vowels (i.e., adjacent to /$d^{ʕ}$/) or nasals cross-spliced with oral vowels (i.e., adjacent to /d/). "Nasal" stimuli were cross-spliced with nasal vowels. The stimuli were presented to nine native MA speakers in a timed lexical repetition task. An RM ANOVA of repetition times showed a significant interaction between nasality, consonant, and directionality [$F(7,324) = 3.755$, $p=0.046$]. For anticipatory coarticulation, vowel nasality resulted in faster RTs, indicating facilitated perception for voiced /ʕ/ ($t(52) = 1.67$, $p=.001$) but not voiceless /ħ/. For carryover coarticulation, vowel nasality resulted in faster RTs for /ħ/ ($t(52)=2.05, p = .02$), but not /ʕ/. In both directions, vowel nasality resulted in faster RTs for nasals. These results indicate that MA listeners use vowel nasality as a perceptual cue to a word with a pharyngeal segment.

5aSCb10. Sharing the beginning versus sharing the end: Spoken word recognition in the visual world paradigm in Japanese. Hideko Teruya and Vsevolod Kapatsinski (Dept. of Linguist., 1290 Univ. of Oregon, Eugene, OR 97403)

Several studies have suggested that in English spoken word recognition, words sharing beginnings compete more than words sharing ends (i.e., the cohort competitor has an advantage relative to the rime competitor). We investigated the relative activation differences between the cohort competitor and the rime competitor in Japanese CVCV words using the visual-world paradigm (Allopenna *et al*., 1998). Results were analyzed using linear mixed effects models (Magnuson *et al*., 2008). Participants fixated referents of rime competitors more than those of cohort competitors when the cohort competitor shared only the first consonant with the target while the rime shared the rest of the word (VCV). When both competitors shared a syllable with the target, the rime competitor was fixated as much as the cohort competitor and the recognition of the target was delayed. The results suggest a syllabic or moraic basis to Japanese spoken word recognition, where lexical activation requires at least one mora of the word to match the signal. The weakness of cohort effects in Japanese challenge fully incremental models of word recognition. Whether the results would also hold for English is unclear as previous visual world spoken word recognition studies in English did not systematically manipulate degree of overlap.

5aSCb11. Low-level adjustments to gross-categorical mismatches in the perception of accented speech. Meghan Sumner (Dept. of Linguist., Stanford Univ., Stanford, CA 94305-2150, sumner@stanford.edu) and Samuel Tilsen (Cornell Univ., Ithaca, NY 14853-4701)

Variation in speech abounds. How do listeners understand a single word produced any number of ways as an instance of one particular word and not another? Sumner (2011) found that listeners perceive unaspirated voiceless stops [p] as a /p/ (rather than the native /b/) when exposed to more variation and when listeners are shifted down a VOT continuum in 5 ms steps. To understand this behavior, we compared order effects and noise in a phoneme categorization task in two conditions, both shifting listeners down a VOT continuum from a native /p/ (60 ms VOT) to a non-native /p/ (0 ms VOT). In one condition, listeners heard p-initial words with VOTS that made small steps down (5 ms) the continuum, with interspersed big jumps up (15–20 ms). In the other, listeners heard the same words with big jumps down and small steps up. Replicating Sumner (2011), listeners were successful in perceiving an unaspirated stop as voiceless. Interestingly, a difference between the conditions emerged, with greater boundary shifts made with big jumps down and small steps up. These results show (1) within-native-category fluctuations have no effect on categorization and (2) big differences in VOT serve as categorization anchors, while small steps drive perceptual adjustments.

5aSCb12. Investigating the effect of speech stimuli on temporal masking release. Evelyn Davies-Venn, Peggy Nelson, Yingjiu Nie, Adam Svec, and Katare Bhagyashree (Univ. of Minnesota, SLHS, 164 Pillsbury Dr., SE, Minneapolis, MN 55455)

The effect of masking release is still the source of numerous active investigations. However, differences in findings are sometimes noted among studies, especially related to potential gate frequency effects. The present study investigated the effect of masking release on listeners using a variety of speech materials . The main goal of this study was to investigate the effect of speech material on measures of masking release for listeners with normal hearing and hearing loss. The test stimuli were IEEE sentences and modified spondee words. To eliminate confounds of audibility and duration, the test stimuli were equated for duration and audibility. Listeners were tested across a wide range of audibility and gate frequencies. Performance and masking release results will be presented for these speech stimuli. [Work supported by NIDCD 008306.]

