

Session 2aAA**Architectural Acoustics and Noise: Acoustics of Multifamily Dwellings**

Eric L. Reuter, Cochair

Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

K. Anthony Hoover, Cochair

*McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362***Chair's Introduction—8:00*****Invited Papers*****8:05****2aAA1. Effects of speaker location and orientation on room sound fields and room to room noise reduction.** Noral D. Stewart and W. C. Eaton (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, noral14@sacnc.com)

ASTM E336 requires that loudspeakers be placed at least 5 m from a partition under test, and if that is not possible, then in the corners opposite the partition. The standard further advises that if directional loudspeakers are used in the corners, they should be faced into the corners. In practice, speakers are usually placed in the corners of residential and smaller office spaces, but not in the corners of large spaces. Recent experience has uncovered two effects that need further study. First, when loudspeakers are faced into the corners of a large room, the sound spectrum is strongly colored with a large dip typically in the 100 to 500 Hz range. This effect can be seen in smaller rooms, but is much less pronounced there so it is not normally noticed. Second, the noise reduction measured for a partition of a large room appears to vary depending on whether the speakers are in the corners or not, with lower noise reduction when the speakers are in the corners. This effect does not appear to depend on whether the speakers are faced into the corners or not. Data and any further observations and conclusions developed will be presented.

8:25**2aAA2. Evaluation of methods for isolating portable loudspeakers during sound transmission testing.** Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

When performing field sound transmission tests, the potential exists for structureborne flanking resulting from poor isolation between the loudspeakers and floor. Many of us resort to using furniture and other creative means of isolation with unpredictable results. This paper will present analysis of a handful mocked up isolation systems with the hope of finding a practical, portable solution.

8:45**2aAA3. Change in Canada's national building code—Overview of new requirements and of projects supporting the change.** Christoph Hoeller, Berndt Zeitler, and Jeffrey Mahn (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

The proposed 2015 edition of the National Building Code of Canada sees a major change in sound insulation requirements. Instead of prescribing requirements for the separating assembly only (in terms of STC values), the Code now sets requirements for the sound insulation performance of the complete system (in terms of Apparent Sound Transmission Class (ASTC) values), including flanking sound transmission. The National Research Council Canada is actively supporting the change in the Code by conducting various projects with industry associations from different construction sectors, in order to provide tools, guidance, and the necessary data for compliance. This presentation provides an overview of the new requirements and of the different paths to compliance. Furthermore, various projects conducted at the National Research Council Canada to support the Code change are presented, including tools and guidance to help practitioners. Detailed descriptions of two of the projects are given in two complementary presentations.

9:05**2aAA4. Change in Canada's national building code—Assessing flanking sound transmission in steel-framed constructions.** Christoph Hoeller and Berndt Zeitler (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

The proposed 2015 edition of the National Building Code of Canada sees a major change in sound insulation requirements. Instead of prescribing requirements for the separating assembly only (in terms of STC values), the Code now sets requirements for the sound insulation performance of the complete system (in terms of Apparent Sound Transmission Class (ASTC) values), including flanking

sound transmission. The National Research Council Canada is actively supporting the change in the Code by conducting various projects with industry associations from different construction sectors, in order to provide tools, guidance, and the necessary data for compliance. This presentation focuses on an ongoing joint project between the National Research Council Canada and the Canadian Sheet Steel Building Institute. In the project, the direct and flanking sound transmission in steel-framed assemblies are being investigated. In the presentation, an overview of the project is given, and updates on the current status of the investigation are provided, including measured data concerning the flanking sound transmission in steel-framed constructions.

9:25

2aAA5. Change in Canada's national building code—Assessing flanking sound transmission in concrete-masonry constructions. Berndt Zeitler, Frances King, Jeffrey Mahn, and Christoph Hoeller (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc-cnrc.gc.ca)

The proposed 2015 edition of the National Building Code of Canada sees a major change in sound insulation requirements. Instead of prescribing requirements for the separating assembly only (in terms of STC values), the Code now sets requirements for the sound insulation performance of the complete system (in terms of Apparent Sound Transmission Class (ASTC) values), including flanking sound transmission. The National Research Council Canada is actively supporting the change in the Code by conducting various projects with industry associations from different construction sectors, in order to provide tools, guidance, and the necessary data for compliance. This presentation focuses on a joint project between the National Research Council Canada and the Canadian Concrete Masonry Producers Association. In the project, the direct and flanking sound transmission in concrete masonry and hybrid building systems were investigated. For masonry walls in combination with concrete floors, the ASTC values were calculated according to ISO 15712-1. For masonry walls in combination with wood joist floors, the ASTC values were measured according to ISO 10848. Furthermore, the effect of linings on concrete masonry walls was investigated. This presentation will provide an overview of each of these issues, including results and recommendations.

9:45–10:00 Break

10:00

2aAA6. A new metric for evaluating mid- and high-frequency impact noise. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Impact isolation within multi-family dwellings is currently evaluated using the single number laboratory metric Impact Insulation Class (IIC) and associated field test metrics. There is wide acceptance that IIC does not adequately quantify low frequency impact noise such as thudding from footfalls which is prevalent in lightweight joist-framed construction. However, it is often assumed that mid- and high-frequency impact sources, such as heel clicks, dragging furniture, and dropping objects, are adequately characterized by IIC. Previous research by the authors have indicated that IIC does not adequately distinguish or rank-order between the acoustical performances of resilient matting located in the upper room of a floor ceiling assembly [LoVerde and Dong, *J. Acoust. Soc. Am.* 120, 3206 (2006); LoVerde and Dong, *Proceedings of ICVS14 (2007)*]. Many condominiums have regulations that require a minimum impact sound rating when replacing or installing hard surface finish flooring, and may require field testing to show compliance with the regulations. As expected, a field IIC metric like AIIC, NISR, or ISR is not a suitable descriptor for acoustical performance. A modified metric is defined that more accurately rank-orders the mid- and high-frequency impact noise performance of assemblies and is better suited for these performance requirements.

10:20

2aAA7. Auralization of sound insulation in buildings. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

In various surveys, it was found that people living in multi-family dwellings and apartment houses are annoyed by noise of their neighbors. Also, it seems that building regulations, for example, the German standard DIN 4109 "Sound insulation in buildings," are insufficient. The degree of annoyance is influenced by the personal conditions of the inhabitants (stress), the value of the dwelling and the duration the inhabitants live there. The effects on humans include disturbance of conversation or listening to the TV or radio in private dwellings as well as communication in office premises, reduced power of concentration during physical or mental work, and disturbance of sleep. All this strongly depends on the kind of noise signal (speech, music, footfall, etc.) and on the context, and thus, it is highly doubtful if single-number quantities such as the STL sufficiently describe the real situation. In this paper, a technique is presented for auralization of complex virtual acoustic scenes including airborne and structure-borne noise in buildings with particular respect to sound propagation across or between coupled rooms. Based on SEA-like sound propagation models in standardized prediction methods (EN 12354), algorithms are designed for FIR filtering of audio signals and applied in listening tests and for creation of audio demos. The auralized sounds can be used during building design processes, in studies of human noise perception, and in development of new metrics for future building codes.

10:40

2aAA8. Measuring noise level reduction using an artificial noise source. Rene Robert, Kenneth Cunefare (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Office 002, Atlanta, GA 30332, rrobert6@gatech.edu), Erica Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), and Javier Irizarry (School of Bldg. Construction, Georgia Inst. of Technol., Atlanta, GA)

Residences near airports may be subjected to significant noise levels. The impact of aircraft traffic noise imposes a higher level of design consideration for the outdoor-to-indoor transmission of sound for residences. Noise Level Reduction (NLR) is a common metric used to quantify the ability for a building element to reduce the transmission of external sound pressure levels generated by aircraft. The aircraft noise mitigation measure is determined by the estimate of NLR and the building location in the airport's noise footprint. While

NLR may be measured using an actual traffic source (i.e., aircraft fly-overs), another practice is to perform measurements using a loudspeaker. An investigation is underway to better understand the loudspeaker methods of measuring NLR for buildings. Specifically, the study was tasked with quantifying various factors of these measurements such as angular dependency. NLR measurements were taken on the façade of a “test house” that was constructed for the purpose of the research. Although the “test house” is a single-room structure, the same procedures can be applied to buildings such as single family residences and multifamily dwellings, among others. The results of the analysis should provide a more comprehensive understanding of NLR measurement procedures implemented in sound insulation programs.

11:00

2aAA9. Challenges facing fitness center designers in multifamily buildings. Scott Harvey (Phoenix Noise & Vib., 5216 Chairmans Court, Ste. 107, Frederick, MD 21703, sharvey@phoenixnv.com)

Amenity spaces in multifamily and mixed use developments have become extremely popular and possibly mandatory to the economic success of the project. Of these amenity spaces fitness centers are extremely common and pose significant design challenges to the noise control engineer. This paper will compare several mitigation techniques used to control fitness center noise from today’s prominent sources including treadmills, group exercise, weight machines, free weights, and cross fit weight drops. Mitigation in both wood and concrete structures will be addressed.

11:20

2aAA10. Sound isolation design options for mixed-use buildings. Sean Connolly (Big Sky Acoust., LLC, PO Box 27, Helena, MT 59624, sean@bigskyacoustics.com)

During the design of mixed-use buildings, developers typically have a general idea about the types of commercial tenants that will be in the building. However, those ideas can change after the building has been designed or built, and can range from an office space to retail to a fitness center to a restaurant with live music. This presents a challenge for noise control design to limit noise in residences located above the commercial spaces. Although some noise mitigation measures can be included as part of tenant improvements, the result may be limited by the building base structure decided upon when some commercial uses had not been originally considered. This paper discusses a menu of noise control design options for mixed-use buildings to separate commercial and residential spaces so a developer can make informed decisions about what types of commercial tenants to allow, core and shell constructions, costs of potential tenant improvements, and which options provide the most flexibility.

11:40

2aAA11. Comparisons of impact and airborne sound isolation among cross laminated timber, heavy-timber mill buildings, concrete, and light weight wood frame floor/ceiling assemblies. Matthew V. Golden (Pliteq, 616 4th St., NE, WA, DC 20002, mgolden@pliteq.com)

While laboratory measurements of the Impact Sound Pressure Level (ISPL) and Transmission Loss (TL) of concrete and lightweight wood-frame constructions are well understood, not much laboratory research has been conducted into the acoustical performance of CLT and heavy-timber mill buildings. Recent work on the performance of these wood based floor/ceilings has been presented at other conferences. This paper will review that previously published research for both the bare assemblies and assemblies that include resilient elements. These resilient elements include recycled rubber floor underlayment and sound isolation clip systems. This paper will then compare the performance of these assemblies to each other and to more common concrete and lightweight wood frame floor/ceiling assemblies. The analysis will also include the comparable strengths and weaknesses of each structural system along with the effectiveness of the various acoustical isolation techniques and their effectiveness on each of various floor ceiling assembly. It will be shown that the acoustical isolation techniques will perform differently on the various floor/ceiling assemblies.

Session 2aAB

Animal Bioacoustics: Bioacoustics Across Disciplines: Detecting and Analyzing Sounds

Elizabeth T. Küsel, Chair

Portland State University, Suite 160, 1900 SW 4th Ave., Portland, OR 97201

Contributed Papers

9:00

2aAB1. The effects of aging on detection of ultrasonic calls by adult CBA/CaJ mice. Anastasiya Kobrina and Micheal Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

Mice are frequently used as an animal model for human hearing research, yet their hearing capabilities have not been fully explored. Previous studies (Henry, 2004; Radziwon *et al.*, 2009) have established auditory threshold sensitivities for pure tone stimuli in CBA/CaJ mice using ABR and behavioral methodologies. Yet, little is known about how they perceive their own ultrasonic vocalizations (USVs), and nothing is known about how aging influences this perception. The aim of the present study was to establish auditory threshold sensitivity for several types of USVs, as well as to track these thresholds across the mouse's lifespan. In order to determine how well mice perceive these complex communication stimuli, several CBA/CaJ mice were trained and tested at various ages in a detection task using operant conditioning procedures. Results showed that mice were able to detect USVs well into their lifespan, and that thresholds differed across USV types. Male mice showed higher thresholds for certain USVs later in life than females. In conclusion, the results suggest that mice are sensitive to their complex vocalizations even into old age, highlighting their likely importance for survival and communication.

9:15

2aAB2. Temporary threshold shift not found in ice seals exposed to single airgun impulses. Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, 1, 100 Shaffer Rd., Santa Cruz, CA 95060, coll@ucsc.edu), Brandon L. Southall (Southall Environ. Assoc. (SEA) Inc., Aptos, CA), Asila Ghoul, Andrew Rouse (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA), and Jillian M. Sills (Dept. of Ocean Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

We measured low-frequency (100 Hz) hearing thresholds in trained spotted seals (*Phoca largha*) and ringed seals (*Pusa hispida*) before and immediately after controlled exposures to impulsive noise from a small (10 in³) seismic airgun. Threshold shifts were determined from psychoacoustic data, and behavioral responses to the impulse noise were scored from video recordings. Four incremental exposure conditions were established by manipulating both the distance and the operating pressure of the airgun, with received sound levels ranging from 190 to 207 dB *re* 1 μ Pa peak SPL and 165-181 dB *re* 1 μ Pa²-s SEL. We found no evidence of temporary threshold shift (TTS, \geq 6 dB) in four subjects tested up to eight times each per exposure condition, including at levels previously predicted to cause TTS. Relatively low-magnitude behavioral responses were observed during noise exposure and indicate that individuals can learn to tolerate loud, impulsive sounds, but this does not necessarily imply that similar sounds would not elicit stronger behavioral responses in wild seals. The maximum exposure values used here can improve precautionary estimates for TTS onset from impulse noise in pinnipeds. However, additional studies using multiple impulses and/or higher exposures are needed to identify the actual noise conditions that induce changes in hearing sensitivity.

9:30

2aAB3. Classification of beaked whale and dolphin clicks measured by environmental acoustic recording system buoys in the northern Gulf of Mexico. Natalia A. Sidorovskaia, Kun Li (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA 70504, nas@louisiana.edu), Azmy Ackleh, Tingting Tang (Mathematics, Univ. of Louisiana at Lafayette, Lafayette, LA), Christopher O. Tiemann (R2Sonic LLC, Austin, TX), Juliette W. Ioup, and George E. Ioup (Physics, Univ. of New Orleans, New Orleans, LA)

The Littoral Acoustic Demonstration Center (LADC) has used its Environmental Acoustic Recording System (EARS) buoys to record sperm and beaked whales and dolphins, with frequencies up to 96 kHz, in the northern Gulf of Mexico in 2007 and 2010. The 2007 experiment was the first to record beaked whales in the Gulf. It has been found that there is considerable overlap in the band of beaked whale signals from 20 to 60 kHz with deep-water dolphin clicks, so traditional energy-band detectors have a high occurrence of false positives. Although acoustic measurements in this frequency range validated by visual observations have been limited, for the Gulf of Mexico species, progress is being made in automatically delineating clicks that belong to beaked whale species observed in the Gulf and those originating from dolphins. Spectrograms of the classified clicks are shown and compared to known spectrograms for beaked whale and dolphin species. Many of the spectrograms show an upsweep in the observed spectrum but others do not. Improved classifiers can provide higher accuracy estimates of the regional abundance trends and effects of environmental changes on both beaked whale and dolphin groups. [Research supported by BP/GOMRI, SPAWAR, ONR, NSF, and Greenpeace.]

9:45

2aAB4. Application of density estimation methods to datasets collected from a glider. Elizabeth T. Küsel, Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu), David K. Mellinger, and Sara L. Heimlich (Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Ocean gliders can provide an inexpensive alternative for marine mammal population density studies. Gliders can monitor bigger spatial areas than fixed passive acoustic recorders. It is a low-noise, low-speed platform, easy to set up, maneuver, and transport on land, deploy, and recover. They can be deployed for long periods and report near real-time results through Iridium modem. Furthermore, gliders can sense the environmental conditions of the survey area, which are important for estimating detection distances. The main objective of this work is to evaluate the use of ocean gliders for population density estimation. Current methodologies developed for fixed sensors will be extended to these platforms by employing both simulations and real experimental data. An opportunistic preliminary sea trial conducted in June 2014 allowed for testing of a Slocum glider fitted with an inexpensive acoustic recording system comprising of two hydrophones connected to an off-the-shelf voice recorder installed inside the glider. Acoustic data recorded in deep waters (>1500 m) off the western coast of Sardinia, Mediterranean Sea, showed the presence of sperm whale echolocation clicks. An improved experiment is planned for the summer 2015.

Preliminary results of both campaigns will be presented with an emphasis on population density estimation.

10:00

2aAB5. A technique for characterizing rhythms produced by singing whales. Eduardo Mercado (Dept. of Psychol., Univ. at Buffalo, Buffalo, NY 14260, emiii@buffalo.edu)

Structured sound sequences produced by baleen whales show strong rhythmicity. Such temporal regularity is widely acknowledged, but rarely analyzed. Researchers instead have focused heavily on describing progressive changes in sequential patterns of sounds revealed through spectrographic and aural impressions. Recent production-based analyses of humpback whale sounds suggest that the acoustic qualities of individual sounds can provide useful information about how whales are generating sounds and also may reveal constraints on cyclical production of sounds that help determine rhythmic patterns. Because past analyses have largely ignored the temporal dynamics of sound production, the extent to which whales vary the rhythmicity of sound production over time is essentially unknown. Production-based analyses can be combined with automated measures of temporal patterns of sound production to generate spectrogram-like images that directly reveal rhythmic variability within sound sequences. Rhythm spectrograms can reveal long-term regularities in the temporal dynamics of sound sequences that may provide new insights into how whales produce sequences as well as how they use them.

10:15–10:30 Break

10:30

2aAB6. An approximate model for foliage echoes to study biosonar function in natural environments. Chen Ming, Anupam K. Gupta, and Rolf Müller (Mech. Eng., Virginia Tech, 210 ICTAS II, 1075 Life Science Circle, Blacksburg, VA 24061, cming@vt.edu)

Natural environments are difficult for current engineered sonars but apparently easy for at least some species of echolocating bats. To better understand the information that foliage echoes provide for biosonar-based sensing, an approximate model of foliage echoes has been developed. The model simplifies the scattering problem through two key assumptions: (i) multi-path scattering was neglected and (ii) all leaves were assumed to be circular disks. Due to the latter, the parameters of the model were reduced to the number of the disks, their positions, sizes, and orientations. The exact far-field scatter from a disk (i.e., simplified leaf) in response to planar incident waves can be obtained by summation over an infinite series of spheroidal wave functions. To reduce the calculation time, the scattered field has been approximated by exponential, polynomial, and cosine fitting functions that also depend on the disk parameters. This allows the simulation of echoes from 100 leaves in 20 s on a standard PC. The model was able to reproduce the echo waveforms from dense and sparse foliage qualitatively. The model should thus be well suited for generation of large echo datasets to explore the existence and utilization of statistical invariants in the echoes from natural environments.

10:45

2aAB7. An examination of the biosonar problem associated with sperm whales foraging on jumbo squid in the Gulf of California. Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu), Kelly J. Benoit-Bird (College of Earth, Ocean, and Atmospheric Sci., Oregon State Univ., Corvallis, OR), William F. Gilly (Hopkins Marine Station, Stanford Univ., Pacific Grove, CA), and Bruce Mate (Hatfield Marine Sci. Ctr., Oregon State Univ., Newport, CA)

The backscatter properties of jumbo or Humboldt squid (*Dosidicus gigas*) were examined *in-situ* by projecting simulated sperm whale (*Physeter macrocephalus*) clicks to tethered squids. The incident signal was a broadband click with a peaked frequency of approximately 17 kHz. Echoes were

collected for three different aspect angles; (broadside, anterior, and posterior, and for different body parts for one squid. The beak, eyes, and arms, probably via the sucker rings, played a role in acoustic scattering though their effects were small. An unexpected source of scattering was the cranium of the squid which provided a target strength nearly as high as that of the entire squid though the mechanism remains unclear. The data support the hypothesis that the pen may be an additional important source of squid acoustic scattering. The detection range of these squids for sperm whales was estimated by performing a parametric analysis. Although many of the squid migrate with the mesopelagic layer towards the surface at night, sperm whales have been observed to forage at depth throughout the day and night which would maintain a relative low echo to clutter ratio, increasing their detection range.

11:00

2aAB8. A comparative study of Pinna motions in horseshoe bats and old world leaf-nosed bats. Xiaoyan Yin (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Shanda South Rd. 27, Jinan, Shandong 250100, China, xiaoyanyin4@mail.sdu.cn), Phat Nguyen, Thomas J. Tucker (School of Visual Arts, Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bats (Rhinolophidae) and Old World leaf-nosed bats (Hipposideridae) are two closely related bat families that stand out for the dynamics of their ears during biosonar behaviors. In bats belonging to both families, the outer ears (pinnae) can undergo substantial, fast, and non-rigid shape changes while the animals emit their biosonar pulse trains. So far, characterization of these motions has been limited to either very general measures (e.g., of overall ear rotation) and/or limited data sets covering only a single species. Here, we have combined high-speed stereo vision with digital animation techniques to reconstruct and compare the motions of the pinnae and the head in one rhinolophid (*Rhinolophus ferrumequinum*) and two hipposiderid species (*Hipposideros armiger* and *Hipposideros pratti*). In parallel, we have also recorded the pulses and echoes received by the animals. We found that the pinna motions in both families frequently overlap in time with the arrival of the echoes, so they could have a functional relevance for echo reception. The pinna motions were found to follow similar patterns in all three species and could be decomposed into three main components. Beyond these fundamental similarities, there were also pronounced quantitative differences between the motions seen in the two families.