5aSCb13. Perception of Korean alveolar fricatives by American learners. Ahrong Lee and Hanyong Park (Dept. of Linguist., Univ. of Wisconsin-Milwaukee, 3243 N. Downer Ave. Milwaukee, WI 53211, ahrlee@uwm.edu)

This paper presents experimental data on how native English-speaking adult learners perceive Korean voiceless alveolar fricatives in a task soliciting sensitivity to prosodic context. College students in a second semester Korean class were asked to identify tokens of the two contrasting Korean fricatives, lax /s/ and tense /ss/, using letters of the Korean alphabet while rating their confidence on a scale from 1 (not confident at all) to 7 (very confident). Four native speakers of Korean ($M = 2$; $F = 2$) produced 24 tokens of the target sounds in construction with /a/ in three different prosodic contexts, viz., prevocalic word-initial, pre-emphatic intervocalic, and post-emphatic intervocalic positions. The results show a tendency for subjects to classify both Korean fricatives as lax /s/ under most conditions, except that tense /ss/ was often separately identified when preceded by an emphasized vowel. Phonetically, Korean /ss/ is of greater duration and intensity in a post-emphatic context than elsewhere, hence more distinct from /s/ in both Korean and English and so, apparently, more easily identifiable for beginning learners.

5aSCb14. English speakers' discrimination of Korean plain versus aspirated affricates. Jessica Maye and Jenna Silver Luque (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208)

The Korean three-way voicing distinction has been of much interest to phoneticians. Korean obstruents may be plain, tense, or aspirated, which differ in VOT, closure duration, and f0 of vowel onset (Cho *et al.*, 2002; Kim, 1994). Previous research on Korean stops has shown that English speakers perceive all three categories as voiceless in initial position (Shin, 2001). The present study extends this to Korean affricates, which are partly cued by frication duration, an acoustic cue that is distinctive in English for differentiating fricatives from affricates (Repp *et al.*, 1978). Twenty monolingual English speakers participated in the study. The stimuli were six tokens each of the plain and aspirated Korean affricates ([tSa] and [tSha]), naturally produced by a male native speaker of Seoul Korean. Participants completed an AX discrimination task, following five practice trials in which they received feedback on their answers. Results showed very poor discrimination that did not differ from chance (mean $d' = 0.892$; $p < 0.001$). These results indicate that despite the relevance of an additional acoustic cue that is distinctive in English (frication duration), the Korean contrast between plain versus aspirated affricates is difficult for English speakers to discriminate, paralleling previous research on Korean stops.

5aSCb15. The perception of Japanese pitch accent in "whispered" speech. Yukiko Sugiyama (Faculty of Sci. and Technol., Keio Univ., 4-1-1 Hiyoshi, Yokohama, Kohoku-ku, Yokohama 223-0061, Japan, yukiko_sugiyama@mac.com)

The perception of Japanese pitch accent was investigated using "whispered" speech in which the F0 was artificially removed. While the F0 is said to be the primary cue for pitch accent in Japanese, it is not certain whether acoustic correlates other than F0 exist. The results of previous production studies that examined vowel duration or devoicing as a correlate of pitch accent are not consistent. The present study attempts to find correlates of pitch accent from the other end, i.e., perception. A native speaker of Tokyo Japanese produced 14 disyllabic minimal pairs that differed only in the presence or absence of accent (e.g., /hana*/ "flower" when accented vs. /hana/ "nose" when unaccented) in a carrier sentence. The utterances were then edited by replacing the F0 by random noise, creating artificial "whispered" speech. Twenty-two native speakers of Tokyo Japanese identified the words they heard in two kinds of stimuli, the 14 minimal pairs as produced by the speaker and the whispered speech. The results suggest evidence of pitch accent in whispered speech. Implications of the results for the nature of Japanese pitch accent and the perception of accent will be discussed.

5aSCb16. Discrimination of Japanese affricate contrast by native speakers of Korean. Hiromi Onishi (Dept. of East Asian Studies, Univ. of Arizona, LSB 102, 1512 First St., Tucson, AZ 85721, honishi@email.arizona.edu)

It is known that Korean learners frequently experience difficulty producing the Japanese syllable /tsu/, which is often perceived as the syllable /tʃu/ by native Japanese speakers. The investigator administered an AXB discrimination test using these affricates (/tsu/-/tʃu/) in order to examine how well Korean native speakers can discriminate this contrast. These affricates were also tested in the shape of a whole syllable with the high-back vowel (/tsu/-/tʃu/). In addition, discrimination of the high-back vowel following the two affricates was also examined. The overall results suggest that native Korean speakers are capable of discriminating Japanese syllable /tsu/ from /tʃu/. However, the participants scored significantly lower for the vowel contrast than the syllable and consonant contrasts. Following the AXB test, an identification test was conducted using Hangul orthography in order to investigate the perceptual assimilation patterns. The results showed that the two syllables are assimilated to different syllables in Korean. The identification of the vowels showed that the /u/ in the contexts of /ts__/ was consistently assimilated to a single Korean category while some kind of vowel with preceding semivowel /j/ was selected most frequently for /u/ in the contexts of /tʃ__/. These results are discussed based on Best's perceptual assimilation model.