11:15

2aAB9. Single-sensor density estimation of highly broadband marine mammal calls. Elizabeth T. Küsel, Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu), and David K. Mellinger (Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Odontocete echolocation clicks have been used as a preferred cue for density estimation studies from single-sensor data sets. Such sounds are broadband in nature, with 10-dB bandwidths of 20 to 40 kHz or more. Estimating their detection probability is one of the main requirements of density estimation studies. For single-sensor data, detection probability is estimated using the sonar equation to simulate received signal-to-noise ratio of thousands of click realizations. A major problem with such an approach is that the passive sonar equation is a continuous-wave (CW) analysis tool (single-frequencies). Using CW analysis with a click's center frequency while disregarding its bandwidth has been shown to introduce bias to detection probabilities and hence to population estimates. In this study, the methodology used to estimate detection probabilities is re-evaluated, and the bias in sonar equation density estimates is quantified by using a synthetic data set. A new approach based on the calculation of arrivals and subsequent convolution with a click source function is also presented. Application of the new approach to the synthetic data set showed accurate results. Further complexities of density estimation studies are illustrated with a data set containing highly broadband false killer whale (*Pseudorca crassidens*) clicks. [Work supported by ONR.]

11:30

2aAB10. Baleen whale localization using a dual-line towed hydrophone array during seismic reflection surveys. Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington, 1501 NE Boat St., Seattle, WA 98195, shimah@ldeo.columbia.edu), Maya Tolstoy (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY), and William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA)

Three dimensional seismic reflection surveys use multiple towed hydrophone arrays for imaging the structure beneath the seafloor. Since most of the energy from seismic reflection surveys is low frequency, their impact on Baleen whales may be particularly significant. To better mitigate against this potential impact, safety radii are established based on the criteria

defined by the National Marine Fisheries Service. Marine mammal observers use visual and acoustic techniques to monitor safety radii during each experiment. However, additional acoustic monitoring, in particular locating marine mammals, could demonstrate the effectiveness of the observations, and improve knowledge of animal responses to seismic experiments. In a previous study (Abadi *et al.*, 2014), data from a single towed seismic array was used to locate Baleen whales during a seismic survey. Here, this method is expanded to a pair of towed arrays and the locations are compared with an alternative method. The experimental data utilized in this presentation are from the seismic experiment conducted by the R/V Marcus G. Langseth near Alaska in summer 2011. Results from both the simulation and experiment are shown and data from the marine mammal observers conducted simultaneously with the experiment are used to verify the analysis. [Sponsored by NSF.]

TUESDAY MORNING, 3 NOVEMBER 2015

RIVER TERRACE 2, 8:15 A.M. TO 11:15 A.M.

Session 2aAO

Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics: Passive-Acoustic Inversion Using Sources of Opportunity I

Karim G. Sabra, Cochair

Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Drive, NW, Atlanta, GA 30332-0405

Kathleen E. Wage, Cochair

George Mason University, 4400 University Drive, Fairfax, VA 22030

Chair's Introduction—8:15

Invited Papers

8:20

2aAO1. Passive acoustic remote sensing in a coastal ocean. Oleg A. Godin (Physical Sci. Div., NOAA-Earth System Research Lab, 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov) and Michael G. Brown (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL)

Sound propagation in shallow water over ranges large compared to ocean depth involves multiple reflections from the sea surface and seafloor. With the acoustic propagation environment being much more dynamic than in a deep ocean, noise interferometry faces new challenges but can also provide additional insights into physical processes in a coastal ocean. This paper will review the results on passive acoustic characterization of the seafloor and the water column, including ocean currents, that were obtained using the ambient noise data collected in 2012–2013 in the Straits of Florida. [Work supported by NSF and ONR.]

8:45

2aAO2. Alternative measurements and processing for extracting seabed information from sea-surface noise correlations. Martin Siderius, Joel Paddock, Lanfranco Muzi (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, siderius@pdx.edu), and John Gebbie (Metron Sci. Solutions, Portland, OR)

In recent years, both theoretical and experimental results have shown that the noise generated at the sea-surface from wind and waves contains valuable information about the seabed. A vertical hydrophone array together with beamforming has been a particularly useful configuration for estimating seabed properties. By cross-correlating a vertically upward looking beam with a downward looking beam (the endfire directions), the bathymetry and seabed layering can be determined. However, there may be additional information about the seabed found by cross-correlating beams in directions away from vertical endfire. In this presentation, two new measurement and processing configurations will be considered: noise cross-correlation of beams from a vertical array in directions away from endfire and cross-correlation on a towed horizontal array. For the vertical array, data and modeling show the existence of strong beam correlations coming from a direction consistent with the seabed critical angle. The towed horizontal array configuration, if possible, would provide an alternative to the vertical array for seabed surveying using noise. Measurements from data collected at several sites along with modeling will be used to explain the results from these new measurement and processing configurations.

9:10

2aAO3. Spatial sampling of seabed properties using a glider equipped with a short hydrophone array. Peter L. Nielsen, Jiang Yong-Min (Res. Dept., NATO-STO CMRE, VS Bartolomeo 400, La Spezia 19126, Italy, Peter.Nielsen@cmre.nato.int), Martin Siderius, and Lanfranco Muzi (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR)

Passive acoustic remote sensing of seabed geophysical properties using naturally occurring ambient noise and mobile underwater platforms has the advantage of providing critical environmental parameters over wide areas for sonar performance predictions. However, although technological advances have provided the opportunity to implement acoustic payloads on mobile underwater vehicles the extent and complexity of the acoustic sensors are limited and can therefore only serve as a sensing platform to derive lower resolution seabed properties. During the NATO-STO CMRE sea trial GLISTEN'15, a glider is equipped with an eight-element rigid hydrophone array to estimate the seabed properties based on natural occurring ambient noise. The glider will operate along tracks where the geoacoustic properties and stratification of the seabed are known to vary significantly from historical data. The results from the discrete sampling of the estimated seabed properties are presented and compared to estimates from a short and longer bottom-moored vertical hydrophone array along the tracks. The latest development in synthetic array extension to improve the resolution of the inferred seabed properties are applied and evaluated by comparison of results between the different arrays. The impact of the acquired seabed characteristics on long range acoustic propagation is assessed.

9:25

2aAO4. Improved passive bottom-loss estimation below 10 kHz using arrays deployable on autonomous underwater vehicles. Lanfranco Muzi, Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu), and Peter L. Nielsen (NATO-STO Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

Accurate modeling of acoustic propagation in the ocean waveguide is important for SONAR performance prediction, and requires, among other things, characterizing the reflection properties of the bottom. Recent advances in the technology of autonomous underwater vehicles (AUV) make it possible to envision a survey tool for seabed characterization composed of a short array mounted on an AUV. The bottom power reflection coefficient (and the related reflection loss) can be estimated passively by beamforming the naturally occurring marine ambient-noise acoustic field recorded by a vertical line array of hydrophones. However, the reduced array lengths required by AUV deployment can hinder the process, due to the inherently poor angular resolution. In this paper, data from higher frequencies are used to estimate the noise spatial coherence function at a lower frequency for sensor spacing beyond the physical length of the array. This results in higher angular resolution of the bottom loss estimate, while exploiting the large bandwidth available to current acquisition systems more efficiently than beamforming does. The technique, rigorously justified for a halfspace bottom, proves to be effective also on more complex bottom types, both in simulation and on experimental data.

Invited Papers

9:40

2aAO5. Optimized extraction of coherent arrivals from ambient noise correlations in a rapidly fluctuating medium, with an application to passive acoustic tomography. Katherine F. Woolfe (Naval Res. Lab., 672 Brookline St. SW, Atlanta, Georgia 30310, katherine.woolfe@gmail.com), Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and William A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA)

Ambient noise correlations can be used to estimate Green's functions for passive monitoring purposes. However, this method traditionally relies on sufficient time-averaging of the noise-correlations to extract coherent arrivals (i.e., Green's function estimates), and is thus limited by rapid environmental fluctuations occurring on short time scales while the averaging takes place. For instance, based on extrapolating results from a previous study [Woolfe *et al.*, 2015], passive ocean monitoring across basin scales (i.e., between hydrophones separated by ~ 1000 km) may require at least 10 weeks of averaging time to extract coherent arrivals; but such an averaging time would be too long to capture some aspects of the mesoscale variability of the ocean. To address this limitation, we will demonstrate with simulation and data that the use of a stochastic search algorithm to correct and track these rapid environmental fluctuations can reduce the required averaging time to extract coherent arrivals from noise correlations in a fluctuating medium. The algorithm optimizes the output of an objective function based on a matched filter that uses a known reference waveform to track a set of weak coherent arrivals buried in noise.

10:05–10:25 Break

10:25

2aAO6. Ambient-noise inversion in ocean geoacoustics and seismic-hazard assessment. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca), Sheri Molnar (Univ. of Br. Columbia, Vancouver, BC, Canada), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper presents methodologies and results for the estimation of seabed and soil-column geophysical properties based on Bayesian inversion of ambient ocean-acoustic and seismic noise, respectively. In both marine and terrestrial applications, beamforming is applied to array-based measurements to select the direction of arrival of ambient-noise energy. For the ocean-acoustic application, wave-generated surface noise is recorded at a vertical line array and beamformed to extract up- and down-going energy fluxes from which bottom loss vs. angle can be computed and inverted for geoacoustic profiles. Results from ambient-noise measurements at the Malta Plateau are presented and compared to controlled-source inversions and core measurements. The terrestrial seismic application is aimed at earthquake-hazard site assessment, which requires knowledge of the shear-wave velocity profile over the upper tens of meters of the soil column. In this case, urban seismic noise is recorded on a geophone array and beamformed to determine the dominant

direction of arrival over short time windows, from which the Rayleigh-wave dispersion curve can be estimated and inverted. Two sites with differing geology are considered, and results are compared to invasive (borehole) measurements.

10:50

2aAO7. High-resolution imaging of the San Jacinto fault zone with a dense seismic array and local seismic noise. Philippe Roux, Albanne Lecointre, Ludovic Moreau, Michel Campillo (ISTerre, Univ. of Grenoble, 1381 rue de la Piscine, Grenoble 38041, France, philippe.roux@ujf-grenoble.fr), Yehuda Ben-Zion (Univ. Of South California, Los Angeles, CA), and Frank Vernon (Scripps Inst. of Oceanogr., San Diego, CA)

A highly-dense Nodal array with 1108 vertical (10 Hz) geophone was deployed around the San Jacinto fault zone for ~4 weeks in 2014 in ~600 m x 600 m box configuration (nominal instrument spacing 10–30 m) centered on the Clark branch of the fault zone south of Anza. The array continuously recorded local ambient noise from which cross-correlations between each station pair were extracted for imaging purpose between 1 Hz and 20 Hz. Using subarrays made of 25 sensors, double beamforming was applied to separate body waves from surface waves. Focusing solely on surface waves in a first step, dispersion curves for surface wave group velocities are obtained with unprecedented accuracy at each point of a 10-m spacing grid. The data inversion reveals depth- and lateral-variations of local structural properties within and around the San Jacinto fault zone.

TUESDAY MORNING, 3 NOVEMBER 2015

CLEARWATER, 8:00 A.M. TO 11:30 A.M.

Session 2aBA

Biomedical Acoustics and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications I

Guillaume Haiat, Cochair

Multiscale Modeling and Simulation Laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Pierre Belanger, Cochair

Mechanical Engineering, Ecole de Technologie Supérieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K, Canada

Invited Papers

8:00

2aBA1. Simulation of ultrasound wave propagation in heterogeneous materials. Michael J. Lowe, Anton Van Pamel, and Peter Huthwaite (Mech. Eng., Imperial College London, South Kensington, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

The simulation of ultrasound waves propagating through heterogeneous materials is useful to a range of applications, including Non-destructive Evaluation (NDE) and materials characterization. In NDE, the challenge is to detect defects within the volume of a component against a background of noise of scattering from the inhomogeneities. In materials characterization, the challenge is to measure properties such as stiffness, texture, and spatial distributions of the inhomogeneities. Until recently, simulation of wave propagation through realistic detailed volumetric representations of heterogeneous materials was infeasible because of the huge computational requirements. However, it has recently become possible to do these kinds of simulations. This talk will show Finite Element model simulations of wave propagation in polycrystalline metals that are representative of the high performance alloys used in electricity power plant components. Validations by examining wave speed, attenuation, and backscatter will be discussed, and example deployments of the models will be presented.

8:20

2aBA2. Semi-analytical methods for the simulation of the ultrasonic non destructive testing of complex materials. Sylvain Chatillon, Vincent Dorval, and Nicolas Leymarie (LIST, CEA, Institut CEA LIST CEA Saclay, Bât. Digiteo - 565, Gif sur yvette 91191, France, sylvain.chatillon@cea.fr)

During the last decade, the role of the NDT simulation has been continuously increasing and diversifying in all the industrial sectors concerned by complex inspection methods. The UT simulation tools gathered in the CIVA software developed at CEA LIST include beam and defect echoes computations. The propagation of elastic waves is computed using an asymptotic paraxial ray approximation. In the case of some heterogeneous materials, such as polycrystalline structures and welds, it requires the definition of equivalent propagation properties for which several methods have been developed. In the case of fine-grained polycrystals, the propagation is first

computed for a homogeneous effective medium. Attenuation filters are applied afterwards and structural noise is modeled as the echoes of a random distribution of scatterers. Their parameters can be set empirically or calculated based on microstructural properties. Polycrystals with coarser grains are modeled using Voronoi diagrams. The propagation in the complex structure of an austenitic weld can be modeled by describing it as piecewise homogeneous, though the impedance contrast between neighboring homogeneous domains may cause inaccurate results. As a part of the MOSAICS (ANR) project, the ray tracing method was extended to smoothly inhomogeneous anisotropic media and applied to more realistic descriptions of welds.

8:40

2aBA3. Optimizing the ultrasonic imaging of metallic components with high microstructural noise. Yousif Humeida, Paul D. Wilcox, and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

Ultrasonic arrays are used extensively in a wide range of non-destructive evaluation applications. Many engineering structures are manufactured from polycrystalline metals that result in high levels of microstructural noise making their inspection extremely challenging. In this paper, an optimization framework that uses fast and efficient forward models to simulate the ultrasonic response for both defects and the grains in scattering media is presented. Crucially, these models include both single and multiple scattering effects, and the optimal inspection depends on which type of scattering dominates. For a particular material, simple experimental measurements are used to extract information, such as attenuation and grain scattering coefficients, which is then used to populate models of the microstructural scattering that can be used in an optimization process. As a demonstration, the detectability of small (0.3 mm) defects in copper, a material with high microstructural scatter, is investigated, and the optimal array size, pitch, array location, central frequency, and frequency bandwidth are found. The performance of the optimization system has been evaluated using experimental measurements of receiver operating characteristic (ROC) curves. For the chosen example, the optimal array configuration results in a probability of detection of 90% with a 1% false alarm rate.

9:00

2aBA4. Imaging of a fractal pattern with time and frequency domain topological derivative. Vincent Gibiat, Xavier Jacob (PHASE-UPS, Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31062 cedex 9, France, vincent.gibiat@univ-tlse3.fr), Samuel Rodriguez (I2M, Bordeaux Univ., Talence, France), and Perrine Sahuguet (PHASE-UPS, Toulouse Univ., Toulouse, France)

While fractal boundaries and their ability to describe irregularity have been intensively studied, only a few studies are available on wave propagation in a medium where a fractal or quasi fractal pattern is embedded. Acoustical propagation in 1D or 2D domains can be modeled using Time Domain Finite Differences or in Frequency domain with Finite element methods. The fractal object is then considered as a subwavelength set of scatterers, and the problem becomes a multiple scattering one leading to acoustic localization. So, as some important part of the energy remains trapped inside the fractal pattern, imaging such a medium becomes difficult and imaging such complex media with classical tools as B-scan or comparable methods is not sufficient. The resolution of the inverse problem of wave propagation can then be achieved with the help of the more efficient imaging methods related with Time Reversal. Using the concept of topological derivative as defined in Time Domain Topological Energy and Fast Topological Imaging Method, respectively, in Time domain and Frequency domain is powerful in that case. Examples in 1D and 2D will be presented including the image obtained for a sliced sponge.

9:20

2aBA5. A non-dispersive discontinuous Galerkin method for the simulation of acoustic scattering in complex media. Abderrahmane Bendali (INSA-Toulouse, Institut de Mathématiques de Toulouse UMR 5219 CNRS, Toulouse, France), Hélène Barucq, Julien Diaz (Inria Bordeaux Sud-Ouest, EPC Magique 3D, Université de Pau et des Pays de l'Adour, UMR CNRS 5132., Pau, France), MBarek Fares (Algo Team, Cerfacs, Toulouse, France), Vanessa Mattesi, and Sébastien Tordeux (Inria Bordeaux Sud-Ouest, EPC Magique 3D, Université de Pau et des Pays de l'Adour, UMR CNRS 5132., Département de Mathématiques, Université de Pau, Ave. de l'Université, 64000 Pau, Toulouse 64000, France, sebastien.tordeux@gmail.com)

In the context of time harmonic acoustic wave propagation, the Discontinuous Galerkin Finite Element Method (DG-FEM) and the Boundary Element Method (BEM) are nowadays classical numerical techniques. On one hand, the DG-FEM is really appropriate to deal with highly heterogeneous media. In comparison with continuous Finite Elements Method (FEM), this method is well adapted to direct solver since its connectivity diagram is significantly smaller than the one of classical FEM. However, it suffers of numerical pollution: the numerical wave does not propagate at the correct velocity. This can be very problematic when this method is used on very large computational domain or at high frequency. On the other hand, the BEM is one of the most efficient method to deal with homogeneous media, especially when accelerated by a multipole method or thanks to the Adaptive Cross Approximation. Moreover, this method is really less affected by numerical pollution. However, BEM is not adapted to heterogeneous media. In this talk, we would like to present a DG-FEM whose shape functions are defined thanks to a BEM. This new numerical discretization method benefits from the advantages of BEM and DG-FEM: low pollution effect, ability to deal with highly heterogeneous media. Numerous simulations will show the efficiency and the accuracy of the method on large domains.

9:40

2aBA6. Influence of material heterogeneity on the distortion of a focused ultrasonic beam. Joseph A. Turner and Andrea Arguelles (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

Recent research associated with elastic wave scattering in heterogeneous materials has shown the importance of the Wigner transform in the formulation of the scattering problem to quantify the transducer beam pattern within the sample. The four-fold Wigner distribution function describes the combined time-frequency, space-wave vector domains simultaneously. To date, this approach has been used successfully to examine the diffusely scattered energy for both pulse-echo and pitch-catch measurement configurations. However, the defocusing effect caused by the scattering has received much less attention than the overall scattering. Here, this problem is posed

using a similar approach as the diffuse scattering problem. The energy distribution is shown to be an expansion in terms of the order of the material heterogeneity. Results for the first-order correction from the homogeneous beam pattern are shown with respect to the various parameters of the problem including the frequency, transducer element size, the transducer focus, and the material heterogeneity (here defined in terms of the grain size of a polycrystalline sample). Experimental results of an ultrasonic pulse propagating through a heterogeneous layer will be shown and compared with the model.

10:00–10:15 Break

10:15

2aBA7. Ultrasonic characterization of cohesive and adhesive properties of adhesive bonds. Michel Castaings, Emmanuel Siryabe, Mathieu Renier, Anissa Meziane (I2M, Univ. of Bordeaux, 351 cours Liberation, I2M (A4) - Univ. of Bordeaux, Talence 33400, France, michel.castaings@u-bordeaux.fr), and Jocelyne Galy (IMP, INSA Lyon, Villeurbanne, France)

The increasing use of adhesively bonded joints requires non-destructive evaluation methods to be developed, mostly for safety reasons. An adhesive joint can be divided into two sensitive zones that may cause mechanical failure: the body of the adhesive layer (cohesive zone) and the interphase between that adhesive and one of the substrate (adhesion zone). Weaknesses of these cohesive or adhesive zones can come, for example, from an incomplete curing of the adhesive or from inappropriate, initial treatment of the substrate surface, respectively. The present research attempts to characterize mechanical properties, which are representative of the adhesive and cohesive states of adhesively bonded assemblies, using a through-transmission ultrasonic method. To simplify the approach, the assemblies are made of two aluminum substrates and an epoxy-based adhesive layer. Six samples have been manufactured with various levels of cohesion or adhesion. Inverse problems are then solved to infer the elastic *moduli* of the adhesive or stiffness coefficients, which are modeling the interfacial adhesion. The potential, limits, and outlook of the proposed method are discussed.

Contributed Paper

10:35

2aBA8. A fractional calculus approach to the propagation of waves in an unconsolidated granular medium. Vikash Pandey and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Postboks 1080, Blindern, Oslo 0316, Norway, vikashp@ifi.uio.no)

Our study builds on the work of Buckingham [JASA (2000)] which employed a grain-shearing (GS) model to describe the propagation of elastic waves in saturated, unconsolidated granular materials. He ensemble averages the random stick-slip process which follows the velocity gradient set up by the wave. The stick-slip process is due to the presence of micro-asperities between the contact surfaces of the grains. This is a strain hardening

process which is represented by a time-dependent coefficient in the Maxwell element, besides the coefficient also gives the order of the loss term in the wave equations. We find that the material impulse response derived from the GS model is similar to the power-law memory kernel of fractional calculus. The GS model then gives two equations; a fractional Kelvin-Voigt wave equation for the compressional wave and a fractional diffusion equation for the shear wave. These equations have already been analyzed extensively in the framework of fractional calculus. Since the Kelvin-Voigt model is used in biomechanics of living tissue, we believe the GS theory could offer insights into ultrasound and elastography as well. The overall goal is to understand the role of different material parameters which affect wave propagation.

Invited Papers

10:50

2aBA9. Improving robustness of damage imaging in model-based structural health monitoring of complex structures. Patrice Masson, Nicolas Quaegebeur, Pierre-Claude Ostiguy, and Peyman Y. Moghadam (GAUS, Mech. Eng. Dept., Universite de Sherbrooke, 2500 Blvd. Universite, Sherbrooke, QC J1K 2R1, Canada, Patrice.Masson@USherbrooke.ca)

Model-based Structural Health Monitoring (SHM) approaches offer higher resolution in damage detection and characterization. However, the performance of damage imaging algorithms relies on the quality of the model and the proper knowledge of its parameters, which might prove challenging to obtain for complex structures. In this paper, models are first compared for ultrasonic guided wave generation by bonded piezoceramic (PZT) transducers, from the well-known pin-force model to analytical approaches taking into account the detailed interfacial shear stress under the PZT, and including an electro-mechanical hybrid model. Then, the modeling of guided wave propagation in complex structures is investigated, more specifically in composite structures, considering: (1) dependency of phase velocity and damping on the angle, (2) steering effect due to the anisotropy of the structure, and (3) full transducer dynamics. Validation of the models is conducted on isotropic and composite materials, by comparing amplitude curves and time domain signals with simulation results from Finite Element Models and with experimental measurements using a 3D laser Doppler vibrometer for principal and non-principal directions. Finally, the sensitivity of the damage imaging algorithms to variability in the model parameters is studied, and the benefit of identifying those parameters *in-situ*, prior to damage imaging, is demonstrated.