5aSCb17. Korean perception of the Japanese voiceless alveolar fricative /s/. Heather E. Simpson (Dept. of Linguist., Univ. of California Santa Barbara, Santa Barbara, CA 93106, hsimpson@umail.ucsb.edu)

Korean voiceless obstruents exhibit a relatively unique phonemic contrast based on laryngeal activity, a three-way tense, lax, and aspirated contrast for stops and affricates and a two-way tense and lax contrast for the alveolar fricative /s/. The relative primacy of various cues to the distinction, particularly for the affricate and fricative contrasts, remains controversial. Patterns in the adaptation of English loanwords have been used to infer the phonological basis for the mapping of English /s/ to Korean, however perception studies have shown that loanword adaptations do not fully account for the synchronic phonological mapping from English loans by Korean native speakers (Schmidt 1996). Mapping from Japanese /s/ to Korean has been much less studied, but according to a loanword study by Ito *et al.* (2006), Japanese /s/ in word-initial and medial position is consistently mapped to Korean lax /s/, and geminate /s/ is consistently mapped to Korean tense /s*/. However, there appears to have been no comparable perception study to show synchronic mapping by Korean native speakers. The current study undertakes a preliminary evaluation of perceptual mapping of Japanese /s/ to Korean and finds that though the word initial Japanese /s/ conforms to loan-word predictions, word-medial /s/ does not.

5aSCb18. Non-native Japanese learners' perception of vowel length contrasts in Arabic and Japanese. Kimiko Tsukada (Dept. of Int. Studies, Macquarie Univ., NSW 2109, Australia, kimiko.tsukada@mq.edu.au)

This study examined the perception of short vs. long vowel contrasts in Arabic and Japanese by four groups of listeners differing in their linguistic backgrounds: native Arabic (NA), native Japanese (NJ), non-native Japanese (NNJ), and Australian English (OZ) speakers. Listeners' first languages differed in the extent to which vowel duration is used contrastively. In both Arabic and Japanese, vowel length is phonemic. English, on the other hand, utilizes vowel duration in a more limited way. Of interest was the discrimination accuracy of NNJ listeners who learned Japanese as a second language beyond childhood. As expected, the NA and NJ groups discriminated their native contrasts more accurately than all the other groups while the NNJ listeners showed a significant shift in their perceptual behavior and outperformed the OZ listeners who had no knowledge of Japanese in discriminating the Japanese vowel length contrasts. Furthermore, NNJ was the only group who showed a balanced pattern of discrimination accuracy in Arabic and Japanese. Taken together, the results obtained in this study suggest that NNJ learned to discriminate Japanese vowel length contrasts to some extent, but the learning did not carry over cross-linguistically to the processing of vowel length contrasts in an unknown language, Arabic.

5aSCb19. Perception of initial stops in tonal and non-tonal Korean. Hyunjung Lee and Allard Jongman (Dept. Linguist., Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, hyunjung@ku.edu)

The current study investigated the perception of the three-way distinction among voiceless stops in non-tonal Seoul Korean and tonal Kyungsang Korean. We addressed whether listeners from these two dialects differ in the way they perceive the three stops. Twenty Korean listeners (nine from Seoul and eleven from Kyungsang) were tested with stimuli in which VOT and F0 were systematically manipulated. Analyses of the perceptual identification functions show that the results replicate the reported phonetic trading relationship between VOT and F0 in Seoul Korean and that this trading relationship is observed in Kyungsang as well. However, the trading relationship differs between the two dialects. Logistic regression analysis further shows that the two dialects use the perceptual cues differently. While both VOT and F0 are effective for Seoul Korean, only VOT is effective for the identification of all three stops for Kyungsang Korean. A similar pattern has been observed acoustically [Lee and Jongman, J. Acoust. Soc. Am. **127** (3), 2023–2023 (2010)]. The results will be discussed in terms of the close link between perception and production across the two different dialects.

5aSCb20. Lexical tone processing by monolingual and bilingual speakers of tone and non tone languages. D. Kyle Danielson (Dept. of Linguist., Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, dkd@ualberta.ca)

This experiment tested early Chinese–English bilinguals (age of arrival in Canada $<$ 2 years) for lexical encoding processes of F0 contour (tone). Earlier experimentation (e.g., Mattock and Burnham, 2006) has indicated that processing of lexical tone differs between the two corresponding monolingual groups, demonstrating that English monolingual infants begin to disregard F0 contour differences in non word minimal pairs by the age of nine months, while Chinese monolingual infants do not exhibit this behavior. In the present study, three groups of participants were tested using a short-term memory encoding task (Dupoux *et al.*, 2010) with non word minimal pairs differentiated only by F0 contour. Although the tone contours utilized in the experiment were non-native to all speakers, Chinese-dominant bilinguals (adult arrival in Canada) performed significantly better than English monolinguals in recalling long non word sequences differentiated only by these contours, while their performance in simple phoneme-differentiated sequences (e.g., [mu, fu]) was equal to that of the English speakers. However, the target group of early Chinese–English bilinguals produced scores corresponding to a bimodal distribution, with some speakers correlating to English monolinguals' performance and others corresponding to Chinese-dominant speakers' scores. Correspondence to one mode or another was analyzed using a series of sociolinguistic factors.

5aSCb21. Effect of vowel height and nasal coda on Mandarin tone perception by second language learners. Tzu-Yun Lai (Foriegn Lang. Dept., National Chiao Tung Univ., No. 513 Female 2 Domitory, 1001 University Rd., Hsinchu, Taiwan 300, Taiwan) and Yi-Wen Su (National Chiao Tung Univ., Taiwan 300, Taiwan)

The present study aims to investigate whether second language (L2) learners of Mandarin will be affected by vowel height sequences and post-vocalic nasals in perceiving Mandarin tones. A total of 101 stimuli from a female native speaker of Mandarin were recorded for constructing the main experiment. Eight Mandarin L2 learners and two native speakers were recruited. The result indicated that L2 learners had a higher error rate when perceiving H-L-H (high-low-high) and nasal coda. However, perceiving H-L (high-low) or L-H (low-high) in tone 2 and tone 3, L2 learners easily mix up these together.

5aSCb22. Effects of speaker variability and noise on identifying Mandarin fricatives by native and non-native listeners. Chao-Yang Lee, Yu Zhang, Ximing Li (Div. of Commun. Sci. & Disord., Ohio Univ., Athens, OH 45701), Liang Tao, and Z. S. Bond (Ohio Univ., Athens, OH 45701)

A fundamental issue in speech perception is how listeners with different characteristics deal with various sources of acoustic variability. This study examined how speaker variability and noise affected native and nonnative identification of the place contrast in voiceless Mandarin fricatives. Monosyllabic Mandarin words fa, sa, xia, and sha produced by three female and three male speakers were mixed with five levels of speech-shaped noise. The stimuli were presented either blocked by speaker or mixed across speakers to 40 native Mandarin listeners and 52 English-speaking listeners with various amounts of Mandarin experience. It was predicted that the mixed presentation and noise would affect nonnative identification to a greater extent because of the imperfect knowledge of the target language. The results showed that nonnative performance was compromised to a greater extent by noise but not by speaker variability. These results were compared to previous studies which compared native and nonnative perception of segmental and suprasegmental contrasts in other adverse conditions. It was concluded that not all sources of acoustic variability affect native and nonnative speech perception similarly.

5aSCb23. Listeners retune phoneme boundaries across languages. Eva Reinisch, Andrea Weber, and Holger Mitterer (Max Planck Inst. for Psycholinguistics, P.O. Box 310, 6500 AH Nijmegen, The Netherlands, eva.reinisch@mpi.nl)

Listeners can flexibly retune category boundaries of their native language to adapt to non-canonically produced phonemes. This only occurs, however, if the pronunciation peculiarities can be attributed to stable and not transient speaker-specific characteristics. Listening to someone speaking a second language, listeners could attribute non-canonical pronunciations either to the speaker or to the fact that she is modifying her categories in the second language. We investigated whether, following exposure to Dutch-accented English, Dutch listeners show effects of category retuning during test where they hear the same speaker speaking her native language, Dutch. Exposure was a lexical-decision task where either word-final [f] or [s] was replaced by an ambiguous sound. At test listeners categorized minimal word pairs ending in sounds along an [f]-[s] continuum. Following exposure to English words, Dutch listeners showed boundary shifts of a similar magnitude as following exposure to the same phoneme variants in their native language. This suggests that production patterns in a second language are deemed a stable characteristic. A second experiment suggests that category retuning also occurs when listeners are exposed to and tested with a native speaker of their second language. Listeners thus retune phoneme boundaries across languages.