11:10

2aBA10. Reconstructing complex thickness variations with guided wave tomography. Peter Huthwaite, Michael J. Lowe, and Peter Cawley (Mech. Eng., Imperial College London, South Kensington, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

Wall thickness mapping is very important for quantifying corrosion within the petrochemical industry. One approach is guided wave tomography, where Lamb-type waves, which travel at different speeds depending on the thickness, due to dispersion, are passed through the region of interest. Wave speed is then reconstructed by a tomographic inversion approach and is converted to thickness by the known dispersion relationship. This approach relies on the assumption that guided waves scatter from the varying thicknesses within the plate in the same way that they would from the equivalent varying velocity field. This talk will investigate the accuracy of this assumption for the complex thickness variations associated with corrosion defects found in industry, and discuss potential approaches to mitigate the effects of any errors associated with this.

TUESDAY MORNING, 3 NOVEMBER 2015

ORLANDO, 8:00 A.M. TO 9:40 A.M.

Session 2aEAa

Engineering Acoustics: Vector Sensors: Theory and Applications

Michael V. Scanlon, Chair

RDRL-SES-P, Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

8:00

2aEAa1. Using vector sensors to measure the complex acoustic intensity field. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., Univ. of Washington and Appl. Phys. Lab., Seattle, WA)

The acoustic intensity vector field, defined as the product of pressure particle velocity fields, describes the flow of energy from an acoustic source. Following a brief introduction of acoustic vector sensors, which includes some direct measurements of acoustic intensity, the intensity field is shown to be composed of active and reactive components. Active intensity streamlines depict the time-averaged flow of acoustic energy and reveal characteristics of acoustic propagation including environmental influences. These streamlines do not characterize reactive intensity, which corresponds to the portion of acoustic intensity that time-averages to zero. Reactive intensity is significant in the near-field of a source and in environments where multipath interference occurs. To examine the interplay between active and reactive acoustic intensity, the acoustic field generated by an airborne source positioned well above a water surface is presented. Acoustic measurements of a passing airplane, made simultaneously above and below this sea-surface, are used to demonstrate properties of active and reactive intensity, including how reactive intensity can serve as an indicator of source altitude and range.

8:20

2aEAa2. Acoustic vector sensors: Principles, applications, and practical experience. Miriam Haege (Sensor Data and Information Fusion, Fraunhofer Inst. for Commun., Information Processing and Ergonomics, Fraunhoferstrasse 20, Wachtberg 53343, Germany, miriam.haege@fkie.fraunhofer.de)

Acoustic sensors can be applied in both the civilian and military domain in order to detect risky acoustic sources and localize them. In the case of military use, they are of high importance in sniper and gunshot detection as well as in the classification of firearms. On the civil side, acoustic sensors are employed, e.g., in environmental acoustic monitoring. This paper focuses on the usage of a special class of acoustic sensors, the so called acoustic vector sensors. Such sensors measure the scalar acoustic pressure field as well as the vectorial velocity field at a single location. The construction and the functional principle of an acoustic vector sensor will be described. This type of sensor was applied in a series of field trials, e.g., sniper localization and aircraft detection. The paper presents the experimental results received from the corresponding measurements and discusses the experiences gained with this sensor type.

2a TUE. AM

8:40

2aEAa3. Acoustic particle velocity sensor: Application to infrasonic detection. Latasha Solomon (US Army Res. Lab, 2800 Powder Mill RD, Adelphi, MD 20783, latasha.i.solomon.civ@mail.mil)

Infrasonics is the study of low-frequency acoustics that is below the normal limit of human hearing. Infrasound has the ability to propagate extremely long ranges and is often utilized to monitor naturally occurring phenomena and explosives. Acoustic particle velocity sensors have shown promise in detection and localization of transient signals in the audio range such as small arms fire, mortars, and rocket propelled grenades. Ideally, this sensor can be used to detect various targets spanning a broad range of frequencies to include that of infrasound. The primary objective of this research is to characterize the acoustic vector sensor's localization performance for infrasonic sources given varying atmospheric conditions using algorithms developed at the Army Research Laboratory (ARL).

8:55

2aEAa4. Contributions to the intensity field in shallow water waveguides. Geoffrey R. Moss, Thomas Deal (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841, geoffrey.moss@navy.mil), and Kevin B. Smith (Naval Postgrad. School, Monterey, CA)

In a typical waveguide propagation, the acoustic intensity field is made up from contributions by a direct wave, bottom and surface reflected waves, and a number of interface waves. In shallow water environments, the laterally traveling headwave, generated by interaction with a faster propagating bottom layer, may become important. Several numerical techniques are used to calculate intensity fields in shallow water environments including normal modes, parabolic equation, and finite element methodologies. Bottom interactions are modeled with equivalent fluid properties, and the relative influence of the laterally traveling head wave is examined for several bathymetries of interest. Each codes' solution and merits are compared when calculating both propagating (active) and stationary (reactive) low frequency acoustic intensities.

9:10

2aEAa5. High sensitive MEMS directional sound sensor with comb finger capacitor electronic readout. Daniel Wilmott, Fabio Alves, and Gamani Karunasiri (Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, gkarunas@nps.edu)

The conventional directional sound sensing systems employ an array of spatially separated microphones to achieve directional sensing by

monitoring the arrival times and amplitudes of different microphones. However, there are insects such as *Ormia ochracea* fly that can determine the direction of sound using its miniature-hearing organ much smaller than the wavelength of sound it detects. The fly's eardrums that are coupled mechanically with separation merely by about 1 mm have remarkable sensitivity to the direction of sound. Our MEMS based sensor, which consists of two 1 mm² wings connected in the middle, similar to the fly's hearing system, was designed and fabricated using silicon on insulator (SOI) substrate. The vibration of the wings in response to incident sound at the bending resonance was measured using a laser vibrometer found to be about 1 $\mu\text{m}/\text{Pa}$. For measuring sensor response electronically, comb finger capacitors were integrated on to the wings and the measured output using a MS3110 capacitive to voltage converter was found to be about 25 V/Pa. The fabricated sensors showed $\cos^2\theta$ directional response similar to a pressure gradient microphone. The directional response of the sensor was measured down to about 30 dB.

9:25

2aEAa6. Cantilever-based acoustic velocity sensors. Joseph A. Bucaro (Excet Inc., 4555 Overlook Ave. SW, Naval Res. Lab., WA, DC 20375, joseph.bucaro.ctr@nrl.navy.mil), Nicholas Lagakos (Sotera Defense Solutions, Mclean, VA), Brian H. Houston, and Maxim Zalalutdinov (Naval Res. Lab., WA, DC)

This paper discusses progress made on the design of an acoustic velocity sensor. An analytic model was developed for the frequency response of a slender cantilever rod forced by the pressure gradient and particle velocity associated with an acoustic wave propagating in a fluid. The model, validated with acoustic response measurements in air, was used to design cantilever sensors, which respond predominately to acoustic particle velocity. One such design utilizes a short cantilever formed from a 125 μm silica glass fiber immersed in a viscous fill fluid whose lateral tip displacement is detected using a multi-fiber optical probe. This velocity sensor is predicted to be able to detect fairly low acoustic sound levels in water. Progress has been made in instrumenting a large pool at NRL to allow accurate propagating acoustic wave response measurements in water of these new velocity sensors down to frequencies below 5 Hz. Measurements made in this facility on various cantilever sensors will be presented and discussed. [Work supported by ONR.]

Session 2aEAb

Engineering Acoustics: Analysis of Sound Sources

Kenneth M. Walsh, Chair

K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

10:15

2aEAb1. Numerical and experimental study of acoustic horn. Clebe J. Vitorino (Pontifical Catholic Univ. of Parana – PUCPR, Curitiba, Paraná, Brazil), Nilson Barbieri (Universidade Tecnológica Federal do Paraná – UTFPR, Curitiba, Paraná, Brazil), Key F. Lima (Pontifical Catholic Univ. of Parana – PUCPR, Rua Imaculada Conceição, 1155, Curitiba, Paraná 80215-901, Brazil, key.lima@pucpr.br), and Renato Barbieri (Univ. of the State of Santa Catarina, Joinville, Santa Catarina, Brazil)

Horns are acoustic elements specially designed for maximum transmission of sound pressure and they are used, for example, in sound systems (mainly, external and warning equipment), musical instruments, cleaning apparatus, receivers, and microwave transmitters. The objective of this work is to develop a methodology for obtaining the ideal shape of acoustic horns by comparing numerical data obtained by computational simulation of mathematical model (Finite Element Method—FEM) and experimental data obtained by acoustic tests in laboratory (two microphones method). The main steps to obtain the ideal geometry of acoustic horns are the definition of the objective function, the evaluation of this function and the optimization technique used. The reflection coefficient of the wave is the parameter optimized by the objective function by using the PSO (Particle Swarm Optimization) method. The results obtained for the optimization process were very satisfactory, especially for the correct control of the optimized geometry. The numerical and experimental data showed some differences due to limitations of the numerical model, but the results were good and appear promising.

10:30

2aEAb2. Sound amplification at T-junctions with merging mean flow. Mats Åbom and Lin Du (The Marcus Wallenberg Lab., KTH-The Royal Inst of Technol., Teknikringen 8, Stockholm 10044, Sweden, matsabom@kth.se)

This paper reports a numerical study on the aeroacoustic response of a rectangular T-junction with merging mean flow. The Mach number of the grazing flow in the main duct is fixed at 0.1. The primary motivation of the work is to explain the phenomenon of high level sound amplification, recently seen experimentally, when introducing a small merging bias flow. The acoustic results are found solving the compressible Linearized Navier-Stokes Equations (LNSEs) in the frequency domain, where the base flow is first obtained using RANS with a k - ϵ turbulence model. It is found that the base flow changes significantly with the presence of a small bias flow. Compared to pure grazing flow, a strong shear layer is created in the downstream main duct, starting from the T-junction trailing edge. That is, the main region of vortex-sound interaction is moved away from the junction to a downstream region much larger than the junction width. The flux of fluctuating enthalpy is calculated to estimate the acoustic source power due to the fluid sound interaction.

10:45

2aEAb3. Simulation of surge in the induction system of turbocharged internal combustion engines. Rick Dehner, Ahmet Selamet, and Emel Selamet (Mech. and Aerosp. Eng., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, dehner.10@osu.edu)

A computational methodology has been developed to accurately predict the compression system surge instabilities within the induction system of turbocharged internal combustion engines by employing one-dimensional nonlinear gas dynamics. This capability was first developed for a compression system installed on a turbocharger flow stand, in order to isolate the surge physics from the airborne pulsations of engine. Findings from the turbocharger stand model were then utilized to create a separate model of a twin, parallel turbocharged engine. Extensive development was carried out to accurately characterize the wave dynamics behavior of induction system components in terms of transmission loss and flow losses for the individual compressor inlet and outlet ducts. The engine was instrumented to obtain time-resolved measurements for model validation under stable, full-load conditions and during surge instabilities. Simulation results from the turbocharger stand and engine agree well with the experimental data from their corresponding setups, in terms of both the amplitude and frequency of surge oscillations.

11:00

2aEAb4. Pressure ripple amplification within a hydraulic pressure energy harvester via Helmholtz resonator. Ellen Skow, Kenneth Cune-fare, and Zachary Koontz (Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, eskow3@gatech.edu)

Noise within a hydraulic system is a high-intensity ambient energy source that can be harvested to enable wireless sensor nodes. The noise is typically due to deterministic sources, generally caused by pumps and actuators, and has dominant frequency components around hundreds of Hertz. Hydraulic pressure energy harvesters (HPEH) are centimeter-sized devices that convert the noise into electricity via coupling of the fluid to piezoelectric materials. HPEH devices produce milliwatt level power, which is sufficient for low-energy sensor nodes. A common device used for amplifying or absorbing acoustic energy is a Helmholtz resonator (HR). Incorporation of a HR into an HPEH has been predicted to increase the HPEH power response by up to 7 dB. The properties of hydraulic oil cause HPEH-sized HR to resonate well above the dominant frequencies. Added compliance into the resonator allows the resonance to be tuned closer to the dominant frequency within the hydraulic system. A prototype HPEH with an integral Helmholtz resonator was developed and tested. The results are also compared to an electromechanical model developed for HPEH-HR devices.

Session 2aED**Education in Acoustics and Musical Acoustics: Effective and Engaging Teaching Methods in Acoustics**

David T. Bradley, Cochair

Physics × Astronomy, Vassar College, 124 Raymond Avenue, #745, Poughkeepsie, NY 12604

Preston S. Wilson, Cochair

*Mech. Eng., The University of Texas at Austin, 1 University Station, C2200, Austin, TX 78712***Invited Papers****8:00****2aED1. Finite element illustrations in the classroom.** Uwe J. Hansen (Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu)

“You hear, you forget. You see, you remember. You do, you understand.” This is Tom Rossing’s favorite education quote. Solutions to wave equations generally result in traveling waves. Imposing boundary conditions usually limits these solutions to discrete normal modes. The one dimensional elastic string is an easy, accessible example, illustrated frequently with a long spring. The two dimensional example of a rectangular stiff plate is a little more complex, and the normal modes are often illustrated with Lissajous figures on a plate driven by a shaker. While full blown FEA programs are prohibitively expensive, ANSYS has an educational package available to students at nominal cost with sufficient memory to demonstrate normal mode vibrations in moderately complex structures. Normal mode vibrations in a rectangular plate with a number of different boundary conditions will be illustrated.

8:20**2aED2. Methodology for teaching synthetic aperture sonar theory and applications to undergraduate physics and oceanography majors.** Murray S. Korman and Caitlin P. Mullen (Dept. of Phys., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

Undergraduate senior level physics majors taking Acoustics and oceanography majors taking Underwater Acoustics and Sonar learn about transmitting and receiving arrays (in one unit of their course) and do laboratory experiments to support and enhance the theoretical developments. However, there is a need to expose the students to a detailed unit on synthetic aperture sonar (SAS) while research at USNA is in progress and a teaching laboratory workstation is being developed. This paper communicates the teaching strategy on the topics describing: (a) how a strip-mapped SAS system works, (b) how matched-filtering relates to pulsed compression for a linearly modulated (LFM) pulsed chirp, (c) how synthetic aperture resolution is vastly improved over a conventional acoustic array, (d) how Fourier analysis is used in SAS, and (e) how a data set of N echoes can be used within a back-projection algorithm to obtain a two dimensional reflectivity image of an area (sea floor). Key points are (1) theory with visualizations to convey the teaching material to seniors in a two week period of time, (2) computer simulations, (3) classroom demonstrations in a water tank or in air, and (4) student involvement in a mini-research project using the computers and demonstration apparatus.

8:40**2aED3. Teaching the characterization of performance spaces through in-class measurements at Bates Recital Hall at the University of Texas at Austin.** Michael R. Haberman, Kyle S. Spratt (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlab.utexas.edu), Dan Hemme (BAi, LLC, Austin, TX), and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust., Dept. of Music, Stanford Univ., Stanford, CA)

The characterization of a performance space provides an excellent opportunity to provide students with first-hand experience with many fundamental aspects of room acoustics including reverberation, linear time-invariant systems, measurement methods, and post-processing of real-world data to estimate room metrics. This talk reports recent measurements made in Bates Recital Hall at the University of Texas at Austin (UT) as part of the graduate course on architectural acoustics in the acoustics program at UT. For one class period, students participated in acoustical measurements in the 700-seat venue. The measurements consisted of recording the signal at numerous locations within the room resulting from various on-stage excitations including exponential chirps, interrupted pink noise, and balloon-pops. The audio files captured during the experiments were provided to the students for calculation of the impulse response at the measurement positions and associated room metrics such as reverberation time, bass ratio, clarity index, and initial time-delay gap. Learning outcomes from this approach will be discussed in light of the experiential learning model which emphasizes abstract conceptualization, experimentation and experience, and reflective observation.

9:00

2aED4. Fundamentals of acoustics, vibration and sound, underwater acoustics and electroacoustic transduction at UMass Dartmouth. David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

This paper summarizes the teaching methods and material covered in four introductory graduate classes that are jointly offered as senior technical electives in Electrical Engineering, Mechanical Engineering, and Physics. The possibility of teaching two or three slightly different courses can be done with common lectures by designing different homework, class projects, and examinations for the mechanical engineering and electrical engineering students. Mechanical students are typically well prepared in mechanics and materials while find the use of equivalent electrical circuits more challenging. The reverse is typical for the electrical students. The classes involve many acoustic demonstrations and those are universally appreciated by students with all backgrounds. A number of examples will be presented.

9:20

2aED5. Standards-based assessment and reporting in introductory acoustics courses. Andrew C. Morrison (Joliet Junior College, 1215 Houbolt Rd., Natural Sci. Dept., Joliet, IL 60431, amorrison@jjc.edu)

Standards-based assessment and reporting (SBAR), also referred to as standards-based grading and other names, is a system for tracking student learning in a course and assigning a grade. SBAR replaces traditional grading systems by eschewing points-based assignments and evaluations. Instead, students are assessed over specific learning objectives called standards. SBAR has several advantages including: emphasis on topic mastery, development of intrinsic motivation for learning, and simplifying the grading process. I have implemented SBAR in an introductory acoustics course for music technology, sonography, and general education students. In order to change from a traditional grading system to SBAR several steps were taken. Standards were developed, assessments were written, and course policies were drafted. I will discuss the system as implemented, some of the challenges faced, and suggest ways in which the acoustics education community might provide support for others wanting to implement SBAR in their courses.

9:40

2aED6. Using sound and music to teach waves. Gordon Ramsey (Physics, Loyola Univ. Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspnphys@yahoo.com)

Sound and music are based on the properties of waves. They are also motivating topics for learning many subjects, from music to physics. The most recent "New Generation Science Standards" (NGSS) requires coverage of waves at all K-12 levels. Studies have shown that active student involvement is important in science education for helping the students understand physical concepts. These facts imply that music and acoustics are perfect avenues for teaching the concepts of waves. Even at the college level, non-science majors can understand how music and physics are related through the understanding of wave phenomena. There are many demonstrations, laboratory investigations and hands-on group activities that can be done at all levels. This paper suggests ways to incorporate sound and music to present waves at the levels of middle school, high school, and beginning college.

10:00–10:15 Break

10:15

2aED7. Characterization and design of sound absorber materials. Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., WA, DC 20064, diegoturo@gmail.com), Aldo A. Glean (Saint-Gobain Northboro R&D Ctr., CertainTeed Corp., Northboro, MA), Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, WA, DC), Teresa Ryan (Eng., East Carolina Univ., Greenville, NC), and John A. Judge (Mech. Eng., The Catholic Univ. of America, WA, DC)

Sound absorbing materials are widely used to mitigate noise in indoor environments. Foams and fiberglass are commonly used for passive noise control in the automotive, aerospace industries and for architectural design. The physics of sound absorption in porous materials is not typically included in introductory acoustics courses. However, characterization and design of sound absorbing materials and modeling of their properties can be valuable for students interested in applied acoustics. In this laboratory-oriented course at the Catholic University of America, we cover design of multilayered sound absorbers and experimental procedures for testing such materials. These include an introduction to data acquisition using LabVIEW and post-processing (with Matlab) of recorded sound as well as microphone calibration, and measurement of sound pressure level, and the frequency response function of a speaker. The second part of the course focuses on room acoustics, acoustic properties of materials, impedance tube measurements, modeling of sound propagation in porous media, and design of sound absorbers. We validate the Zwicker and Kosten model using a sample with straight cylindrical pores and apply the Delany-Bazley model to predict acoustic behavior of fibrous materials. Finally, multilayered materials are designed using the impedance translation theorem and tested with impedance tube measurements.

10:35

2aED8. Evolution of experiential learning in an acoustics elective course. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 49504, dludwigs@kettering.edu)

Starting in 2007, PHYS-388, Acoustics in the Human Environment, has been part of the Acoustics Minor at Kettering University. Originally envisioned to capture foundational concepts of acoustics that would be essential to a wide variety of engineering and scientific applications, this course is aimed at a junior/senior level audience to reflect the initiative and maturity required of the student. Topics emphasize the interdisciplinary nature of acoustics in industry, incorporating digital signal processing, psychoacoustics, and applications in room acoustics and environmental noise. Its evolution from a face-to-face studio environment to a hybrid, and then fully online course has retained a hands-on experiential course design and themes of art and design. Challenges arose in teaching the course online, and solutions to promote learner engagement and consistency are described.

2aED9. Physics of music field trips. Juliette W. Ioup and George E. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu)

Many studies on educational techniques have shown that students learn best when they are actively engaged. In introductory physics classes, this means each student makes measurements and performs calculations; in music performance classes, this means each student practices a musical instrument and analyzes music scores. In the two-semester sequence of Physics of Music lectures and laboratories taught at UNO, students participate in a variety of hands-on activities, including measuring quantities for available musical instruments and then calculating various physical parameters from the measured values. Another helpful activity is going on “field trips” to other locations, such as the auditorium in the UNO Performing Arts Center (once to study the concert grand piano there and once to make room acoustic measurements), the small recording studio on campus (lecture given there by the UNO recording engineer demonstrating various types of equipment), and to a chapel across the street from campus (pipe organ demonstrated by the organist of the chapel, acoustic measurements made by students). An extra benefit is that these trips are attractive to both non-science and science students, assisting in both recruitment and retention. Suggestions for and cautions about various field trips from experiences teaching these courses will be presented.

Contributed Papers

11:15

2aED10. Investigation of acoustical spectral analysis to gain a better understanding of tone development in single reed instruments. Charles E. Kinzer (Dept. of Music, Longwood Univ., Farmville, VA 23909, kinzerce@longwood.edu), Stanley A. Cheyne, and Walter C. McDermott (Phys. & Astronomy, Hampden-Sydney College, Hampden-Sydney, VA)

A series of frequency spectra were measured and analyzed, and the results were used for comparison on a series of tone development exercises for saxophone and clarinet players. The musical exercises were focused on manipulations of the embouchure, oral cavity, and vocal tract as a means for altering the overtone content of the tone produced. The acoustical measurements were used to foster greater understanding on the part of the musicians of the importance of the player’s physiology on the production of a musical tone, and help develop the player’s ability to alter the tone in an intentional manner. The results of this study will be presented and discussed.