5aSCb24. Speech-in-noise perception for new sentence-recognition materials: Normative data for a diverse non-native listener population. Rajka Smiljanic (Linguist., Univ. of Texas at Austin, 1 University Station B5100 Austin, TX 78712-0198), Lauren Calandruccio, and Stacey Rimikis (The City Univ. of New York, NY 11357)

Even though the US population is becoming more diverse, there are no normalized sentence recognition tests for non-native English listeners. Recently, a large set of new test materials aimed at assessing speech

recognition and hearing abilities for native and non-native listeners were developed by Smiljanic and Calandruccio [J. Acoust. Soc. Am. **128**(4), 2486]. Five hundred unique sentences divided into 20 lists were created. The target keywords were chosen from the most frequent lexical items occurring in naturally elicited conversations with 100 non-native talkers with varied linguistic and cultural backgrounds. The lists were equalized for syntactic structure, syllable count, high-frequency phonemes, and word frequency. Perception results revealed equal mean performance across lists for native listeners. Currently, we are in the process of collecting normative data for these materials. Results of speech-in-noise tests for a diverse group of non-native English listeners with various native language backgrounds and levels of English experience will be presented. Regression analyses will be performed to investigate how the listeners' linguistic experience, competence, and proficiency relate to their listening performance on these new sentence-recognition materials. The results of this study add to our current understanding of non-native listener-related factors that affect speech recognition in adverse listening conditions.

5aSCb25. Phonemic perception under noise condition by early and late learners of English. Tetsuo Harada (Graduate School of Education, Waseda Univ., 1-6-1 Nishi Waseda, Shinjuku, Tokyo 169-8050, Japan, tharada@waseda.jp)

This study investigated long-term age effects of minimal input in an English as a foreign language (EFL) environment on the perception of English phonemes. Ten native speakers of American English, 10 university students who started studying English in Japan for a few hours a week between ages of three and eight (early learners), and 10 university students who began study in junior high school at the age of 12 or 13 (late learners) participated in a phonemic discrimination test. The selected target phonemes were tense vs. lax vowels ([i, ɪ], [u, ʊ]) and word-initial approximants ([l, r]). Three monosyllabic words were selected for each phoneme (e.g., *beat, bit*). Each pair of words was tested as six different tri-word trials (e.g., AAB, ABA) with and without the presence of a white noise. The participants were asked to choose the odd out in each trial. Results showed that although the early and late learners did not significantly differ across the phonemes in the discrimination test given without any noise, the former outperformed the latter under noise condition ($p < 0.05$). The findings support Lin *et al.*'s (2004) hypothesis that early learners are likely to establish more robust phonemic categories than late learners.

5aSCb26. Spanish learners' perception of American and Southern British English vowels. Paola Escudero (MARCS Auditory Labs., Bldg. 1, Univ. of Western Sydney, Bullecourt Ave., Milperra, NSW 2214, Australia) and Katerina Chladkova (Univ. of Amsterdam, 1012VT Amsterdam, The Netherlands)

Second-language (L2) studies show that learners differ in their vowel perception patterns, which can be attributed to their L2 proficiency or to different initial stages at the onset of L2 learning. The present study assessed the L2 perception of Southern British English (SBE) and American English (AE) vowels in 196 Spanish learners of English with different general comprehension proficiencies. Escudero and Chladkova (2010) showed that naive Spanish listeners perceived AE vowels differently than SBE vowels. The current results show that L2 learners' performance matches that of naive listeners. For instance, learners identified AE /ae/ correctly in only 3% of the time, as opposed to 32% for the same SBE vowel. This is because naive listeners assimilate AE /ae/ to Spanish /e/, which is the same native vowel they choose for AE /I/ and /E/, while SBE /ae/ is mostly assimilated to Spanish /a/. In addition, there was a significant effect of L2 proficiency on vowel identification accuracy and on the use of vowel duration. More proficient learners identified L2 vowels more accurately and used duration in a more native-like way than less proficient learners. Implications of our findings for models of L2 speech perception are discussed.

5aSCb27. Cross-language perception of Brazilian Portuguese vowels by Californian English speakers. Polina Vasiliev (Dept. of Spanish and Portuguese, UCLA, 5310 Rolfe Hall, Los Angeles, CA 90095, pvasiliev@ucla.edu) and Paola Escudero (Univ. of Western Sydney, Milperra, NSW 2214, Australia)

The present study investigates the perceptual assimilation and auditory categorization of Brazilian Portuguese vowels by monolingual speakers of Californian English. In the perceptual assimilation task, listeners classified 140 vowel tokens in terms of 10 native English vowel categories by choosing from written English words containing the vowels. The stimuli were isolated vowels extracted from nonce words produced by 10 male and 10 female native speakers of Brazilian Portuguese. In the auditory categorization task, the same listeners identified six Portuguese vowel contrasts which were presented in an XAB format, where X was a subset of the same natural vowel tokens from the previous task and A and B were synthesized prototypes of the Portuguese vowels. The results of the perceptual assimilation task demonstrate that Californian English listeners assimilate two out of six Portuguese vowel contrasts to more than two native vowel categories, resulting in many instances of multiple category assimilation (MCA). In the auditory categorization task, most of the contrasts with the lowest accuracy were those that showed MCA, which demonstrates that this pattern of assimilation may be the cause of Californian English listeners' perceptual difficulties with Portuguese vowels. Predictions for L2 acquisition are made.