11:30

2aED11. The science of adhesion. Roger M. Logan (Teledyne, 12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

This presentation is designed to promote the study of science to pre-college audiences. It has been well-received at a variety of pop culture conventions by attendees that vary from high school students interested in building better costumes to NASA Ph.D.’s whose mission is to build better satellites. ASA attendees will be strongly encouraged to use this (or a similar)

presentation as an outreach tool to help recruit the next generation of STEM scholars.

11:45

2aED12. Development of an educational electro-mechanical model of the middle ear. Juliana Saba, Hussnain Ali, Jaewook Lee, John Hansen, Son Ta, Tuan Nguyen, and Cory Chilson (Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, jns109020@utdallas.edu)

In the United States, 1 in 5 people (20%, or 48.1 M), 12 years or older, have hearing loss in one or both ears,^{1, 2} and approximately 3 out of 1000 children are born with some degree hearing loss.^{1, 3} Educating the public on potential causes of hearing loss is often overlooked but imperative; like the remedial task of listening to headphones at a high volume. This in part is due to lack of interactive educational tools to demonstrate sound sensation/perception and apparatuses of the natural safety mechanisms against high intensity sounds. This study aids to encourage particularly a younger generation by increasing public health awareness with the design of a standalone, interactive, and educational electro-mechanical model that exhibits middle ear motion. The model includes: (i) anatomical 3-bone configuration (malleus, incus, and stapes), (ii) cochlea fluid environment, (iii) electrical stimulation of auditory nerve fibers, and (iv) an informational display regarding the natural safety mechanism, sound conduction process, the role of the cochlea in sound sensation, and how cochlear implants/hearing aids assist auditory rehabilitation. Since the model encourages hands-on learning, its placement is desired in either a classroom or a museum striving to reduce any ear-damaging habits and negligence by increasing cognizance.

Session 2aNS**Noise: Damage Risk Criteria for Noise Exposure I**

Richard L. McKinley, Cochair

Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Hilary L. Gallagher, Cochair

*Air Force Research Lab., 2610 Seventh St. Bldg. 441, Wright-Patterson AFB, OH 45433-7901***Chair's Introduction—8:30*****Invited Papers*****8:35**

2aNS1. Evaluation of high-level noise exposures for non-military personnel. William J. Murphy, Edward L. Zechmann, Chucru A. Kardous (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), and Scott E. Brueck (Div. of Surveillance Hazard Evaluations and Field Studies, Hazard Evaluations and Tech. Assistance Branch, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Noise-induced hearing loss is often attributed to exposure to high-level impulsive noise exposures from weapons. However, many workers in industries such as mining, construction, manufacturing, and services are exposed to high-level impulsive noise. For instance, construction workers use pneumatic tools such as framing nailers that can produce impulses at levels of 130 dB peak sound pressure level or greater. Miners have exposures to roof bolters and jack-leg drills that produce more of a continuous, but highly impulsive noise. In some areas of manufacturing the exposures include drop forge processes which can create impacts of more than 140 dB. Finally, in the services sector, law enforcement personnel who maintain proficiency with firearms experience the full gamut of small-caliber firearms during training. This paper will examine the noise exposures from recordings that the NIOSH Hearing Loss Prevention Team have collected. The noises will be evaluated with different damage risk criteria for continuous and impulse noise where appropriate.

8:55

2aNS2. Role of the kurtosis metric in evaluating hearing trauma from complex noise exposures—From animal experiments to human applications. Wei Qiu (Auditory Res. Lab., SUNY Plattsburgh, 101 BRd. St., Plattsburgh, NY 12901, wei.qiu@plattsburgh.edu), Meibian Zhang (Zhejiang Provincial Ctr. for Disease Control and Prevention, Hangzhou, China), and Roger Hamernik (Auditory Res. Lab., SUNY Plattsburgh, Plattsburgh, NY)

A number of animal experiments and epidemiologic studies in humans have demonstrated that current noise standard underestimated hearing trauma by complex noise. While energy and exposure duration are necessary metrics they are not sufficient to evaluate the hearing hazard from complex noise exposure. The temporal distribution of energy is an important factor in evaluating NIHL. Kurtosis incorporates in a single metric all the temporal variables known to affect hearing (i.e., peak, inter-peak interval, and transient duration histogram) that makes kurtosis as one of the candidate metrics. Our previous animal studies show that both kurtosis and energy are necessary to evaluate the hazard posed to hearing by a complex noise exposure. In this study, we focus on how to use the knowledge from animal model into humans. Methods are presented to solve the following questions: (1) How to calculate the kurtosis? (2) What is the relation between kurtosis and the hearing trauma? (3) How to extract a single number from the distribution of kurtosis that would best correlate with noise trauma in real industrial noise environments? Human data study shows that the kurtosis metric may be a reasonable candidate for evaluating the risk of hearing trauma complex noise exposures.

9:15

2aNS3. Assessing acoustic reflexes for impulsive sound. Gregory A. Flamme, Stephen M. Tasko, Kristy K. Deiters (Speech Pathol. and Audiol., Western Michigan Univ., 1903 W. Michigan Ave., MS 5355, Kalamazoo, MI 49008, greg.flamme@wmich.edu), and William A. Ahroon (Auditory Protection and Performance Div., US Army Aeromedical Res. Lab., Fort Rucker, AL)

The acoustic reflex is an involuntary contraction of the middle ear muscles in response to a variety of sensory and behavioral conditions. Middle ear muscle contractions (MEMC) have been invoked in some damage-risk criteria for impulsive noises for over 40 years and one damage-risk criteria proposes that MEMC precede the impulse for a warned listener via response conditioning. However, empirical data describing the prevalence, magnitude, and time-course of reflexive MEMC elicited by impulsive stimuli as well as non-acoustic stimuli and behaviors are scant. Likewise, empirical support for anticipatory MEMC is limited and studies often fail to control for attention or concomitant muscle activity. The current study is a large-scale, multi-experiment project designed to address these limitations in

a laboratory and field environment. MEMC are detected using click train stimuli as probes. Reflexive MEMC are elicited using tones, recorded gunshots, and non-acoustic stimuli (e.g., controlled release of compressed nitrogen gas to the face). Anticipatory MEMC are assessed across varying levels of distraction, beginning with participant instructions to pay attention to the conditioning stimulus and culminating in the assessment of anticipatory MEMC during live-fire exercises with rifles.

9:35

2aNS4. Measurement of high level impulse noise for the use with different damage risk criteria. Karl Buck (ISL (retired), 17 rue de la Résistance, Bartenheim 68870, France, kagebuck@aol.com), Pascal Hamery, Sebastien De Mezzo, and Florian Koenigstein (APC, ISL, Saint-Louis, France)

On the battlefield, but also during training, a soldier is continuously exposed to various types of noise (impulse and continuous). This exposure is not only noise generated by his own weapon but also by weapons or vehicles of close by troops. The exposure levels are between 160 dB peak for small arms and 190 dB peak at the soldier's ear for some anti tank weapons, with A-durations from 0.3 ms (small caliber) to 4 ms for large caliber weapons (e.g., Howitzers). In order to protect the soldier to noise exposures which may induce hearing loss, damage risk criteria (DRC) are implemented, and proposed for the prediction of the potential risk due to a certain noise exposure. Depending on the type of criteria (Pressure-Time-History or A-weighted Energy based), the recording and evaluation of different physical signal parameters has to be done in accordance to the used DRC. The paper will present the problems which may arise when recording impulse (weapon) noise with very high peak pressure levels and discuss measurement techniques compatible with the used DRCs. The paper will also discuss problems which may arise during the use and development of portable noise dose meters for the use in the military environment.

9:55

2aNS5. Damage risk criteria for high-level impulse noise and validation data. Armand L. Dancer (52 chemin Blanc, Authume 39100, France, armand.dancer@orange.fr)

The existing DRCs for high-level impulse noise will be briefly described along with their relative merits. Classical DRCs (CHABA, Mil-STD 1474D Z-curves, Pfander, Smoorenburg...) overestimate the hazard of large weapons, do not assess the actual efficiency of the HPs, and are not compatible with the occupational DRCs. The AHAHAH model is potentially very powerful. However, the model needs to "know" the exact pressure-time history of the impulse at the subject's ear and the human middle ear transfer function for high-level impulses. Unexpected artifacts of measurements send the model on wrong tracks! Last but not least the "parameters" of this model need to be "adjusted" to be in agreement with the experimental results obtained on a large number of soldiers (to be presented). The LAeq8 method with a limit of 85 dB allows a limitation of the hearing hazard comparable to that aimed at by the other DRCs. It allows the assessment of the hazard for all kinds of weapon noises (free field and/or reverberant conditions) and for combined exposures (impulse and continuous noise) either on protected or unprotected ears. Finally, the auditory hazard is evaluated along the same rules in the military and in occupational exposures (ISO 1999).

10:15–10:30 Break

10:30

2aNS6. $L_{IAeq100ms}$, An A-duration adjusted impulsive noise damage risk criterion. Richard L. McKinley (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, richard.mckinley.1@us.af.mil)

Impulsive noise damage risk criteria (DRCs) have been the subject of much debate nationally and internationally for more than 30 years. Several approaches have been used in proposed DRCs including: curves defining exposure based on peak level and A or B duration; auditory hazard units based on the analytical auditory model known as AHAHAH; and L_{Aeq} metrics based on the equal energy concept. Each of the approaches has positive and negative attributes. One of the issues with L_{Aeq} metrics has been the over estimation of hazard with long duration impulses such as those coming from blasts, artillery, large mortars, or shoulder launched missiles. The presentation will describe and discuss the $L_{IAeq100ms}$ impulsive DRC which includes an adjustment based on the A-duration of the impulsive noise and a method of computing protected impulsive noise exposures using impulsive insertion loss data from hearing protectors measured with methods defined in ANSI S12.42.

10:50

2aNS7. Development of the auditory hazard assessment algorithm for humans model for accuracy and power in MIL-STD-1474E's hearing analysis. G. R. Price (Human Res. & Eng. Directorate, Army Res. Lab., PO Box 368, Charlestown, MD 21914, ahanalysis@comcast.net) and Joel T. Kalb (Human Res. & Eng. Directorate, Army Res. Lab., Aberdeen Proving Ground, MD)

MIL-STD-1474E uses the Auditory Hazard Assessment Algorithm for Humans (AHAHAH) model to calculate hazard from intense impulses, its theoretic basis providing greatly increased power and accuracy. It is an electro-acoustic analog paralleling the ear's physiology, including the ear's critical non-linearities and it calculates hazard from basilar membrane displacements. Successfully developed and tested first with an animal model, a parallel version for the human ear was developed and validated with human data. Given a pressure history as input, AHAHAH predicts hazard for the 95%ile susceptible ear. It also makes a movie that allows engineering insight into amelioration of hazard. Hearing protection is accommodated by using input from an acoustic manikin or by implementation of a mathematical protector model using REAT data to calculate input waveforms from free field data. The AHAHAH model is also currently used by the Society of Automotive Engineers for calculation of airbag noise hazard, by the Israeli Defense Forces for impulse noise analysis and is being considered by ANSI's S3 Bioacoustics Committee Working Group 62 (Impulse Noise with Respect to Hearing Hazard) as a basis for an ANSI impulse noise standard.

11:10

2aNS8. A biomechanically based auditory standard against impulse noise. Philemon C. Chan (L-3 ATI, 10180 Barnes Canyon Rd., San Diego, CA 92121, Philemon.Chan@L-3com.com)

This paper addresses two critical issues in developing a damage risk criteria against impulse noise injury, namely, (1) difficulty in setting a threshold for large weapon noise involving hearing protecting devices (HPDs) and small arms noise involving no HPDs; and (2) having a standard procedure to account for the effects of HPDs. The rational way to resolve these issues is to develop biomechanical-based standard using a physics-based model. The Auditory Hazard Assessment Algorithm for the Human (AHAAH) is a biomechanical model that simulates the transmission of sound energy from free field through the ear canal and middle ear to the cochlea. Extensive research has been performed by subjecting the AHAAH to a rigorous verification and validation process. Findings show that the AHAAH middle ear is overly compressive and corrections were made to the annular ligaments parameters. The human data from the historical Albuquerque walk-up study with volunteers wearing HPDs were used to validate the model and develop the dose-response curve for the injury threshold. Calculations were then performed for the German rifle noise tests with volunteers not wearing HPDs, and the predictions show excellent comparison with the injury outcomes, hence providing an independent validation of the revised model.

Contributed Paper

11:30

2aNS9. Nonlinearity in the auditory hazard assessment algorithm for humans. Paul D. Fedele and Joel T. Kalb (Army, DOD, 520 Mulberry Point Rd., Attn: RDRL-HRS-D, Aberdeen Proving Ground, MD 21005-5425, paul.d.fedele.civ@mail.mil)

The Auditory Hazard Assessment Algorithm for Humans (AHAAH) is a software application that evaluates hearing damage risk associated with impulsive noise (<http://www.arl.army.mil/ahaah>). AHAAH applies pressure response dynamics across the external, middle, and inner ear, to biomechanically model the ear's physical response to impulsive sound. Cumulative strain-induced fatigue in the cochlea's organ of Corti determines the

risk of auditory hazard. AHAAH includes nonlinear behavior observed in stapes displacement and associated with the annular ligament in the middle ear. AHAAH's nonlinear behavior has been validated by Price (2007) based on human test results produced by Johnson (1966, 1993, and 1997). Presented analyses results show that the risk of hearing hazard cannot be predicted solely on the basis of waveform energy (A-weighted or not) or waveform peak pressure, because of the middle ear nonlinearity. The risk of hearing hazard does not necessarily behave monotonically with any summary waveform characterization. Although the AHAAH may seem complex, it analyzes response to the full time-dependence of the waveform to accurately analyze hearing damage risk through the nonlinear elements of the human middle ear.

2a TUE. AM

TUESDAY MORNING, 3 NOVEMBER 2015

DAYTONA, 8:35 A.M. TO 12:00 NOON

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Flow-Induced Vibration

Robert M. Koch, Chair

Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Chair's Introduction—8:35

Invited Papers

8:40

2aSA1. Recent Japanese research activities on flow induced vibration and noise. Shigehiko Kaneko (Dept. of Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, kaneko@mech.t.u-tokyo.ac.jp)

In this presentation, recent Japanese research activity on Flow Induced Vibration and Noise mainly done by Kaneko laboratory, Department of Mechanical Engineering, the University of Tokyo, will be presented. Topics cover vortex induced vibration related to Japanese fast breeder reactor Monju thermo-well in line oscillation, sloshing and sloshing damper in connection with the liquid separator designed for Floating Production, Storage and Offloading (FPSO) system, galloping and galloping damper used for cable stayed bridges, combustion oscillation of gas turbine combustor taking account of chemical reaction process, and pipeline acoustics leading to acoustic fatigue. In the end, 30 years of history of data base group activity in Japan Society of Mechanical Engineers (JSME) will be introduced.

9:10

2aSA2. Investigating coupled flow-structure-acoustic interactions of human vocal fold flow-induced vibration. Scott Thomson (Dept. of Mech. Eng., Brigham Young Univ.-Idaho, AUS 106c, Rexburg, ID 83460, thomsons@byui.edu)

Flow-induced vibration of the human vocal folds is a central component of sound production for voiced speech. During vocal fold oscillation, tightly coupled flow, structure, and acoustic dynamics form a system that is rich in multi-physics phenomena, such as large deformation and large strain of exceedingly flexible and multi-layered tissues, repeated collision between vocal folds, coupling between structural modal frequencies and acoustic resonances, and the presence of non-trivial flow features such as the Coanda effect, flow separation, and axis switching. One of the aims of voice production research is to better understand these physical phenomena. In this presentation, tools and techniques for studying vocal fold flow-structure interactions will be discussed. Synthetic vocal fold replicas that exhibit flow-induced oscillations comparable to those of the human vocal folds will be introduced. These replicas are fabricated using three-dimensional prototyping, molding, and casting techniques, in which the multi-tissue layer structure of the human vocal folds is simulated using multiple layers of silicone of differing material properties. Experimental techniques used to characterize replica dynamic responses will be presented. Computational models that include fully coupled fluid, solid, and acoustic domains to simulate vocal fold vibration will be introduced. Several applications of these models and approaches will be discussed.

9:40

2aSA3. Fluid structure interactions with multicell membrane wings. Manuel Arce, Raphael Perez, and Lawrence Ukeiley (Mech. and Aerosp. Eng., Univ. of Florida, MAE-A Rm. 312, PO BOX 116250, Gainesville, FL 32611, ukeiley@ufl.edu)

Flexible wing surfaces can be observed in many natural flyers and their use in small engineered flying vehicles has translated too many beneficial properties. These benefits have manifested themselves in the aerodynamic forces as well as flight stability which are an effect of how the flow and the membranes interact both statically and dynamically. In this work, time dependent particle image velocimetry and digital image correlation are used to study the fluid structure interaction for flow over a membrane wing. The wings examined here are multi-cell silicon rubber membrane wings which have a scalloped free trailing edge with different levels of pretension. The pretension effects the natural frequencies of the membranes and is shown to affect the extension magnitude and membrane motion frequency which are both also affected by aerodynamic loading. Examinations of the membranes through time based and frequency domain analysis motions shows they are highly correlated with the flow. The velocity measurements demonstrated the effects of membranes motion alter the characteristics of the flow over wing leading to changes in the overall aerodynamic properties such as the stall angle and the wake deficit.

10:10

2aSA4. Computational flow noise. Donald Cox, Daniel Perez, and Andrew Guarendi (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, donald.l.cox@navy.mil)

This work focuses on combining the capabilities of computational fluid dynamics with computational structural acoustics to enable the calculation of flow noise primarily for undersea vehicles. The work is limited to the non-coupled problem, where the flow calculations are made over a non-deforming boundary with the goal of calculating wall pressure fluctuations and using them as loads on a finite element structural acoustics model. The ultimate goal of this work is to develop the capability to calculate flow noise for three-dimensional undersea structures for which analytical approaches are not possible. Results will be presented that make use of wall pressure fluctuations calculated using Large Eddy Simulations (LES) and variants of Improved Delayed Detached Eddy Simulations (IDDES).

10:40–11:00 Break

11:00

2aSA5. Using flow-induced vibrations for structural health monitoring of high-speed naval ships. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

It has been demonstrated theoretically and experimentally that an estimate of the impulse (or structural) response between two receivers can be obtained from the long-time average of the cross-correlation of diffuse vibrations (or ambient noise) recorded at these two receivers in various environments and frequency ranges of interest: ultrasonics, underwater acoustics, seismology, and structural health monitoring. Indeed, those estimated impulse responses result from the cumulated contributions over time of random vibrations (e.g., as created by flow-induced vibrations) traveling along the test structure and being recorded by both. Hence, this technique provides a means for structural health monitoring using only the ambient structure-borne noise (e.g., generated by flow-induced vibrations) only, without the use of active sources. We will review work conducted using (1) high-frequency random vibration (100 Hz–5 kHz) data induced by turbulent boundary layer pressure fluctuations and measured on a hydrofoil and a plate at the Navy's William B. Morgan Large Cavitation Channel. (2) Low frequency random vibration data (1 Hz–50 Hz) collected on high-speed naval ships during at-sea operations were strong wave impact loading took place. [Work sponsored by ONR.]

11:30

2aSA6. Wave dispersion in highly deformable, fluid-filled structures: Numerical and experimental study of the role of solid deformation and inertia. Patrick Kurzeja and Katia Bertoldi (John A. Paulson School of Eng. and Appl. Sci., Harvard Univ., 29 Oxford St., Pierce Hall, Rm. 410, Cambridge, MA 02138, patrick.kurzeja@rub.de)

The application and scientific interpretation of wave measurements in fluid-filled structures strongly depend on the frequency regime of interest. This includes, for example, absorption bands, inverse calculation of elastic moduli, and non-destructive crack localization. The wave properties significantly differ between the low-frequency regime (where viscous forces couple fluid and solid) and the high-frequency regime (where inertia forces allow for multiple decoupled wave modes with individual speeds and attenuation). Thus, knowledge of the separative transition frequency is crucial for a reliable prediction, but respective approximations like Biot's characteristic frequency still originate from stiff structures. Soft materials such as biological tissues or synthetic materials are neglected regarding their high deformability. Hence, this presentation demonstrates the change of wave properties from low to high frequencies in soft, fluid-filled structures and highlights the influence of solid deformability and inertia. In particular, it will present: an experimental design to control stiffness and density of a single porous structure by buckling mechanisms with negligible influence on permeability; microscale simulations to identify the underlying wave

modes; and peculiarities occurring in soft fluid-filled structures such as significant dispersion of the P1-wave speed.

11:45

2aSA7. Experimental investigation of the acoustic damping of In-duct orifices with bias flow. Chenzhen Ji and Dan Zhao (Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore 639798, Singapore, cji1@e.ntu.edu.sg)

Geometry of orifice is investigated by the experiments to evaluate the acoustic damping capacity of orifice plates in a duct. Four kinds of plates with complex orifice shapes are fabricated by using modern 3D printing technology. To characterize acoustic damping performance of these plates, sound absorption coefficient is used as an index determined by using the classical two-microphone technique. It is found that the geometric shapes of the perforated orifices can affect their sound absorption performances, and the damping performances of different shaped orifices depend on the frequency range. The length of downstream duct is also proven to determine the damping performance of perforated plates. The shorter the downstream pipe length, the narrower frequency range corresponding to lower power absorption. Moreover, the bias flow is shown to play a critical role on the sound absorption capacity of orifice plate in the experiments. Sound absorption coefficient is found to increase first and then decreased with increased Mach number.

TUESDAY MORNING, 3 NOVEMBER 2015

GRAND BALLROOM 8, 8:30 A.M. TO 10:00 A.M.

Session 2aSCa

Speech Communication: Speech Production Potpourri (Poster Session)

Sarah H. Ferguson, Chair

Communication Sciences and Disorders, University of Utah, 390 South 1530 East, Room 1201, Salt Lake City, UT 84112

Authors will be at their posters from 8:30 a.m. to 10:00 a.m. To allow authors an opportunity to view other posters in their session, all posters will be on display from 8:00 a.m. to 12:00 noon.

Contributed Papers

2aSCa1. Utterance-initial voiced stops in American English: An ultrasound study. Suzy Ahn (Dept. of Linguist., New York Univ., 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu)

In English, phonologically voiced consonants are often phonetically voiceless in utterance-initial position. Other than Westbury (1983), there is little articulatory evidence regarding utterance-initial voicing in American English. The current study uses ultrasound imaging and acoustic measures to examine how tongue position correlates with phonation in American English, comparing phonated voiced stops, unphonated voiced stops, and voiceless stops in utterance-initial position. Eight speakers of American English recorded voiced/voiceless stops at three places of articulation (labial, alveolar, and velar), in three different environments (utterance-initial, post-nasal, and post-fricative), and with two different following vowels (high/low). One adjustment for initiating or maintaining phonation during the closure is enlarging the supraglottal cavity volume primarily via tongue root advancement. In utterance-initial position, there was a clear distinction between voiced stops and voiceless stops in the tongue root for the alveolar and velar places of articulation. Even without acoustic phonation during closure, the

tongue root is advanced for voiced stops in comparison to voiceless stops for supraglottal cavity enlargement. These results suggest that speakers have the same target for both phonated and unphonated stops in utterance-initial position (i.e., shorter VOT), but other articulatory adjustments are responsible for the presence or absence of phonation.

2aSCa2. Aerodynamic factors for place-dependent voice onset time differences. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu), Mohammad Mehdi Alemi (Mech. Eng., Virginia Tech, Blacksburg, VA), and David J. Zajac (Dental Ecology, Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Studies have shown that voice onset time (VOT) tend to increase as place of articulation moves further back in the oral cavity. Different aerodynamic factors have been postulated for place-dependent VOT differences; although no direct aerodynamic measures have been reported in this regard.

The objective of this study was to investigate aerodynamic factors which lead to variation of VOT according to place of articulation. The speech materials of the study were /pa, ta, ka/, each produced 30 times by an adult female (27 yrs) in the carrier phrase “say – again”. SPL was targeted within a ± 3 dB range. Intraoral air pressure (P_o) was obtained using a buccal-sulcus approach. VOT, P_o , and maximum P_o declination rate (MPDR) were measured for each stop. Results showed that: (a) the further back the place of articulation, the longer the VOT; (b) P_o was greatest for the velar stop, intermediate for the alveolar stop, and smallest for the bilabial stop; and (c) the MPDR index showed slower pressure drop for the velar stop compared with the other two stops. Results provide empirical evidence for the role of oral pressure differences, mass of articulators, and cross-section area of the constriction in place-dependent variations of VOT.

2aSCa3. Effects of following onsets on voice onset time in English. Jeff Mielke (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu) and Kuniko Nielsen (Linguist, Oakland Univ., Rochester, MI)

Voice Onset Time (VOT) in English voiceless stops has been shown to be sensitive to place of articulation (Fischer-Jorgensen 1954), to contextual factors such as the height, tenseness, and duration of the following vowel and the voicing of coda consonants (Klatt 1975, Port & Rotunno 1979), to prosodic factors like stress and pitch (Lisker & Abramson 1967), and also to F_0 (McCrea & Morris 2005) and speaking rate (Kessinger & Blumstein 1997, Allen 2003). We report two additional factors involving following consonants. We analyzed 120 /p/- and /k/-initial words produced by 148 Canadian English speakers ($n = 17742$). VOTs of the initial stops were measured semi-automatically and all other segment durations were measured using forced alignment. The results of a mixed-effects regression support earlier findings that VOT is longer in /k/, directly related to following vowel duration, inversely related to speech rate, longer before tense vowels, and shorter before voiceless codas. Additionally, we find that VOT is shorter when the next syllable starts with a phonetically voiceless plosive (i.e., excluding flapped /t/), and that the most relevant measure of vowel duration includes the duration of postvocalic liquids, even those that are typically analyzed as onsets.

2aSCa4. Individual interaction between hearing and speaking due to aging. Mitsunori Mizumachi (Dept. of Elec. Eng. and Electronics, Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, 804-8550, Japan, mizumach@ecs.kyutech.ac.jp)

It is well known that a hearing loss is induced by aging in a high frequency range. It is easy to imagine that the aging also alters characteristics of voice, because you can roughly estimate the speaker's age. In general, those aging phenomena are discussed independently. In speech communication, however, the speech chain [Dense & Pinson, 1993] must dominate the interaction on the aging effects between hearing and speaking. Individual interaction between them is investigated using both his pure-tone audiometry test threshold and his recordings of read utterances. In this study, 21 Japanese elderly males, whose ages ranged from 62 to 85 years old, participated in pure tone audiometry and recording of Japanese sentence and word utterances. Concerning three elderlies with presbycusis, who are aware of hearing loss in daily lives, hearing abilities gradually decrease in proportion to frequency over 2 kHz, and spectral energies increase in the high frequency range over 4.5 kHz. In another case of high-frequency deafness, the spectral energy over 4.5 kHz increases significantly. On the other hand, elderly speakers with normal hearing do not cause energy lift of speech in high frequencies.

2aSCa5. Degree of articulatory constraint predicts locus equation slope for /p,t,s,ʃ/. Sara Perillo, Hye-Young Bang, and Meghan Clayards (Dept. of Linguist., McGill Univ., Montreal, QC, Canada, sara.perillo@mail.mcgill.ca)

The degree of articulatory constraints (DAC) model (Recasens, Pallarès, & Fontdevila, 1997) proposes that consonants involving the movement of

the tongue dorsum are more resistant to coarticulation than those with a more fronted articulation. We assessed this claim using locus equation (LE) slopes as indicators of coarticulation. Participants were asked to produce V1(t).CV2 sequences as part of two-word phrases in a scripted dialog, where C is one of /p, t, s, ʃ/. LE were derived by measuring F_2 at V2 onset and midpoint. Since LE slopes approaching 1 indicate high levels of coarticulation, it was hypothesized that segments with the lowest DAC would have the steepest slopes (/p/>/t/>/s/>/ʃ/) and this is what we found, lending support to the DAC model. A secondary hypothesis assessed the effect of emphatically stressing C on the LE. Participants partook in a dialog involving a “mishearing” of either the target C (Prominent condition) or the preceding V1(t) (Control condition), and they repeated the two word sequence. We expected participants to emphasize the misheard segment and reduce coarticulation if the C was misheard (lower LE slope). Our findings indicate that only the LE slopes of sibilants /s/ and /ʃ/ were reduced under prominence, perhaps due to their high DAC values.

2aSCa6. Effect of practice type on acquisition and retention of speech motor skills. Stephen M. Tasko (Speech Pathol. and Audiol., Western Michigan Univ., 1903 W Michigan, Kalamazoo, MI 49008-5355, stephen.tasko@wmich.edu)

There is a growing literature focused on how speakers acquire and retain speech motor skills. While motor learning experiments provide practical information for improving speech treatment and instructional programs, identifying the specific conditions under which speech skills are enhanced or diminished also offers a window into the underlying organization of the speech motor system. The current study examines how speech motor performance on a challenging speech task varies for different forms of speech practice. Subjects include 40 healthy adult speakers. The challenging speech task is a set of tongue twisters produced at specified speech rates markedly faster than habitual rate. Subjects are assigned to one of three speech practice conditions (imitating an auditory target, listening to an auditory target, or using a magnitude production task) or a control task. Speech motor performance on the challenging speech task is assessed prior to, immediately after, and one day following the speech practice condition. Speech motor performance measures include speech rate accuracy and articulatory accuracy. Improved performance immediately following practice suggests speech skill acquisition while continued improvements during follow up testing suggests speech skill retention. The effect of different practice conditions on speech motor skill acquisition and retention will be described.

2aSCa7. Experimental validation of a three-dimensional finite-amplitude nonlinear continuum model of phonation. Mehrdad Hosnieh Farahani and Zhaoyan Zhang (Head and Neck Surgery, UCLA, UCLA Surg - Head & Neck, BOX 951794, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, mhosnieh@ucla.edu)

Due to the complex nature of the phonation process, simplification assumptions (e.g., reduced flow model, small strain vocal fold deformation) are often made in phonation models. The validity of these assumption is largely unknown because the overall behavior of these phonation models often has not been validated against experiment. In this study, a three-dimensional finite-amplitude nonlinear continuum model of the vocal folds is developed and compared to results from experiments using a self-oscillating physical model of the vocal folds. The simulations are based on a nonlinear finite element analysis, whereby large displacement and material nonlinearity are taken into account. The vocal-fold model is coupled with a reduced-order flow solver based on Bernoulli equation. Preliminary results show that the model is able to qualitatively reproduce experimental observations regarding phonation threshold and typical vocal fold vibration patterns. [Work supported by NIH.]

2aSCa8. A dual task study of the effects of increased cognitive load on speech motor control. Katherine M. Dawson (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., 365 5th Ave., New York, NY 10016, kdawson2@gradcenter.cuny.edu), Grace Bomide (Speech-Language-Hearing Sci., Lehman College, New York, NY), Mark Tiede (Haskins Labs., New Haven, CT), and DH Whalen (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., New York, NY)

Research has shown that behavioral task performance suffers when cognitive load is increased. One method for observing this phenomenon is the so-called dual task paradigm, which has been applied in previous research manipulating the motor, linguistic, and cognitive demands of speech tasks [Dromey & Benson, *JSLHR*, 46(5), 1234–1246 (2003)]. Trade-offs between (and within) domains necessary to maintain task performance probe sensitivity to increased load and the nature of variability in speech (i.e., does speech become more or less variable in increased load situations). In the current study, cognitive load on a speech motor task (mono- and di-syllable repetition) is manipulated using simple, competing memory, visual attention, and inhibition tasks, with concurrent recording of speech acoustics and kinematics. A preliminary analysis of acoustic data from five participants measured duration, amplitude and F0 of utterances during the single and dual task conditions. Results from the dual task conditions suggest a complex trade-off between the amplitude, duration and F0 measures, which differed systematically among the memory, attention, and inhibition tasks. In contrast, the baseline speech measures varied idiosyncratically among participants. Analysis of kinematic data should assist in clarifying how the interactions among these variables are affected by the different types of cognitive load.

2aSCa9. Model based comparison of vocal fold dynamics between children and adults. Michael Döllinger, Denis Dubrovskiy, Eva Beck (Dept. for Phoniatrics and Pediatric Audiol. at the ENT Dept., Univ. Hospital Erlangen, Bohlenplatz, 21, Erlangen, Bavaria 91054, Germany, michael.doellinger@uk-erlangen.de), and Rita Patel (Dept. of Speech and Hearing Sci., College of Arts and Sci., Indiana Univ., Bloomington, IN)

In clinical practice, pediatric vocal fold vibration patterns are visualized by methods and standards derived from the adult population. Quantitative evaluations of vocal fold vibratory changes which are connected to growth and development of children are missing, although it is known that the pediatric larynx is not simply a smaller version of the adult one. The aim of this study was to optimize the oscillations of a biomechanical two-mass-model (2MM) for children and adults and to judge whether dynamic differences exist. High speed recordings (4000 fps) at sustained phonation (vowel /i/) were recorded and analyzed. After glottis segmentation, vocal fold trajectories for 11 children and 23 adults (9 men, 14 women) were investigated. Model parameters were achieved by numerical optimization of the 2MM towards the vocal fold trajectories. Differences in oscillating masses, tissue stiffness, and subglottal pressure were identified and quantified. Children showed increased vocal fold stiffness as well as increased subglottal pressure values. Differences between children vs. men were more distinctive than between children vs. women. In summary, the study gives quantitative evidence of differences between pediatric and adult laryngeal dynamics and confirms the applicability of the 2MM towards children. Next steps will include analyses of disordered pediatric voices.

2aSCa10. Acoustic correlates of velar flutter associated with nasal emission during /s/ in children with velopharyngeal dysfunction. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu), Mohammad Eshghi (Inst. of TeleCommun. Systems, Technische Universität Berlin, Berlin, Germany), and David J. Zajac (Dental Ecology, Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Velar flutter can accompany obligatory nasal air emission in children with cleft palate. It usually occurs as a result of air passing through a

partially closed velopharyngeal port that creates turbulence and tissue vibration due to aerodynamic-elastic forces. Nasal air emission can also occur without flutter. In this case, the velopharyngeal port is relatively large and turbulence is generated at the anterior nasal valve without flutter. In this study we applied auto-correlation to discriminate velar flutter from non-flutter nasal air emission. Three children with nasal turbulence and velar flutter and three children with nasal turbulence without flutter were recorded using the oral and nasal microphones of the Nasometer during production of /si/. Nasal emission of the /s/ sound captured by the nasal microphone was isolated and the auto-correlation functions of the signals were graphed using MATLAB. Results showed that nasal emissions with velar flutter have auto-correlation functions with periodic/quasi-periodic patterns. However, the auto-correlation functions of the non-flutter nasal emissions showed noisy fluctuations without periodic oscillations. Findings revealed that auto-correlation function can be used clinically as an acoustic technique to detect tissue vibration accompanied with nasal air emission.

2aSCa11. Deriving long-distance coarticulation from local constraints. Edward Flemming (Linguist & Philosophy, MIT, 77 Massachusetts Ave., 32-D808, Cambridge, MA 02139, flemming@mit.edu)

Coarticulatory effects can extend over two or more syllables. For example, we find in a study of English nonce words of the form [bV₁C₁əC₂V₂t] that F₂ of V₁ is shifted toward the F₂ of V₂. One approach to such long-distance coarticulatory effects posits direct interactions between the segments involved. For example, coproduction models attribute coarticulatory variation to temporal overlap between segments, so coarticulatory effects of V₂ on V₁ imply that a V₂ gesture begins two syllables earlier, during V₁. An alternative account posits that long-distance coarticulation results from iterative local coarticulation. That is, V₁ can show coarticulatory effects of V₂ because each intervening segment can partially assimilate to the next, resulting in a chain of coarticulatory effects between the two vowels. Since the iterative coarticulation analysis posits that long-distance coarticulation is mediated by intervening segments, it predicts that (i) coarticulatory variation at V₁ due to V₂ should be predictable from variation at the following segment, with no independent effect of later segments, and (ii) if intervening segments resist local coarticulation they should also attenuate non-local coarticulation across them. Neither prediction follows if distant segments can interact directly with each other. Both predictions are confirmed.

2aSCa12. Assessing vowel centralization in dysarthria: A comparison of methods. Annalise Fletcher, Megan McAuliffe (Dept. of Commun. Disord., Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, annalise.fletcher@pg.canterbury.ac.nz), Kaitlin Lansford (School of Commun. Sci. & Disord., Florida State Univ., Tallahassee, FL), and Julie Liss (Dept. of Speech and Hearing Sci., Arizona State Univ., Phoenix, AZ)

Previous literature has consistently reported correlations between acoustic vowel centralization and perceptual measurements of dysarthria. However, the strength of these relationships is highly variable, and many of the techniques used to measure vowel centralization have not been directly compared. This study evaluates methods of assessing vowel centralization and listeners' perceptions of dysarthria—with the aim of strengthening the relationship between these variables. Sixty-one speakers of New Zealand English (NZE; 17 healthy older individuals and 44 speakers diagnosed with dysarthria) read a standard passage. Metrics of vowel centralization were calculated using first and second formants of the [e:], [i:] and [o:] NZE point vowels. The results demonstrate that both the use of a flexible formant extraction point, and changes to the frequency unit in which formants are measured, can strengthen the relationship between acoustic and perceptual measures. Furthermore, applying these formant values to different metrics of vowel centralization, and changing the instructions listeners are given to rate dysarthria, can also reduce levels of unexplained variation in the relationship. In combination, these changes accounted for 18–26% more variance between vowel centralization measurements and listener perceptions of dysarthria in both male and female speakers.

Session 2aSCb

Speech Communication: Analysis and Processing of Speech Signals (Poster Session)

Alexander L. Francis, Chair

Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Authors will be at their posters from 10:30 a.m. to 12:00 noon. To allow authors an opportunity to see other posters in their session, all posters will be on display from 8:00 a.m. to 12:00 noon.

Contributed Papers

2aSCb1. Analysis of distinctive feature matching with random error generation in a lexical access system. Xiang Kong, Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, 50 Vassar St. Rm. 36-523, Cambridge, MA, sshuf@mit.edu)

A matcher for a distinctive feature-based lexical access system is tested using degraded feature inputs. The input speech comprises 16 conversation files from a map task in American English, spoken by 8 female speakers. A sequence of predicted features are produced from a generation algorithm, and the results are randomly degraded at levels from zero to full degradation, for various combinations of the features. Two series of experiments are conducted: the first progressively degrades only single features while leaving all others intact, while the other builds up the system using single, then multiple features. From these experiments, introducing errors into particular articulator-free features, such as vowel, consonant, or sonorant; or articulator-bound features, such as the aspirated feature, pharyngeal features, the nasal feature, the velar feature, or the lateral and rhotic features, do not strongly degrade matching performance. However, matcher performance is more sensitive for errors in the other articulator-free features, and for the articulator-bound features related to vowel place and consonant place, especially, the tongue blade features. For combinations of features, degrading consonantal features, vowel place features, or tongue blade features leads to faster decline in performance, suggesting that these features play more important roles in lexical access.

2aSCb2. Suitability of speaker normalization procedures for classifying vowels produced by speakers with dysarthria. Kaitlin L. Lansford (School of Commun. Sci. and Disord., Florida State Univ., 201 W. Bloxham, Tallahassee, FL 32306, klansford@fsu.edu) and Rene L. Utianski (Dept. of Neurology, Mayo Clinic-Scottsdale, Scottsdale, AZ)

Speaker normalization, a process whereby the perceptual system of a listener recalibrates to accommodate individual speakers, is proposed to account for the ease with which we understand speech produced by multiple speakers with different sized and shaped vocal tracts. A variety of vowel-, formant-, and speaker-intrinsic or extrinsic transforms have been proposed to model speaker normalization of vowels produced by multiple speakers (e.g., Mel, Bark, and Lobanov methods). Suitability of such normalization procedures has been examined extensively in non-disordered speaker populations. Unknown at this point, however, is the appropriateness of normalization procedures for transforming spectrally distorted vowels produced by speakers with dysarthria. Thus, we examined the suitability of two transforms, Bark and Lobanov, for normalizing vowels produced by a heterogeneous cohort of 45 speakers with dysarthria. Non-normalized (Hertz) and Bark transformed vowel tokens were classified via discriminant function analysis (DFA) with 55% and 56% accuracy, respectively. Classification accuracy of vowel tokens normalized using Lobanov's method was 65%. The results of the DFAs were compared to perceptual data, which revealed listeners identified vowel tokens with 71% accuracy. These results suggest vowel-extrinsic and formant- and speaker-intrinsic normalization methods

(e.g., Lobanov) are better suited to model speaker normalization of dysarthric vowels.

2aSCb3. Combining gestures and vocalizations to imitate sounds. Hugo Scurto, Guillaume Lemaitre, Jules Françoise, Frédéric Voisin, Frédéric Bevilacqua, and Patrick Susini (IRCAM, 1 Pl. Stravinsky, Paris 75004, France, hugo.scurto@ircam.fr)

Communicating about sounds is a difficult task without a technical language, and naïve speakers often rely on different kinds of non-linguistic vocalizations and body gestures (Lemaitre *et al.* 2014). Previous work has independently studied how effectively people describe sounds with gestures or vocalizations (Caramiaux, 2014, Lemaitre and Rocchesso, 2014). However, speech communication studies suggest a more intimate link between the two processes (Kendon, 2004). Our study thus focused on the combination of manual gestures and non-speech vocalizations in the communication of sounds. We first collected a large database of vocal and gestural imitations of a variety of sounds (audio, video, and motion sensor data). Qualitative analysis of gestural strategies resulted in three hypotheses: (1) voice is more effective than gesture for communicating rhythmic information, (2) textural aspects are communicated with shaky gestures, and (3) concurrent streams of sound events can be split between gestures and voice. These hypotheses were validated in a second experiment in which 20 participants imitated 25 specifically synthesized sounds: rhythmic noise bursts, granular textures, and layered streams. Statistical analyses compared acoustics features of synthesized sounds, vocal features, and a set of novel gestural features based on a wavelet representation of the acceleration data.

2aSCb4. Direct measurement of the dynamic range for rectangular speech passbands, from threshold to rollover. James A. Bashford and Richard Warren (Psych., Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, bashford@uwm.edu)

Measurement of passband intelligibility can be confounded by appreciable contributions from transition bands under filtering conditions conventionally considered steep. Eliminating appreciable contributions outside of speech passbands can require slopes of several thousand dB/octave [Warren *et al.*, JASA. 115, 1292-1295]. By employing effectively rectangular passbands, it is possible to determine their intrinsic intelligibilities, and also their dynamic ranges as determined by their threshold amplitudes and their decrease in intelligibility at high levels ("rollover"). Uncontaminated measures of these limits were obtained in the present study using 1-octave passbands (Experiment 1) and 1/3-octave passbands (Experiment 2) using rectangular speech bands (4800 dB/octave slopes) that spanned the frequency range from 0.25 to 8.0 kHz. Results obtained for the bands presented singly and in pairs, at levels ranging from threshold to 80 dB SL, indicate that [1] the speech dynamic range substantially exceeds 30 dB across most of the spectrum, and that [2] intelligibility rollover occurs at relatively low levels, exceeding approximately 70 dB. The use of rectangular speech bands for clinical assessment will be discussed. [Research supported by NIH.]