5aSCb28. Perceptual confusability of French nasal vowels. Solene Inceoglu (Dept. of Second Lang. Studies, Michigan State Univ., UPLA 101, East Lansing, MI 48823, inceoglu@msu.edu)

The present study investigated native and non-native speakers' perception of the French nasal vowels [ɛ̃-õ-ɑ̃] and the oral vowel [o]. Thirty-four L1 American English intermediate learners of French and thirty-four French native speakers were asked to identify 60 one-syllable word items containing the four vowels [ɛ̃-õ-ɑ̃-o] in word-final position produced by a native speaker and randomly distributed across three conditions: audio-only, audio-visual and visual-only. Identification results revealed that overall performance was better in audio-visual and audio than in visual condition and that non-native speakers showed greater identification for [õ] than for [ɛ̃] and poorer identification for [ɑ̃]. A confusion matrix revealed that across conditions, the vowel [ɑ̃] resulted in the most misidentification and was often mistaken for [ɑ̃]. For both native and non-native speakers [õ] was sometimes confused with its oral counterpart [o]. The data also indicate different patterns of identification of the three nasal vowels in the three conditions. For instance the visual condition was particularly helpful for identifying [ɑ̃]. The obtained results will be discussed in relation to existing second language acquisition research studies.

5aSCb29. Timing of perceptual cues in Scots Gaelic. Natasha Warner, Andrew Carnie, Muriel Fisher, Jessamyn Schertz, Lionel Mathieu, Colin Gorrie, Michael Hammond, and Diana Archangeli (Dept. Ling., U. Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu)

Scots Gaelic, an endangered language, has several typologically unusual sound distinctions. Work on English or other languages cannot predict what perceptual cues Gaelic listeners might use to perceive these distinctions. The current work uses gating experiments, run in Scotland with 16 native listeners (monolingual in Gaelic until at least school age), to investigate timing of perceptual cues to the palatalization, preaspiration, and nasal frication contrasts. Results show that perceptual information about consonant palatalization is located primarily in the consonant itself, with weaker cues in the preceding vowel. For preaspiration, there is no perceptual information in the preceding vowel, but even half of the preaspiration provides a sufficient cue. The claimed nasal fricatives are particularly interesting, as true nasalized fricatives may be aerodynamically impossible, but nasalization could be realized on the preceding vowel, or without frication. The results show that listeners are only marginally able to hear this distinction, and that information does not increase through the signal: The little perceptual information present is already available during the preceding vowel. This confirms aerodynamic results showing some neutralization of this distinction and some nasalization during the preceding vowel. Overall, the results help us determine how information is conveyed using typologically unusual distinctions.

5aSCb30. Examining the role of variability in emotional tone of voice during online spoken word recognition. Maura L. Wilson and Conor T. McLennan (Dept. of Psych., Cleveland State Univ., 2300 Chester Ave., Cleveland, OH 44115, m.l.wilson90@csuohio.edu)

Previous studies demonstrate that listeners are faster to recognize words recently spoken by the same talker, relative to a different talker, when processing is relatively slow. The purpose of the present study was to examine intra-talker variability in emotional tone of voice on listeners' online perception of spoken words. Based on previous work, we predicted reduced priming in the mismatch condition, when words are repeated in a different emotional tone of voice (e.g., sad and frightened), relative to the match condition, when words are repeated in the same emotional tone of voice (e.g., frightened)—when processing is relatively slow (experiment 2, hard lexical decision) and equivalent priming in the match and mismatch conditions when processing is relatively fast (experiment 1, easy lexical decision). The present study should provide a greater understanding of the role that variability in emotional tone of voice plays in listeners' online perception of spoken words.