2aSCb5. Maintaining speech intelligibility at 100 dB using arrays of subcritical width rectangular bands. Richard Warren and Peter Lenz (Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, rmwarren@uwm.edu)

The Speech Intelligibility Index employs 16 contiguous 1/3-octave bands that sample the importance of frequencies across the speech spectrum. The present study employed the same Center Frequencies (CFs) using “Everyday Speech” sentences, but reduced the original 1/3-octaves having 26% bandwidths to 4% effectively rectangular bands (4800 dB/octave slopes). The resulting array of 16 subcritical-width bands had an intelligibility of 96% when heard at 60 dB despite having less than 16% of the 1/3-octave bandwidths. But, increasing the amplitude to 100 dB produced a decrease in intelligibility (“rollover”) to 86%. In a parallel experiment, when the sixteen bands had a bandwidth of 40 Hz for each of their CFs, the intelligibility was 95% at 60 dB and decreased to 91% at 100 dB. But when a “chimera” or hybrid was created with a width of 40 Hz for all CFs from 0.25 kHz to 1 kHz, and a width of 4% for CFs from 1 kHz (bandwidth of 40 Hz) to 8 kHz (bandwidth of 320 Hz), then intelligibility was 99% at 60 dB, and 97% at 100 dB. Hybrids of this type may be of use in hearing aid design. [Research supported by NIH.]

2aSCb6. Arrays of subcritical width rectangular speech bands with interpolated noise maintain intelligibility at high intensities. Peter Lenz and James A. Bashford (Psychology, Univ. of Wisconsin - Milwaukee, PO Box 413, Milwaukee, WI 53201, plenz@uwm.edu)

Speech intelligibility declines at high intensities for both normally hearing and hearing-impaired listeners. However, this rollover can be minimized by reducing speech in high frequency regions to an array of noncontiguous bands having vertical filter slopes (i.e., rectangular bands) and widths substantially narrower than a critical band. Normally hearing listeners were

presented with “Predictability Low” sentences consisting of a 500-Hz lowpass pedestal band and an array of ten 4% bands spaced at approximately 1/3-octave (alternate ERBn) intervals from 1000 Hz to 8417 Hz. The pedestal band was fixed at 70 dB and the subcritical-band array varied from 55 to 100 dB in peak level. Intelligibility did not vary significantly for levels from 65 to 95 dB, ranging from 86 to 89%. Array intelligibility did significantly decrease to 82% when speech level was increased to 100 dB. However, intelligibility was restored to 88% when lower level rectangular noise bands (−30 dB relative spectrum level) were interpolated between speech bands. It is suggested that subcritical width filtering reduces rollover by limiting firing rate saturation to a subset of fibers within critical bands, and that interpolated noise further reduces saturation via lateral inhibition. Hearing aid applications will be discussed. [Research supported by NIH.]

2aSCb7. Effect of depression on syllabic rate of speech. Saurabh Sahu and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland College Park, 8125 48 Ave., Apt. 101, College Park, MD 20740, ssahu89@umd.edu)

In this paper, we are comparing different methods to measure syllable rate of speech. Basically our method counts the number of vowels and divides it by the duration of speech. We use the energy content in 640–2800 Hz and 2000–3000 Hz to eliminate nasals and glides. Energy content in 0–400 Hz as well as pitch information and helps eliminate the unvoiced fricatives. We compare our method with Jong *et al.* (Behavior research methods. 2009; 41 (2): 385–390.) who wrote a Pratt script and with another method that estimates the syllable rate from peak modulation rate of speech. We have seen that the latter measure tracks the changes in HAMD scores and therefore seems sensitive enough to measure changes in the degree of depression. We will determine if these other methods will show the same sensitivity

TUESDAY MORNING, 3 NOVEMBER 2015

CITY TERRACE 7, 9:00 A.M. TO 11:15 A.M.

Session 2aSP

Signal Processing in Acoustics: Detection, Feature Recognition, and Communication

Geoffrey F. Edelman, Chair

U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375

Contributed Papers

9:00

2aSP1. Correlation trends in Naval Surface Warfare Center Panama City Division’s database of simulated and collected target scattering responses focused on automated target recognition. David E. Malphurs, Raymond Lim, Kwang Lee, and Gary S. Sammelmann (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, david.malphurs@navy.mil)

In recent years, NSWC PCD has assembled a database of sonar scattering responses encompassing a variety of objects including UXO, cylindrical shapes, and other clutter-type objects deployed on underwater sand and mud sediments and inspected over a large range of aspect angles and frequencies.

Data available on these objects consist of a simulated component generated with 3D finite element calculations coupled to a fast Helmholtz-equation-based propagation scheme, a well-controlled experimental component collected in NSWC PCD’s pond facilities, and a component of measurements in realistic underwater environments off Panama City, FL (TREX13 and BayEX14). The goal is to use the database to test schemes for automating reliable separation of these objects into desired classes. Here, we report trends observed in an on-going correlation analysis of the database projected onto the target aspect vs frequency plane to clarify the roles of the environment, the data collection process, and target characteristics in identifying suitable phenomena useful for classification. [Work supported by ONR and SERDP.]

2aSP2. Doppler discrimination of a constant velocity scatterer at depth in shallow water. Christopher Camara, David Anchieta, Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), and Praswath Mahrajan (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, danchieta@umassd.edu)

Doppler as a discriminant for a shallow water moving rigid scatterer from a single element receiver is considered here. A mono-static source-receiver configuration emits a single tone to ensonify a moving object. Inference regarding the depth and speed of the moving object are sought from the amplitude and Doppler of the direct arrival and the surface interacting arrival. Computation of the full posterior probability distribution of the returned amplitudes and frequencies given the received waveform and the prior distribution on target depth is made by Markov Chain Monte Carlo sampling. A Gibbs sampler is employed to construct the posterior joint density of all parameters. Conditional and marginal densities of the amplitudes are analytically tractable while those of the frequencies are made with an importance sampling approach. Confidence intervals are computed and employed to address depth discrimination.

9:30

2aSP3. High-frequency, vertically directional short-range underwater acoustic communications. Geoffrey F. Edelmann, Lloyd Emokpae, and Simon E. Freeman (U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, WA, DC 20375, edelmann@nrl.navy.mil)

The underwater acoustic channel is a challenging environment for achieving high data rate communications due to multipath, attenuation, noise, and propagation delay. Performing and maintaining adaptive channel equalization requires significant computational overhead, leading to costly and power-hungry devices. Here we describe a low-cost reconfigurable acoustic modem platform (RAMP) intended to facilitate a cable-less benthic hydrophone array made from inexpensive and replaceable nodes. The high data rate acoustic modem is modulated on a carrier frequency of 750 kHz via binary phase shift keying (BPSK). Each modem is spaced approximately 10 m from adjacent units. Due to the vertical directivity of the transducer the half-maximum envelope of the main lobe is approximately 3°, thereby mitigating multipath from the bottom. Data at rates of up to 125 kbps will be shown from at-sea experimental measurements made in Panama City Beach, FL. [This work was supported by the Office of Naval Research.]

9:45

2aSP4. Prediction of localization error in generating a focused source. Min-Ho Song (Musicology, Univ. of Oslo, Institutt for musikkvitenskap ZEB-bygningen 2. etg Sem Sælands vei 2, Oslo 0371, Norway, minho.song@imv.uio.no), Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea), and Yang-Hann Kim (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

This paper proposes a method of predicting human localization error for a focused source. A focused source is a virtual source located in between of a loudspeaker array and a listener. However, generation of the focused source cannot avoid the artifact due to causality, listeners always perceive pre-echoes before the desired sound. Since the human hearing system is sensitive to a preceding waves, it can lead a listener to perceive a virtual source in undesired direction. Because the repeating pre-echoes are observed for ineligibly long interval (~100 ms), it is not clear to distinguish timbral distortions and echoes from the localization error due to the summing localization of the human auditory system. Therefore, a suppression condition was defined from the precedence effect to separate the localization error from timbral distortions. After applying the suppression condition, the energy vector model was used to quantify the localization error. Combining the suppression condition and the energy vector model, localization error in horizontal plane for each listening spot considering positions of focusing point, array shape, driving solutions, spatial sampling, and truncation can be predicted. The examples show that the prediction method clearly holds up with focused source observations reported from relevant literature.

10:00–10:15 Break

2aSP5. Using automatic speech recognition to identify dementia in early stages. Roozbeh Sadeghian (Electrical and Comput. Eng. Dept., State Univ. of New York at Binghamton, 4400 Vestal Parkway East, Binghamton, NY 13902, rsadegh1@binghamton.edu), David J. Schaffer (ORC Inst. for Inter-generational Studies, State Univ. of New York at Binghamton, Binghamton, NY), and Stephen A. Zahorian (Electrical and Comput. Eng. Dept., State Univ. of New York at Binghamton, Binghamton, NY)

Early non-invasive diagnosis of Alzheimer's disease (AD) and other forms of dementia is a challenging task. Early detection of the symptoms of the disorder could help families and medical professionals prepare for the difficulties ahead, as well as possibly provide a recruitment tool for clinical trials. One possible approach to a non-invasive diagnosis is based on analysis of speech patterns. Subjects are asked to describe a picture and their description (typically 1 to 3 minute speech sample) is recorded. For this study, a database of 70 people were recorded, 24 with a clinical diagnosis of probable or possible Alzheimer's disease. When these data were combined with 140 other recorded samples, a classifier built with manually transcribed versions of the speech was found to be quite accurate for determining whether or not a speech sample was obtained from an Alzheimer's patient. A classifier built using automatically determined prosodic features (pitch and energy contours) was also reasonably accurate, with several subsets of pitch and energy features especially effective for classification, as assessed by cross validation. The manually transcribed text has now been replaced by automatically transcribed text using automatic speech recognition (ASR) technology. The main objective of this paper is to report on the relative effectiveness of several ASR approaches, including public domain ones, for this task.

10:30

2aSP6. Context recognition on smartphones by categorizing extracted features from acoustic and other sensor data. Miho Tateishi (School of Sci. for Open and Environment Systems, Keio Univ. Graduate School of Sci. and Technol., Hiyoshi 3-14-1, Kohoku-ku, Yokohama 223-8522, Japan, m.tateishi-3303@keio.jp)

This work presents a method of context recognition on smartphones using several built-in sensors. This system is developed for the Android. The three sensors are a microphone, an accelerometer, and a light sensor, which are correlated with human senses or movement. Context recognition is a method of being aware of the user's context or contiguous environment. In existing systems, raw time series data are directly used for the recognition. Our system aims to define data categories, which are connected with contexts, by extracting features from time series data. These features are made up of processed signals from each sensor. In particular, acoustic data turns into several different features; volume, spectrum average, peak spectrum appearance ratio, and correlation of these signals over a defined period of time. With acceleration and light features, it can classify similar contexts in the same category. For example, riding the train appears in one discreet category. Another category exemplifies the situation in a library, PC room, or laboratory which can be described as the "quiet" work place. As a part of automatic context awareness, features are formed hierarchical structure in order to make this method efficient.

10:45

2aSP7. The rear end collision and the wheel flying off protection using the reinforcement learning with the adaptive sound caution. Kazuhide Okada (Tele-Commun. Dept., College of Micronesia, P.O. Box 614, Daini, Kolonia, Kolonia, Pohnpei FM96941, Micronesia, rainbow_vc@yahoo.co.jp)

Safety driving is to watch both the front and the side from the driver seat. In order to avoid a rear end collision for the other car and a wheel flying off its axle for the said car itself at the same time, Q-learning of reinforcement learning was used in this study. In this study, the image of the front whole window is always captured by camera. Here, a steering angle divided by eight categories toward the front is the action a , of a wheel flying off protection and car speed divided by ten categories is that of the rear end collision protection. The status s_t is distance from the sideview mirror to the side ditch for the wheel flying and the length between the roof of the car which is running ahead and its rear bumper. In the training phase, so as to maximize the sum of the rewards taken from the environment as the vehicle's front window, the

system updates the recursion Quantity of a state action combination $Q(s_t, a_t) = Q(s_t, a_t) + \alpha (r_{t+1} + \gamma \max_{[a]} Q(s_{t+1}, a) - Q(s_t, a_t))$ (t:time, α :learning rate, γ : discount) periodically until its $Q(s_t, a_t)$ value is converged individually. When rewards both for the rear end collision and wheel flying off become mature, the automatic operating routines are regarded to be completed. And when making use of this system, the code still watches the car distance from the ditch and the relative level of the front car and the adaptive caution sound which has three spectrum peaks along 0.5 through 6 kHz are rung, adjusting frequency, loudness, and duty corresponding to the severity if the car approaches the crisis.

11:00

2aSP8. Stable QAM development with less BER on convergence time-compression type Q-learning for mass audio signal transmission. Kazuhide Okada (Tele-Commun. Dept., College of Micronesia, P.O. Box 614, Daini, Kolonia, Kolonia, Pohnpei FM96941, Micronesia, rainbow_vc@yahoo.co.jp)

This paper presents method which protects the temporary hang-up on the communication line and sustains the demodulation of the clear audible

signal at the receiver, when the mass sound data is sent from the transmitter. QAM is one of the digital modulation technology, mapping the modulating signal not only toward the phase but also toward the amplitude on Q-I constellation, derived from QPSK. This modulation can pack larger data in the fixed period than QPSK. But once the communication path is exposed by stuff jitter or random jitter, the quantization error on its coordinate occurs, which means coordinate axes often subtly rotate with returning to the original position. In order to minimize such quantization errors on demodulation, Q-learning as one of Reinforcement Learning was used in this study. In the design of the feedback system comprehending the agent and the environment, the angle of the reverse vibration of the upper each axis for the quick restoring to the normal quantization becomes an action a_t . And the reward r_t is the relative baud rate, while the status s_t is BER, as I/F between Agent and the environment. The Quantity of a state action combination $Q(s_t, a_t)$ was updated as the index which measures the value of actions, with the Quantity computation steps decreased by TTD (Truncated Temporal Difference) and Log-time overlooking at $Q(s_t, a_t)$ in the training process of the experiment. The degree of control on coordinate's axes vibrations triggered by injected jitter was evaluated by the visible decrease in BER.

TUESDAY MORNING, 3 NOVEMBER 2015

GRAND BALLROOM FOYER, 9:00 A.M. TO 5:00 P.M.

Exhibit and Exhibit Opening Reception

The instrument and equipment exhibit is located near the registration area in the Grand Ballroom Foyer.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

The Exhibit will open on Monday with an evening reception with lite snacks and a complimentary drink.

Exhibit hours are Monday, 2 November, 5:30 p.m. to 7:00 p.m., Tuesday, 3 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 4 November, 9:00 a.m. to 12:00 noon.

Coffee breaks on Tuesday and Wednesday mornings (9:45 a.m. to 10:30 a.m.) will be held in the exhibit area as well as an afternoon break on Tuesday (2:45 p.m. to 3:30 p.m.).

The following companies have registered to participate in the exhibit at the time of this publication:

Brüel & Kjær Sound & Vibration Measurement—www.bksv.com

Freudenberg Performance Materials—www.Freudenberg-pm.com

G.R.A.S Sound & Vibration—www.gras.us

PCB Piezotronics—www.pcb.com/

Sensidyne—www.sensidyne.com

Springer—www.Springer.com

Teledyne Reson—www.teledyne-reson.com

Session 2pAAa**Architectural Acoustics and Musical Acoustics: Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio I**

Timothy W. Leishman, Chair

*Physics and Astronomy, Brigham Young University, N247 ESC, Provo, UT 84602***Chair's Introduction—1:20*****Invited Papers*****1:25****2pAAa1. Directional characteristics of musical instruments, and interactions with performance spaces.** Jürgen Meyer (Braunschweig, Germany) and Uwe J. Hansen (Indiana State Univ., 64 Heritage Dr, Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu)

The seminal work: "Acoustics and the Performance of Music" by Jürgen Meyer, translated by Uwe Hansen, includes a summary of decades of groundbreaking measurements in the acoustics laboratory of the PtB (Physikalisch-technische Bundesanstalt—Physical and Technical Federal Institution—Germany's Bureau of Standards). Interactions with faculty and students of the School of Music in Detmold, Germany, as well as with numerous audio engineers and performers have contributed to an understanding of the significance of these data. Directional characteristics of a number of musical instruments will be reviewed and discussed, as well as their effects on seating arrangement in the orchestra and interactions with performance spaces.

1:45**2pAAa2. Sound radiation properties of musical instruments and their importance for performance spaces, room acoustics measurements or simulations, and three-dimensional audio applications.** René E. Caussé, Markus Noisternig, and Olivier Wausfel (Ircam - UMR STMS CNRS - UPMC, 1 Pl. Igor Stravinsky, Paris 75004, France, Rene.Causse@ircam.fr)

The directionality of the radiated sound is very specific to each musical instrument. The underlying radiation mechanisms may, for instance, depend on the structure of the vibrating body (e.g., string and percussion instruments) or on the spatial distribution of the opening holes (e.g., bells and open finger holes for wind instruments). A good knowledge of the radiation pattern of instruments is essential for many applications, such as orchestration, room acoustics, microphone techniques for live sound and recording, and virtual acoustics. In the first part, we will review previous works on sound source radiation measurement and analysis, discuss the underlying acoustic principles, and try to identify common mechanisms of radiation in musical instruments. In the second part, we will illustrate various projects undertaken at IRCAM and dedicated to the measurement and modeling of the directivity of instruments, to the objective and perceptual characterization of room acoustics, and to the real-time synthesis of virtual source radiation for musical performances. For this latter, several approaches are discussed according to the underlying physical formalisms and associated electroacoustic setups (e.g., spherical loudspeaker arrays, wave field synthesis).

2:05**2pAAa3. Database of musical instruments directivity pattern.** Noam Shabtai, Gottfried Behler, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstraße 5, Aachen D-52074, Germany, gkb@akustik.rwth-aachen.de)

The directivity pattern of an acoustic source describes the manner in which the sound radiates from it in the spatial domain. It may be used in virtual reality applications to improve the sense of realism perceived by the user. This work presents a directivity pattern database of 41 historical and modern orchestral instruments. The generation of this database includes the recording session in an anechoic chamber using a surrounding spherical microphone array, followed by a preliminary stage of isolating steady parts from the raw signals. Then, calibration is applied by normalizing the signals with the electrical channel gains and with the microphone gains. The fundamental frequency and overtones are then detected and the energy at each harmonic is saved for each played tone. Source centralization is applied in order to align the acoustic center of the sound source to the physical center of the microphone array. Last, a directivity pattern is generated in the spherical harmonics domain for each third-octave band by averaging the directivity pattern at all the overtones with a frequency belongs to that band.

2:25

2pAAa4. Challenges of musical instrument reproduction including directivity. Franz Zotter and Matthias Frank (Inst. of Electron. Music and Acoust., Inffeldfasse 10/3, Graz 8010, Austria, zotter@iem.at)

Reproduction of music from a solo instrument or singers by a single loudspeakers can suffer from lack of presence and liveness if the directivity is missing. In particular, natural solo music contains the effects of a time-varying directivity with particularities in the directivity index, shape, yielding different coloration of diffuse reverb, early reflection, and direct sound. Focusing on the root of technical realization, it would be desirable to capture and reproduce a recording of an instrument played within a spherically surrounding microphone array in an anechoic chamber. This contribution reviews fundamentals (the soap bubble model) and technical solutions for this particular recording and playback problem, utilizing surrounding spherical microphone arrays and compact spherical loudspeaker arrays, also an application in which a trombone with directivity had been transmitted live from Graz to Paris. However, these solutions need to be carefully used as they hide some essential challenges: Surrounding spherical arrays suffer from the acoustical centering problem and the comb-filtering artifact it creates, and compact spherical loudspeaker arrays for directivity synthesis are subject to a trade off between bandwidth and spatial resolution against temporal resolution. For some of these challenges, alternative approaches will be addressed.

2:45

2pAAa5. Development, evaluation, and validation of a high-resolution directivity measurement system for live musical instruments. K. J. Bodon and Timothy W. Leishman (Phys., Brigham Young Univ., Provo, UT 84602, joshuabodon@gmail.com)

A measurement system has been developed to assess high-resolution directivities of live musical instruments. It employs a fixed, semicircular microphone array, a musician/instrument rotation system, and repeated note playing to produce 5-degree angular resolutions in both the polar and azimuthal angles. Its 2,522 spherical measurement positions reveal feature-rich, frequency-dependent directivity patterns. To date, a total of 16 wind and string instruments have been measured with the system. They were recorded as musicians repeated chromatic scales over standard working ranges following 5-degree rotations in the azimuthal angle, until a full revolution was completed. Directivity patterns of the first five partials of each note have been calculated and plotted as individual directivity balloons. While the approach provides high-resolution directivity results with reasonable numbers of microphones and data acquisition channels, it also has disadvantages, including lengthy recording and processing times. Special techniques have been developed to reduce the effects of nonideal measurement circumstances, including playing variances, musician movement, etc. A series of validation tests were performed using loudspeakers to simulate musicians under varying but controlled conditions. This presentation will discuss the methods and results of the work and provide comparisons to lower-resolution measurements.

3:05–3:20 Break

3:20

2pAAa6. Non-impulsive signal deconvolution for computation of violin sound radiation patterns and applications in sound synthesis. Alfonso Perez Carrillo, Jordi Bonada (Music Technol. Group, Universitat Pompeu Fabra, Roc Boronat 138, Barcelona 08018, Spain, alfonso.perez@upf.edu), Vesa Valimäki (Aalto Univ., Helsinki, Finland), Andres Bucci (Music Technol. Group, Universitat Pompeu Fabra, Barcelona, Spain), and Jukka Patynen (Aalto Univ., Helsinki, Finland)

This work presents a method to compute violin body impulse responses (BIR) based on deconvolution of non-impulsive signals. This newly conceived approach is based on a frame-weighted deconvolution of excitation and response signals. The excitation, consisting of bowed glissandi, is measured with piezoelectric transducers built into the violin bridge and the response is measured as sound pressure with microphones. Based on this method, several research works have been carried out in the areas of acoustics and sound synthesis. First, by placing multiple microphones at different angles around the violin, we were able to compute a dense grid of 3D sound radiation patterns without restrictions in the frequency range. Second, the computed BIRs can be convolved with a source signal (captured with the same bridge-transducer and using the same violin), obtaining a highly realistic violin sound very similar to that of a microphone recording. The multiple impulse responses at different directions make has been used to enhance sound synthesis with spatialization effects. Finally, a bowing machine was built to perform repeatable glissandi and therefore be able to compute BIRs across different violins. The bowing machine has been used to compute cross-BIRs that map the pickup signal of electric violins to the radiated acoustic sound of acoustic violins, which allows to imitate the sound of any measured acoustic violin with an electric counterpart.

3:40

2pAAa7. Influence of the instrumentalist on the sound of the concert harp. Jean-Loïc Le Carrou (Sorbonne Universités, UPMC Univ Paris 06, UMR CNRS 7190, LAM-D'Alembert, 11 rue de Lourmel, LAM - D'Alembert / CNRS / UPMC, Paris 75015, France, jean-loic.le_carrou@upmc.fr), Delphine Chadefaux (Aix-Marseille Univ., Inst. of Movement Sci., UMR CNRS 7287, Marseille, France), Baptiste Chomette, Benoît Fabre (Sorbonne Universités, UPMC Univ Paris 06, UMR CNRS 7190, LAM-D'Alembert, Paris, France), Francois Gautier (Laboratoire d'Acoustique de l'Université du Maine, UMR CNRS 6613, Le Mans, France), and Quentin Leclère (Laboratoire Vibrations Acoustique, INSA Lyon, Villeurbanne, France)

The sound of musical instruments comes from a subtle mix between its mechanical behavior and its interaction with the instrumentalist. For the concert harp, the instrumentalist defines the initial conditions which determine the vibratory contents of the strings. This vibration is then radiated through the soundboard and the sound-box over a range of 7 octaves. Besides, this radiation may be affected by the instrumentalist's physical presence next to the instrument. The aim of the talk is to show, thanks to ten years of research on the physics of the concert harp, how the instrumentalist and the instrument act in the sound from the instrumentalist gesture to the radiated sound. For that, specific set-up and models are carried out in order to carefully analyze each important step of the sound production : plucking, strings coupling, dynamical behavior of the soundboard and radiated sound. These studies are performed when the instrument is isolated, without the harpist, or in playing situation in a musical context. The results show, for instance, that the instrumentalist gesture is a part of the spectral content of the sound, whereas the instrument's design has consequences on the directivity of each string's partial.

2p TUE. PM

4:00

2pAAa8. A study of variance of spectral content and sound radiation in timpani player. Brett Leonard and Scott Shinbara (School of Music, Univ. of Nebraska at Omaha, 6001 Dodge St., SPAC 217, Omaha, NE 68130, bleonard@unomaha.edu)

Timpani, although limited in pitch material, have the ability to produce many subtle colors, exceeding most other membranophones. One of the most notable characteristics of timpani is the perceptual “bloom” of the sound as distance is increased with the drum. Timpanists spend many hours working on control of these sounds and the bloom through application of intricate mallet technique and striking location. Anecdotal evidence of these differences is passed down through generations of teachers and students, but very little objective data exists about the actual sound of the drums, and the variations that occur between players. This study endeavors to reveal the objective differences between players and techniques, particularly as it relates to the “bloom” of the sound as you move away from the drum. Control of this directivity and expansion of perceived sound at a distance may be the single most important factor in the quality of a timpanist’s sound. Measurements are taken at different distances and locations around the drum for more than 15 different subjects, revealing a complex and interesting spectral pattern radiating from the drum.

4:20

2pAAa9. Electric guitar—From measurement arrays to recording studio microphones. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St, Ste. 3, Lowell, MA 01854, alex@fermata.biz), Jim Anderson, and Agnieszka Roginska (New York Univ., New York, NY)

A joint research effort by the audio recording programs at the University of Massachusetts Lowell and New York University has made use of a 32-microphone measurement array in the quantification and visualization of the spectral radiation of musical instruments. Work to date has focused on electric guitar and piano. The measured directivities of the guitar amplifiers offer rich insight for the recording engineer. Traditional microphone selection and placement strategies formed over decades, before such data existed, are found to have merit. The data also shed light on those potentially unattractive microphone locations to be avoided. The measurements, taken with high spatial resolution, reveal a process for microphone placement as much as providing a window into showing exactly where to place them. Measurements of the acoustic radiation from electric guitar amplifiers reveal a spatial complexity that many recording engineers anticipate, and add valuable further insight.

Contributed Paper

4:40

2pAAa10. High-resolution measurements of speech directivity. Jennifer K. Whiting, Timothy W. Leishman, and K. J. Bodon (Dept. of Phys. and Astronomy, Brigham Young Univ., N203 ESC, Provo, UT 84606, lundjenny@comcast.net)

Directivity patterns of loudspeakers are often included in room acoustics simulation packages, but those of live sources are less common, partly because of the scarcity of reliable high-resolution data. In recent years, researchers at Brigham Young University have explored high-resolution directivities of musical instruments. Their methods have now been adapted to directivity measurements of live speech. The approach uses a semicircular array of 37 microphones spaced with five-degree polar-angle increments.

A subject sits on a computer-controlled rotating chair with his or her mouth aligned at the axis of rotation and circular center of the microphone array. He or she repeats a phonetically balanced passage at each of 72 five-degree azimuthal-angle increments. Transfer functions between a reference microphone signal from the rotating reference frame and every array microphone signal enable computations of high-resolution frequency-dependent directivity balloons. Associated coherence functions allow judgment of frequencies for which directivity data can be trusted. This presentation discusses the results of these measurements and compares them to previous measurements of speech and singing-voice directivities. Animations of directivity balloons over frequency show a more complete picture of speech directivity than has been previously published.

Session 2pAAb**Architectural Acoustics and Noise: Measuring Sound Fields in Healthcare Environments**

Gary Madaras, Cochair

Making Hospitals Quiet, 4849 S. Austin Ave., Chicago, IL 60638

James S. Holthrop, Cochair

*AcoustiControl LLC, 2464 Taylor Road Suite 214, Wildwood, MO 63040***Chair's Introduction—2:55*****Invited Papers*****3:00****2pAAb1. A multinational comparison of measurement methods and metrics in acoustic standards and guidelines for healthcare environments.** Gary Madaras (ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com)

The field of architectural acoustics is in the initial stage of a paradigm shift when it comes to quantifying sound fields in healthcare facilities. A growing number of practitioners and researchers are questioning whether existing acoustic measurement methods and metrics relate well to patient perception of quietness or medical outcomes. A multinational overview will be provided of various acoustic standards and guidelines for healthcare facilities. Acoustic measurement methods and metrics from different countries are compared to identify commonalities and discrepancies. An update is provided on the progress of the World Health Organization's revision of their 1999 Community Noise Guidelines, particularly the hospital noise section. Information is compiled and presented in order to begin the process of possibly defining new acoustic measurement methods and metrics that relate more strongly to patient perception of quietness and medical outcomes.

3:20**2pAAb2. Measurement of loud noncontinuous exterior noise sources on patients.** James S. Holthrop (AcoustiControl LLC, 2464 Taylor Rd. Ste. 214, Wildwood, MO 63040, jim@acousticontrol.com)

A method to measure non continuous exterior noise sources such as helicopters, waste removal trucks, emergency vehicles, and semi-trucks will be presented. The noise levels that can be generated by these sources can exceed 90 dBA, which can impact patients within the hospitals. As hospitals are 24 hour facilities these sources of noise can happen during both daytime and nighttime hours. Data will be presented on the various acoustical methodologies to quantify this type of noise to access the impact on patients in the hospital.

3:40**2pAAb3. Permanent sensor-based hospital noise discovery.** John Bialk (W2288 County Rd. E, Neshkoro, WI 54960, john@quietyme.com)

John Bialk, CEO of Quietyme, will discuss how using smart sensors and advanced analytics to measure sound levels in hospitals has revealed what really happens in healthcare settings. By measuring decibel levels once per second in every patient room, hallway, and nurse's station, Quietyme is able to uncover the exact sources of patient noise disturbances and better understand solutions to reduce them. With nearly 100 million points of hospital sound data, John Bialk has a rare and purely objective perspective on hospital noise and will explain the difference between what the data has revealed and common misconceptions. In addition, Quietyme has helped with a variety of noise studies testing rooms and John will reveal findings that relate to hard surface vs. carpet, healthcare noise levels during construction and more.

4:00**2pAAb4. Experiences evaluating acoustics in occupied hospitals.** Erik Miller-Klein and Matthew Roe (SSA Acoust., LLP, 222 Etruria St., Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

The acoustic performance of patient rooms for four fully operational Veterans Affairs hospitals were completed over the past year. The nurse and facility managers goal was to identify the causes and solutions to improve the "quiet at night" scores. The testing included measuring speech privacy between patients and staff, background sound, occurrence rate of noise impacts during nighttime hours, and reverberation time. The testing procedures and methods were completed in cooperation with the hospital staff to optimize accuracy of the results while maintaining patient privacy, safety, and comfort. This included short duration pink noise measurements from room to room, and from nurses stations to patient care areas; these measurements provided accurate speech privacy metrics, though required direct approval and coordination with the nursing staff. The occurrence rate and background noise level was evaluated with 12 hour continuously logging sound level meters in patient rooms on tripods near patients with the approval of patients and guidance of nursing staff.

4:20

2pAAb5. Measuring quiet time in neonatal intensive care units. Jonathan Weber, Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, jonryanweber@gmail.com), Ashley Darcy Mahoney (Nell Hodgson Woodruff School of Nursing, Emory Univ., Atlanta, GA), Myra Rolfes, Heather Cooper (Neonatal Intensive Care Unit, Children's Healthcare of Atlanta, Atlanta, GA), and Brooke Cherven (Nursing Res. and Evidence Based Practice, Children's Healthcare of Atlanta, Atlanta, GA)

The soundscape of critical care wards such as Neonatal Intensive Care Units (NICUs) are of particular concern due to the extremely sensitive nature of the patient population. NICUs must be conducive to providing care that enables infants to adapt to the extrauterine world without undue environmental stressors. Although the American Academy of Pediatrics (AAP) and others have set recommended noise limits in the NICU, studies consistently show units exceeding these standards. More nuanced aspects of NICU noise, such as source type, spectral content, fluctuations, and speech intelligibility are also of concern. A long-term study is being conducted that aims to improve NICU soundscapes, including measuring the impact of a Quiet Time (QT) evidence-based practice change. The study is a unique collaboration between engineering, architecture, nursing, and medicine. Detailed acoustic measurements were taken over a 18-month period to assess the soundscape in pre-QT, short-, mid-, and long-term post QT implementation periods. The study methodologies and results will be discussed, including considerations for the complexities of measuring sound fields in NICUs. Results are being used to identify and evaluate soundscape interventions and therefore advance understanding of how to design, measure, and implement healthy NICU soundscapes.

4:40

2pAAb6. How to identify and control noise that could cost your hospital money. Joe Mundell (Sonicu, 19 W. Main St., Greenfield, IN 46140, jmundell@sonicu.com)

Noise levels in hospitals are problematic. Today, patient opinions on noise, because of HCAHPS, determine hospital reimbursement. It's critical for hospitals to reduce noise levels in their facilities or face reductions in reimbursement. Patient perceptions of noise in hospitals is difficult to accurately measure and determine root cause. Many hospitals are implementing programs to address noise, but very few are making these decisions based on systematic sound monitoring data within their facility. Sound monitoring equipment installed in hospitals can accurately measure and identify both sources and patterns of noise in hospital. Real-time and historical sound data can help inform changes needed. We compare historic sound data from our experience monitoring sound levels in NICU's and compare findings to improving sound levels throughout hospital. We discuss how visual alarming alone is insufficient to maintain long term improvements. We also discuss the use of measuring "sound events" to identify true source of noise disturbances. We argue that hospitals that measure "sound events" can systematically improve their HCAHPS scores. In conclusion, sound recording equipment within hospitals will provide actionable data. This data can be used to implement structural and procedural improvements allowing hospitals to maximize reimbursement and improve outcomes.

Contributed Paper

5:00

2pAAb7. Opportunity for the session's participants to share their insights and lessons learned. Gary Madaras (ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com) and Jim Holtrop (AcustiControl, St. Louis, MO)

This session on measuring sound fields in healthcare environments will conclude with an open microphone period whereby participants can

contribute their brief insights and lessons learned from their past experiences performing acoustical measurements in healthcare settings. Contributions should be limited to measurement methods, standards, and interpretation of findings.

Session 2pABa

Animal Bioacoustics: Bioacoustics Across Disciplines: Emitting Sound

Philip Caspers, Chair

Mechanical Engineering, Virginia Tech, 1110 Washington Street, SW, MC 0917, Blacksburg, VA 24061

Contributed Papers

1:00

2pABa1. How flying CF-FM echolocating bats adapt to acoustically jammed environments: Quantitative evaluation. Daiki Goto, Shizuko Hiryu, Kohta I. Kobayashi, and Hiroshi Riquimaroux (Doshisha Univ., 1-3 Miyakotani Tatara, Kyotanabe 610-0321, Japan, dmo1003@mail4.doshisha.ac.jp)

Echolocating bats face the acoustical interferences from sounds of other conspecifics; they can fly without colliding each other and avoid surrounding obstacles. The purpose of this study was to reveal how CF-FM bats extract their own echoes in acoustically jammed environments. The Japanese horseshoe bats (*Rhinolophus ferrumequinum nippon*) were flown with conspecifics in the flight chamber. As the number of flying bats increased from one to seven, the duration of constant frequency (CF) components decreased, whereas the terminal frequency modulated (TFM) components were extended in both time and frequency ranges. In order to quantitatively evaluate behavioral responses under jamming conditions, a flying bat was exposed by artificially synthesized CF-FM pulses. As a result, the bats also changed the CF and TFM components as observed in the group flight experiment. This shows that the bats modify the characteristics of pulses to adapt to acoustical jamming and not to adapt to spatial jamming owing to other flying bats. These results suggest that TFM component was more important than CF component in extracting their own echoes during flight in acoustically jammed conditions. We will examine echolocation behavior when we manipulate CF components in context of jamming avoidance during Doppler-shift compensation.

1:15

2pABa2. Three-dimensional sonar beam control of FM echolocating bats during natural foraging revealed by large scale microphone array system. Kazuya Motoi, Miwa Sumiya (Life and Medical Sci., Graduate School of Doshisha Univ., Tataramiyakodani 1-3, Kyotanabe, Kyoto 610-0321, Japan, dmp1019@mail4.doshisha.ac.jp), Dai Fukui (The Univ. of Tokyo Hokkaido Forests Graduate School of Agricultural and Life Sci., Furano, Hokkaido, Japan), Kohta I. Kobayashi, Emyo Fujioka, and Shizuko Hiryu (Life and Medical Sci., Graduate School of Doshisha Univ., Kyotanabe, Kyoto, Japan)

In this study, 3-D flight paths and directivity pattern of the sounds emitted by *Pipistrellus abramus* during natural foraging were measured by a large scale microphone array system. The results show that the bats approached prey with covering the direction of them within their sonar beam. The means of horizontal and vertical beam widths were 49 deg and 46 deg, respectively. Just before capturing prey, the bats decreased the terminal frequency (TF) of the pulse. Simultaneously, the beam widths were expanded to 64 deg (horizontal) and 57 deg (vertical). We assumed a circular piston model to estimate how much the beam width was changed by decreasing the frequency of emitted pulse. It was found that the observed expansion of the beam width was smaller than those of theoretical estimations. This suggests that the bats decrease the TF of pulse for compensating their beam width narrowed by taking a large bite for the prey. We also measured echolocation calls and flight behavior of *Myotis macrodactylus* during natural foraging. *M. macrodactylus* uses FM echolocation pulse

which is similar to *P. abramus*, but they forage for prey above the water surface. We compare echolocation strategies between two FM bats with different foraging habitat.

1:30

2pABa3. Dynamic baffle shapes for sound emission inspired by horseshoe bats. Yanqing Fu (Biomedical Eng. and Mech., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, fyanqing@vt.edu), Philip Caspers, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bat noseleaves are intricate baffle structures that diffract the animals' ultrasonic biosonar pulses upon emission. Furthermore, horseshoe bats dynamically change the shapes of their noseleaves through muscular actuation. Motions have been previously described for two lancet parts, anterior leaf and lancet. In both cases, the observed motions resulted in changes to the opening angle of the noseleaf baffle. Here, experiments were carried out with simplified baffle shapes that mimic the dynamics seen in horseshoe bats. For the baffle walls to have an effect on the outgoing wavefields, the sound outlets have to be narrow at least in one direction so that their near-fields generate substantial sound pressure amplitudes on the surface of the baffle. The baffle geometry was found to play an important role in the generation of dynamic signatures in the emitted pulses. As the opening of the baffle was varied by small increments, concave baffle surfaces were found to result in much larger dynamic changes to the beampatterns than straight baffles surfaces. Hence, concave baffles were able to introduce large dynamic signatures into the pulses even for small changes in opening angles. This may match the situation in horseshoe bats where the concerned baffles are also concave.

1:45

2pABa4. Numerical modeling of acoustic propagation in Harbor porpoise's (*Phocoena phocoena*) head. Chong Wei (College of Ocean & Earth Sci., Xiamen Univ., Hawaii Inst. of Marine Biology, Lilipuna Rd., Kaneohe, Hawaii 96744, weichong3310@foxmail.com), Whitlow W. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI), Darlene Ketten (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Zhongchang Song (College of Ocean & Earth Sci., Xiamen Univ., Xiamen, China), and Yu Zhang (Key Lab. of Underwater Acoust. Commun. and Marine Information Technol. of the Ministry of Education, Xiamen Univ., Xiamen, China)

Harbor porpoises (*Phocoena phocoena*) use narrow band echolocation signals for locating prey and spatial orientation. In this study, acoustic impedance values of tissues in the porpoise's head were calculated from the Hounsfield Units (HU). A two-dimensional finite element model was set up base on the computed tomography (CT) scan data to simulate the acoustic propagation through animal's head. The far field transmission beam pattern in the vertical plane and the waveforms of the receiving points around the forehead were compared with prior measurement results, the simulation results were qualitatively consistent with the measurement results. The role of the main structures in the head such as air sacs, melon and skull in the acoustic propagation was investigated. Additionally, the relative sound pressure level within the porpoise's sonar field across the transitional near and far field was obtained to compare with the spherical spreading loss.

2:00–2:15 Break

2:15

2pABa5. Seeing the world through a dynamic biomimetic sonar. Philip Caspers (Mech. Eng., Virginia Tech, 1110 Washington St., SW, MC 0917, Blacksburg, VA 24061, pcaspers@vt.edu), Jason Gaudette (Naval Undersea Warfare Ctr., Newport, RI), Yanqing Fu (Eng. Sci. and Mech., Virginia Tech, Blacksburg, VA), Bryan Todd, and Rolf Mueller (Mech. Eng., Virginia Tech, Blacksburg, VA)

The outer baffle surfaces surrounding the sonar pulse emission and reception apertures of the biosonar system of horseshoe bats (family Rhinolophidae) have been shown to dynamically deform while actively sensing the environment. It is hypothesized that this dynamic sensing strategy enables the animal, in part, to cope with dense unstructured sonar environments. In the present work, a biomimetic dynamic sonar system inspired by the biosonar system of horseshoe bats has been assembled and tested. The sonar head features dynamic deforming baffles for emission (mimicking the bats' noseleaf) and reception (pinnae). The dynamic baffles were actuated to change their geometries concurrently with the diffraction of the emitted ultrasonic pulses and returning echoes. The time-variant signatures induced by the dynamic baffle motions were systematically characterized in a controlled anechoic setting and the interaction between emission and reception dynamic signatures was investigated. The sonar was further tested in the context of natural environments with a specific focus on the interaction of dynamic ultrasonic pulse packets with natural targets. For both experimental approaches, a sonar with static baffle shape configuration was used as a reference to establish the impact of dynamic features.

2:30

2pABa6. Use of vibrational duetting mimics to trap and disrupt mating of the Asian citrus psyllid, a devastating pest in Florida groves. Richard Mankin (Ctr. for Medical Agricultural and Veterinary Entomology, USDA ARS, 1700 SW 23rd Dr., Gainesville, FL 32608, Richard.Mankin@ars.usda.gov) and Barukh Rohde (Elec. and Comput. Eng., Univ. of Florida, Gainesville, FL)

The Asian citrus psyllid (ACP) is the primary vector of a devastating disease of citrus, huanglongbing, and efficient surveillance of ACP at low population densities is essential for timely pest management programs in Florida. ACP males search for mates on tree branches by producing vibrational calls that elicit duetting replies from receptive females. The males then search for the location of the reply. We constructed a vibration trap by using a microcontroller with signal detection software, a contact microphone to detect ACP calls, and a piezoelectric buzzer to produce calls. The buzzer plays back a female reply when a male calls, which stimulates the male to search and find it. In this report, we discuss the construction and operation of the vibrational trapping system. In addition, we discuss methods that have been developed in laboratory studies to interfere with ACP courtship and mating. Our goal is to develop field-worthy systems that target ACP infestations and reduce their populations.

2:45

2pABa7. Observations on the mechanisms and phenomena underlying vocalizations of the gray seals. Lukasz Nowak (Inst. of Fundamental Technol. Res., Polish Acad. of Sci., ul. Pawinskiego 5B, Warszawa 02-106, Poland, lnowak@ippt.pan.pl) and Krzysztof E. Skora (Hel Marine Station, Univ. of Gdansk, Hel, Poland)

Gray seals vocalize both underwater and above the water surface using variety of sounds. The analysis of the differences in acoustic parameters of the emitted sounds suggests that the underlying phenomena may involve several different, independent mechanisms, which have not been yet investigated and understood. The aim of the present study is to introduce some important conclusions regarding those mechanisms and phenomena, based on the results of the original, long-term experimental investigations, and

observations carried out at the sealarium of the Hel Marine Station of the University of Gdansk. Several thousands of vocalizations emitted by the mature specimens and cubs were recorded above and beneath the water surface, and analyzed for their acoustic parameters using the dedicated, developed software. The observations also involved video recording combined with synchronous acquisition of underwater sounds, which allowed to link the acoustic phenomena with the specific behavior of the animals. Based on the obtained results, the equivalent mechanical models of the corresponding anatomical structures responsible for generation of various sounds are proposed. An own, original classification of the vocalizations of the gray seals, based on the assumed separation of the involved generation mechanisms, is introduced.