5aSCb31. Perception of the proximal voice features for emotional speech. Sona Patel, Tanja Banzigar, and Klaus R. Scherer (NCCR Ctr. for Affective Sci., Univ. of Geneva, 7 Rue des Battoirs, CH-1205 Geneva, Switzerland, Sona.Patel@unige.ch)

The expression of emotions results in a number of changes to the acoustic signal. Considering that prosodic events may be encoded into the speech signal using multiple acoustic features, many acoustic variations may be redundant for listeners. An understanding of the acoustic changes that listeners can reliably use to differentiate emotional speech will help in the modeling emotions, for example, from a Brunswikian perspective. In the Brunswikian model of emotion communication, the acoustic features are represented as "distal cues" and the subjective perception of these changes are represented as "proximal percepts." Few attempts have been made to understand the proximal percepts of emotional speech in relation to the acoustic properties they represent. In this study, 19 listeners were asked to rate a set of 160 samples (2 sentences expressed in 8 emotions by 10 speakers) using 12 visual-analog scales (8 prosodic scales and 4 emotion dimensions). Results showed high reliability for all scales except roughness, articulation, and intonation. The emotions were especially differentiated on the dimensions of loudness, sharpness, and speech rate. Regression analysis was used to determine the underlying acoustic cues represented by these scales. The results of this analysis will be discussed.

5aSCb32. Perception of Canadian French word-final vowels by English-dominant and French-dominant bilinguals. Franzo Law II (Dept. of Psych., Univ. of Wisconsin-Madison, Waisman Ctr., 1500 Highland Ave, Madison, WI 53705, flaw@wisc.edu)

Self-identified English-dominant and French-dominant bilinguals from Montreal participated in a modified vowel identification task. Group differences in accuracy and speed for identifying experimental vowels /e, ɛ, o, u, y, ø/ were investigated relative to control vowels /i, a/, expected to be easiest and fastest to identify by both groups. Of interest was the performance on front-rounded /y-ø/ (non-phonemic in English) and /e-ɛ/ (phonologically contrastive in both languages, but // is disallowed word-finally in English). Both groups performed well overall in identifying experimental vowels although the French-dominant group was comparatively more accurate and faster. The English-dominant group was slower than the French-dominant group in identifying /y/ and /e/. Mouse cursor movements captured trial-by-trial revealed that both groups often moved the cursor toward the response button for /u/ before correctly identifying /y/. Results showed that English-dominant participants demonstrated less-automatic perception of most experimental vowels. However, performance speed and mouse patterns of the French-dominant group varied among native vowel categories, implying possible interactions between automaticity and auditory salience. Productions of /e-ɛ/ by participants were analyzed to explore the relative robustness in production of this contrast among participants. Correlations between task performance and measures of French proficiency are explored. [Work supported by NIH F31DC008075.]

5aSCb33. A perceptual study of monosyllabic Mandarin tones produced by three-year-old children growing up in Taiwan. Puisan Wong (Dept. of Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212)

Wong *et al.* (2005) reported that monolingual Mandarin-speaking three-year-old children in the U.S. have not produced adult-like Mandarin lexical tones in monosyllabic words. This study adopted the same methods to examine monosyllabic Mandarin tones produced by three-year-old children in Taiwan. Four hundred and thirty eight monosyllabic tones were collected from ten children and seven mothers in Taiwan. These productions and the 92 productions by the four U.S. mothers reported in Wong *et al.* (2005) were low-pass filtered to retain pitch information, but eliminate lexical information, and were presented to five Mandarin-speaking adults in Taiwan for tone categorization. U.S. mothers' tones were categorized with ceiling accuracies, similar to those judged by U.S. Mandarin-speaking adults reported in Wong *et al.* (2005). Taiwan mothers' tones were judged with lower accuracies, with the accuracy of tone 2 significantly lower than U.S. mothers'. Taiwan children's four tones were judged with 63%, 24%, 27%, and 74% accuracy, all significantly lower than US and Taiwan mothers except for tone 1. Children made more errors in tones 2 and 3 than tones 1 and 4. These findings were consistent with those in Wong *et al.* (2005), suggesting similar developmental patterns for children growing up in the U.S. and Taiwan.

5aSCb34. Context effects on reduced speech perception. Brian C. Ten Eyck (Dept. of Linguist., Univ. of Arizona, P.O. Box 210028, Tucson, AZ 85721, bteneyck@email.arizona.edu)

Acoustic information found in casual, conversational speech is often greatly impoverished and widely variable relative to more formal pronunciation. Previous work has highlighted the role of context in facilitating the comprehension of reduced speech at the individual word level; the experiment reported here measured the impact of varying available context on perceptual accuracy of multi-word speech reductions in American English. Forty-two subjects were exposed twice to sixteen 4–10 word reductions, the second time in one of four counterbalanced contextual conditions: (1) several seconds of audible preceding discourse; (2) an identical written transcript of the preceding discourse; (3) audible discourse of the pose-reduction clause ending; and (4) no additional context. Results were scored from subjects' typed responses based on the number of words correctly identified in each condition. By-subjects and by-items ANOVAs reveal that mean accuracy is correlated with the availability of contextual information. However, pairwise differences in mean accuracy across conditions indicate that discourse context in-and-of itself does not play a significant role in resolving multi-word speech reductions, while speaker-specific acoustic cues do. These results suggest that acoustic cues present in reduced multi-word utterances represent a preferred information source relative to syntactic or discourse context in interpreting massively reduced speech.

5aSCb35. Cross-language perception of Brazilian Portuguese vowels by Californian English listeners. Polina Vasiliev (Dept. of Spanish and Porutugese, Univ. of California, Los Angeles, 5310 Rolfe Hall, Los Angeles, CA 90095, pvasiliev@ucla.edu) and Paola Escudero (Univ. of Western Sydney, Milperra, NSW 2214, Australia)

The present study investigates the perceptual assimilation and auditory categorization of Brazilian Portuguese vowels by monolingual speakers of Californian English. In the perceptual assimilation task, listeners classified 140 vowel tokens in terms of 10 native English vowel categories by choosing from written English words containing the vowels. The stimuli were isolated vowels extracted from nonce words produced by 10 male and 10

female native speakers of Brazilian Portuguese. In the auditory categorization task, the same listeners identified six Portuguese vowel contrasts which were presented in an XAB format, where X were a subset of the same natural vowel tokens from the previous task and A and B were synthesized prototypes of the Portuguese vowels. The results of the perceptual assimilation task demonstrate that Californian English listeners assimilate two out of six Portuguese vowel contrasts to more than two native vowel categories, resulting in many instances of multiple category assimilation (MCA). In the auditory categorization task, most of the contrasts with the lowest accuracy were those that showed MCA, which demonstrates that this pattern of assimilation may be the cause of Californian English listeners' perceptual difficulties with Portuguese vowels. Predictions for L2 acquisition are made.

5aSCb36. Training Korean listeners to perceive phonemic length contrasts in Japanese: Effects of speaking rate variation and contrast type. Mee Sonu (GITI, LASS Lab., Waseda Univ., Sophia Univ., 29-7 Bldg., 1-3-10 Nishi-Waseda, Shinjuku-ku, Tokyo 169-0051, Japan, sonumee@toki.waseda.jp), Keiichi Tajima (Hosei Univ., Tokyo 102-8160, Japan), Hiroaki Kato (NICT, Kyoto 619-0288, Japan), and Yoshinori Sagisaka (Waseda Univ., Shinjuku-ku, Tokyo 169-0051, Japan)

Japanese phonemic length contrasts are difficult to perceive for native-Korean listeners learning Japanese as a second language (L2). Aiming at an effective L2 training method for L2 learners, two experiments were conducted. Experiment 1 evaluated which acoustic cues Korean listeners rely on when categorizing the phonemic length contrast. Experiment 2 examined how differences in speaking rate variation (slow, normal, and fast) and contrast type (vowel contrast vs. consonant contrast) would affect the effectiveness of perceptual training, using a minimal-pair identification task with words embedded in carrier sentences. There were four training conditions comprising combinations of two contrast types (vowel or consonant length) and two speaking-rate variations (single rate or three different rates). Results show that L2 listeners exploit absolute segmental duration to identify phonemic length contrast rather than durational criteria that vary according to speaking rate. Moreover, the trained groups significantly improved in their overall accuracy after training. However, none of the training groups showed significant generalization to untrained contrast types. These results suggest that the effect of training is limited and does not generalize to untrained contrast types even when speaking rate variation is incorporated during training. [Work supported in part by the Grant-in-Aid for Scientific Research (B), JSPS.]

5aSCb37. Lexical and metrical cues for English word segmentation by Mandarin second language learners of English. Yu-Ting Luo and Chien-Tzu Liu (Dept. of Foreign Lang. and Lit., Natl. Chiao Tung Univ., 3/F, Humanities Bldg. 2, 1001 Ta-Hsueh Rd., Hsinchu 300, Taiwan)

This paper investigates the roles of lexical cues and metrical cues in word segmentation by Mandarin L2 learners of English with high and low English proficiency. A cross-modal form priming paradigm is adopted in the experiment. In the experiment, 33 participants (16 advanced learners, 17 beginning learners) heard a five-syllable phrase (e.g., anythingcorri, anything is the and corri is the ′), produced by a native American English speaker. After 100 ms, a three-syllable letter string (e.g., the corridor) was shown on the screen. The participants task was to decide whether a target was a real word. The contexts, primes, and targets have real word and non-word version and have either SW or WS (S: strong; W: weak) stress patterns. The results indicate that: (1) Lexical cues are the main strategy for advanced learners; (2) Beginning learners rely more on metrical cues; (3) When the stress pattern of contexts and primes are identical, it took participants less time to response. The results support previous studies that lexical and metrical are the cues listeners would rely on when doing segmentation. Furthermore, the results suggest that the strategies listeners rely on would change as they become more proficient in English.

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