3:00

2pABa8. Variations of fin whale's 20 Hz calls in the Gulf of California. Andrea Bonilla-Garzón (Biología Marina, Universidad Autónoma de Baja California Sur, Km 5.5 Carretera al Sur, Mezquitito, La Paz, Baja California Sur 23080, Mexico, naborillag@unal.edu.co), Eduardo Romero Vivas (Centro de Investigaciones Biológicas del Noroeste CIBNOR,S.C., La Paz, Baja California Sur, Mexico), and Jorge Urbán-Ramírez (Biología Marina, Universidad Autónoma de Baja California Sur, La Paz, Baja California Sur, Mexico)

The fin whale's song has been broadly described in different areas of the world, and it is well characterized, being the spectrogram the main representation used to extract descriptions and measures. Males produce the 20 Hz calls, which consist of a down-swept pulse series (18–30 Hz) with a fundamental frequency of 20 Hz. Each pulse has duration of approximately 1 s, with aggregation patterns of singlet, doublets and triplets. A time analysis of calls recorded by a High-frequency Acoustic Recording Packages (HARPs) localized in Punta Pescadero and Bahía de los Ángeles, south and north of the Gulf of California, México in the 2004 through 2008, revealed variations of the pulse that are not easily discernible through the use of the spectrogram. Results of the preliminary analysis of 100 calls are presented. Regional differences in duration, structure, and shape found could indicate a geographical separation of population units of fin whales in the Gulf of California.

3:15

2pABa9. The metabolic costs of producing clicks and social sounds differ in bottlenose dolphins (*Tursiops truncatus*). Marla M. Holt, Dawn P. Noren (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. East, Seattle, WA 98112, Marla.Holt@noaa.gov), Robin C. Dunkin, and Terrie M. Williams (Dept. of Ecology and Evolutionary Biology, Univ. of California Santa Cruz, Santa Cruz, CA)

Dolphins produce many types of sounds known to have distinct qualities and functionalities. Whistles, which function in social contexts, are much longer in duration and require close to twice the intranasal air pressure to produce relative to biosonar click production. Thus, it is predicted that whistle production would be energetically more costly but this prediction is complicated by the fact that clicks are generated at much higher signal intensities. We used flow-through respirometry methods to measure metabolic costs of social sound and click production in two bottlenose dolphins. For all signal types, metabolic rates were related to the energy content of the signals produced. When metabolic costs were compared for equal energy sound generation, clicks were produced at negligible costs relative to resting and at a fraction of the cost of social sound production. However, while the performed repetition rates during click production were similar to field measurements, those of social sounds were much higher compared to typical field values. Even when metabolic costs are adjusted for more realistic whistle repetition rates, results indicate that whistle generation is more energetically costly. These results have implications for predicting the biological consequences of vocal responses to noise under different behavioral contexts.

Session 2pABb

Animal Bioacoustics: Bioacoustics (Poster Session)

Benjamin N. Taft, Chair

Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405

Authors will be at their posters from 3:30 p.m. to 5:15 p.m. To allow authors an opportunity to see other posters in their session, all posters will be on display from 1:00 p.m. to 5:15 p.m.

Contributed Papers

2pABb1. Mice ultrasonic detection and localization in laboratory environment. Yegor Sinelnikov (Acoust., Stevens Inst. of Technol., 126 Liberty Ave., Port Jefferson, NY 11777, ysinelnikov@yahoo.com), Alexander Sutin, Hady Salloum, Nikolay Sedunov, Alexander Sedunov (Acoust., Stevens Inst. of Technol., Hoboken, NJ), and David Masters (Dept. of Homeland Security, Sci. and Technol. Directorate, WA, DC.)

The acoustic detection and localization of mice movement by monitoring their ultrasonic vocalization has been demonstrated in laboratory environment using ultrasonic system with three microphones that provides recording of ultrasound up to 120 kHz. The tests were approved by Stony Brook University Institutional Animal Care and Use Committee protocol. Signals were recorded in a set of discrete sequences over several hours. The locomotor activity was characterized by durations up to 3000 ms and wide spectral content, while the syllable vocalization constituted shorter 200 ms events, with a set of identifiable up and down frequency modulated tones between 3 kHz and 55 kHz. The Time Difference of Arrival (TDOA) to various microphones was calculated using cross correlation method and was applied for estimation for mice location. Mice are among the invasive species that have a potential of crossing borders of United States unnoticed in containers. This study demonstrates the feasibility of using acoustic methods for detection of potential rodent intrusions. [This work was sponsored by DHS S&T.]

2pABb2. Effect of noise on each song element in Bengalese finch: Change of acoustic features. Shintaro Shiba, Kazuo Okanoya, and Ryosuke O. Tachibana (Graduate School of Arts and Sci., The Univ. of Tokyo, 3-8-1, Komaba, Meguro-ku, Tokyo 153-8902, Japan, shiba.shintaro@gmail.com)

Certain acoustic features of vocalization involuntarily change in response to the presence of background noise (e.g., the Lombard effect). These changes, observed in many species, are considered to be induced by audio-vocal interaction. Bengalese finch (*Lonchura striata* var. *domestica*) is suitable for this study because they need real-time auditory feedback of their own song. By investigating the effect of noise on each distinct element (i.e. notes) of their song, we can get more detailed knowledge about audio-vocal interaction. Here we demonstrate the changes of intensity and fundamental frequency (F0) of notes of Bengalese finches' songs under noises. Two band-pass noises (High/Low) in two levels (Loud/Soft) were used. The High/Low noises had spectral bands of 4.0–7.8/0.2–4.0 kHz each, and the Loud/Soft noises were set to 70/60 dBA, respectively. As a result, intensity in two High conditions and F0 in High Soft condition increased, while these features of some notes decreased in Low conditions. Also some individual lowered F0 of almost all the notes in all the conditions. The results suggest that the change of these features might depend on the relation between noise frequency and the original note characteristics, and also individual tendency. [Work supported by JSPS KAKENHI #26240019.]

2pABb3. A study of Foley Sound based on analysis and compare of the bird's chirping. Ahn Iksoo (TeleCommun. & Information, soolsil Univ., 369 sangdo-ro. Dongjak-gu. Seoul. Korea, Seoul 156-743, South Korea, aibestman@naver.com), Seonggeon Bae (Daelim Univ., Anyang, South Korea), and Myungjin Bae (TeleCommun. & Information, Soolsil Univ., Seoul, South Korea)

This research verifies use value of Foley sound of bird's chirping used in Radio drama as sound contents by comparing and analyzing it with Actual sound. Radio drama utilizes sound of bird's chirping to show season, time, and place in various ways. Currently, this sound is used as sound effect in media by recording Actual sound of bird's chirping on site with portable recorder. Before when no portable recording device existed, they had to make Foley sound. To have comparative analysis of Actual and Foley sound of bird's chirping, production method of Foley sound tools and their usage were studied. To conclude, as Foley sound tools used for sound of bird's chirping are very unique and interesting there is a high possibility of developing them into Sound contents for Performance and Exhibition.

2pABb4. Rats became positive or negative states when listening to specific vocalizations. Yumi Saito (The Graduate School of Art and Sci., The Univ. of Tokyo, 3-6-17-302, Onitaka, Ichikawa, Chiba 2720015, Japan, ren_doll2000@yahoo.co.jp), Hiroko Kagawa (Brain Sci. Inst., RIKEN, Komaba, Japan), Shoko Yuki, and Kazuo Okanoya (The Graduate School of Art and Sci., The Univ. of Tokyo, Komaba, Tokyo, Japan)

Emotional contagion is the process in which one became the same emotional states of the others via a behavioral signal, and many species use acoustic cues as this signal. For example, in human beings, emotional contagion occurs when they get emotional vocalizations of others, like laughter or crying. Likewise, rats emit 50 kHz or 22 kHz specific ultrasonic vocalizations (USVs) associated with positive or negative context. In order to measure whether these acoustic cues have the valence of attractiveness or aversiveness, we utilized the cognitive bias task in rats. Rats were trained to respond differently for two neutral stimuli. One resulted in a positive outcome while the other produced negative outcome. Then the intermediate stimulus between the two was used to test whether that was interpreted as being positive or negative. After the 50 kHz USVs stimulation, rat interpreted the neutral stimuli as being positive, while after the 22 kHz USV stimulation, the same stimuli were regarded as negative. Results suggest that USVs can bring about positive or negative emotional states of listeners, and our findings indicated that specific USVs can work as acoustic emotional signal in the conspecific. [Work supported by JSPS KAKENHI #23118003.]

2pABb5. Neural responses to songbird vocalizations in the nucleus taeniae of the amygdala in Bengalese finches. Tomoko Fujii (Dept. of Life Sci., Graduate School of Arts and Sci., The Univ. of Tokyo, 3-8-1, Komaba, Meguro-ku, Tokyo 153-8902, Japan, 5152187362@mail.ecc.u-tokyo.ac.jp), Maki Ikebuchi (Cognition and Behavior Joint Res. Lab., RIKEN Brain Sci. Inst., Saitama, Japan), and Kazuo Okanoya (Dept. of Life Sci., Graduate School of Arts and Sci., The Univ. of Tokyo, Tokyo, Japan)

Emotions are important psychological processes that induce adaptive behavior in response to communication signals. Many species of songbirds form complex social relationships and communicate with others through a large variety of vocal sounds. Thus, songbirds could be a good model to search for the neural basis of emotion especially in a context of communication. Anatomical studies have shown that the nucleus taeniae of the amygdala (TnA) in birds corresponds to the medial amygdala in mammals. While the amygdala is suggested to be involved in recognition of conspecific vocalizations in rats and bats, the function of the TnA still remains unclear. The present study aimed to explore auditory response properties of the songbird TnA by electrophysiology. We examined activity of the Bengalese finch TnA neurons during the presentations of conspecific and heterospecific vocalizations, as well as synthesized sound. We demonstrated for the first time that a population of TnA neurons exhibited selective auditory responses to songbird vocalizations. Our findings suggested involvement of the songbirds TnA for the recognition of communicative sounds. Further investigation into the TnA response properties should be fruitful in understanding the relationship between emotion and vocal signals in animals. [Work supported by JSPS KAKENHI Grant # 26240019.]

2pABb6. Relating click-evoked auditory brainstem response waveforms to hearing loss in the bottlenose dolphin (*Tursiops truncatus*). Krysta L. Gasser Rutledge (Program in Audiol. & Commun. Sci., Washington Univ. in St. Louis, Campus Box 8042, 660 South Euclid Ave., St. Louis, MO 63110, gasserk@wusm.wustl.edu), Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), and James F. Finneran (U.S. Navy Marine Mammal Program, San Diego, CA)

Hearing sensitivity in captive Atlantic bottlenose dolphins was assessed using a portable electrophysiologic data collection system, a transducer attached to the pan region of the mandible, and non-invasive recording electrodes. The auditory steady-state response (ASSR) was evoked using sinusoidal amplitude-modulated tones at half octave steps from 20–160 kHz and utilized to determine the upper frequency limit of hearing (i.e., the frequency at which threshold was ≤ 120 dB re 1 μ Pa). An auditory brainstem response (ABR) was then recorded to a moderate-amplitude click (peak-equivalent sound pressure level of 122 dB re 1 μ Pa) and examined to determine if relationships existed between the upper frequency limit of hearing and the waveform characteristics of the click-evoked ABR. The ASSR and click-evoked ABR were measured in 6 bottlenose dolphins with varying hearing sensitivity and frequency range of hearing. A significant relationship existed between click-evoked ABR wave amplitudes and the upper frequency limit of hearing. Test times for assessment using frequency-specific ASSR and click-evoked ABR were ~ 45 minutes and 1 minute, respectively. With further definition of normative data, measurement of click-evoked ABRs could form the basis of an expedited electrophysiologic method for hearing screening in marine mammals.

2pABb7. Auditory sensitivity shift by attention in Mongolian gerbil. Hirayuki Miyawaki, Ayako Nakayama, Shizuko Hiryu, Kohta I. Kobayashi, and Hiroshi Riquimaroux (Graduate School of Life and Medical Sci., Doshisha Univ., 1-3 Tataru Miyakodani, Kyotanabe-shi, Kyoto-fu 610-0394, Japan, dmp1015@mail4.doshisha.ac.jp)

Mongolian gerbil, *Meriones unguiculatus*, communicate with others by various sounds. About 80% of those sounds range over 20 kHz. A threshold of hearing in this range, however, was about 20 dB higher than their most sensitive frequency (1 to 16 kHz). We proposed a hypothesis that the auditory sensitivity heightened when gerbils communicated with others. In order to test the idea, a cochlear microphonics (CM) was recorded under various behavioral context as a measure of auditory sensitivity. Under paired condition, a subject was set with another gerbil. The condition simulated the subject gerbils in their group, and the subject was considered to pay attention to

sounds of the company. The CM response under the situation was higher than under single situation. Then, we investigated if the CM increase was specific under the situation. Subject was trained to pay attention to a sound of conditioned stimulus by electric shock as a negative reinforcer. The sound worked as an alarm sound for gerbils. The CM response increased while the gerbil was paying attention to the sound. Those results suggest that the sensitivity of auditory periphery is raised by attention in various behavioral contexts.

2pABb8. Vocal plasticity after laryngeal nerve lesion in rodent. Hiroki Iwabayashi, Shizuko Hiryu, Kohta I. Kobayashi, and Hiroshi Riquimaroux (Graduate School of Life and Medical Sci., Doshisha Univ., 1-3 Tataru Miyakodani, Kyotanabe-shi, Kyoto-fu 610-0394, Japan, dmp1007@mail4.doshisha.ac.jp)

Several species of animals, human, bat, and songbird adaptively control and change their spectro-temporal structure of vocalizations depending on auditory feedback. On the other hand, most of the common laboratory rodent generally vocalize relatively simple sound, and have been regarded as having only limited or no vocal plasticity. In this experiment, we conducted unilateral mutilation of the inferior laryngeal nerve in Mongolian gerbil, *Meriones unguiculatus*, and recorded the effect of surgery and recovery of vocalization for about 2 months for evaluating their vocalization plasticity. Mongolian gerbil have several types of vocalization. A short (<50 ms) high frequency (>25 kHz) vocalization, called "greeting call," is the most commonly observed in their colony. After mutilation of the nerve, they stopped producing greeting call. From 18 days after surgery (18 DAS) recovery began; spectro-temporal structure of high frequency vocalization altered day by day, and on 66 DAS they were able to vocalized calls similar to greeting call. Currently, we are investigating role of audition on the recovery by combining the laryngeal neurotomy and auditory deprivation. Neomycin was administered in order to deafen the gerbil. We will discuss if auditory feedback is capable to modify vocal output in the rodent.

2pABb9. Analysis of bearings of vocalizing marine mammals in relation to passive acoustic density estimation. Julia A. Vernon, Jennifer L. Miksis-Olds (Appl. Res. Lab, The Penn State Univ., The Graduate Program in Acoust., The Penn State Univ., State College, PA 16803, jav232@psu.edu), and Danielle Harris (Ctr. for Res. into Ecological and Environ. Modeling, The Univ. of St. Andrews, St. Andrews, Fife, United Kingdom)

The use of passive acoustic monitoring in population density estimation of marine mammals is a current area of interest, providing an efficient and cost-effective alternative to visual surveys. One challenge that arises with this method is uncertainty in the distribution of individuals. With large arrays where instruments are placed randomly with respect to the animals, it is often assumed that animals are uniformly distributed with respect to the instruments; however with sparse arrays this assumption is likely violated and could lead to bias in the density estimates. Distribution can be better determined through consideration of the horizontal azimuths or bearings of vocalizing animals. This paper presents bearing estimates of fin whales around Wake Island in the Equatorial Pacific Ocean, using ambient recordings from the Comprehensive Nuclear-Test Ban Treaty Organization (CTBTO) hydrophones at this location. Bearings were calculated for calls detected automatically. Multiple automatic detectors were assessed for optimal performance. Spectrogram correlation was found to produce the best results and bearings were calculated on calls detected with this method. The bearings were calculated using time delay information from the cross-correlation of received signals. Seasonal variation in animal distribution is also discussed. [This work was supported by the Office of Naval Research.]

2pABb10. Pulsed sounds produced by amazon river dolphin (*Inia geoffrensis*) in the Brazilian Amazon: Comparison between two water turbidity conditions. Jéssica F. Melo, Thiago Amorim, and Artur Andriolo (Zooloogy, Universidade Federal de Juiz de Fora/ Federal University of Juiz de Fora R. José Lourenço Kelmer, s/n - Campus Universitário, Juiz de Fora, Minas Gerais 36036-900, Brazil, jessicafmelo@live.com)

Pulsed sounds produced by the amazon river dolphin compose their acoustic repertoire, and possibly have communicative function. We

analyzed the acoustic behavior of amazon river dolphin under two water turbidity conditions in the Brazilian Amazon. Data were collected during three days when animals exhibited foraging behavior. The sounds were classified according to spectrographic visual characteristics. The acoustic parameters were obtained for each category. The Wilcoxon test was applied to compare the acoustic parameters between black water (BW) and white water (WW). In a total of 525.47 minutes of recording in Juami-Japurá Ecological Conservation Unit, 70.6% was in black water and 29.4% in white water. We

found seven types (A, B, C, D, E, F, and G) of pulsed sounds. Types A, C, D and F were found exclusively in black water, while B (BW 46.4%; WW 53.6%), C (BW 80.8%; WW 19.2%) and G (BW 50.7%; WW 49.3%) were present in both conditions. The type B showed significant differences ($p < 0.01$) in low frequency, center frequency and peak frequency. Types C and G showed no difference in the parameters between the waters. This result indicates that water turbidity plays a role on the acoustic behavior of amazon river dolphin.

TUESDAY AFTERNOON, 3 NOVEMBER 2015

RIVER TERRACE 2, 1:30 P.M. TO 4:55 P.M.

Session 2pAO

Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics: Passive-Acoustic Inversion Using Sources of Opportunity II

Kathleen E. Wage, Cochair

George Mason University, 4400 University Drive, Fairfax, VA 22030

Karim G. Sabra, Cochair

Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Drive, NW, Atlanta, GA 30332-0405

Chair's Introduction—1:30

Invited Papers

1:35

2pAO1. Resolving small events within a dense urban array. Nima Riahi and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, nriahi@ucsd.edu)

We use data from a large 5200 element geophone array that blanketed 70 km² of the city of Long Beach (CA) to characterize very localized urban seismic and acoustic phenomena. Such small events are hard to detect and localize with conventional array processing techniques because they are only sensed by a tiny fraction of the array sensors. To circumvent this issue, we first identify significant entries in the large coherence matrix of the array (5200 × 5200 entries) and then use graph analysis to reveal spatially small and contiguous clusters of receivers with correlated signals. This procedure allows us to track local events over time and also characterize their frequency content. The analysis requires no prior medium information and can therefore be applied under conditions of relatively high scattering. We show how the technique exposes a helicopter traversing the array, several oil production facilities, and late night activity on a golf course.

2:00

2pAO2. Ships as sources of opportunity in acoustic source localization. Christopher M. Verlinden (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu)

Ships, which can be tracked using the Automatic Identification System (AIS), represent underutilized acoustic sources of opportunity (SOP) that can potentially be used to localize unknown sources, invert for environmental parameters, and extract information about the ocean environment such as the local time dependent Green's Function. An application of ships as SOP's used to localize targets is presented here. Rather than use replica fields for matched field processing (MFP) derived from acoustic models requiring detailed environmental input, data derived replicas from ships of opportunity can be used for assembling a library of replicas for MFP. The Automatic Identification System (AIS) is used to provide the coordinates for the replica library and a correlation based processing procedure is used to overcome the impediment that the replica library is constructed from sources with different spectra and will be used to locate another source with its own unique spectral structure. The method is simulated and demonstrated with field experiments. AIS information is retrieved from the United States Coast Guard Navigation Center (USCG NAVCEN) Nationwide AIS (NAIS) database.

2pAO3. Inversion of ship-noise scalar and vector fields to characterize sediments: A critical review of experimental results (2007–2014). Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels, Brussels Capital 1050, Belgium, jhermand@ulb.ac.be)

During geophysical and ecosystem surveys in Mediterranean and Atlantic shallow waters, research vessels, fishing boats, and passenger ferries were used as acoustic sources of opportunity to characterize bottom sediments of different types including mud, fluid mud, and sand. The ships sailed a straight course at constant engine speed while a compact vertical array of pressure or pressure-gradient sensors, deployed from a small boat, sampled the generated noise field over a broad frequency range. Different acoustic observables were investigated that do not depend on the knowledge of ship noise characteristics to mitigate the impact of such uncertainty. For sole pressure measurement, extraction and parameterization of striation structures in range-frequency spectrograms at different receiver depths allow to determine effectively compression wave speed and thickness of a sediment layer as compared to a reference. For additional pressure gradient measurement, vertical impedances are estimated at different depths whose range and frequency dependence are highly sensitive to bottom properties and, in particular, density. Global optimization via genetic algorithm and sequential Bayesian estimation via particle filtering maximize the degree of similarity between predicted and measured impedance data. The paper will critically review the experimental results and compare them with ground truth data. [Work supported by ONR, FNRS-CNPq, WBI-CAPEs, PREFACE project EC DG Env FP7.]

Contributed Paper

2:50

2pAO4. Estimation of low frequency sound attenuation in marine sediments. Rui Duan (Inst. of Acoust. Eng., Northwestern PolyTech. Univ., Xi'an, China), N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada, chapman@uvic.ca), Kunde Yang, and Yuanliang Ma (Inst. of Acoust. Eng., Northwestern PolyTech. Univ., Xi'an, China)

This paper presents a method for estimating low frequency sound attenuation from information contained in normal modes of a broadband signal. Propagating modes are resolved using the time-warping technique applied to signals from light bulb sound sources deployed at relatively short ranges of 5 and 7 km in the Shallow Water '06 experiment. A sequential inversion approach is designed that uses specific features of the acoustic data that are

highly sensitive to specific geoacoustic model parameters. The first feature is the modal group velocity which is inverted for sediment sound speed and sediment layer thickness. The second feature is the modal amplitude function which is inverted for water depth and receiver depths. The third feature is related to the modal amplitude spectrum and is inverted for source depth and sound attenuation. In each subsequent stage, estimates from the previous stage(s) are used as known values. The sequential inversion is stable and generates geoacoustic model parameter estimates that agree very well with results from other experiments carried out in the same region. Notably, the inversion obtains an estimated value of $0.08 \text{ dB}/\lambda$ in the band 120–180 Hz for the de-watered marine sediment characteristic of the continental shelf at the site.

3:05–3:25 Break

Invited Paper

3:25

2pAO5. Inversion of normal mode functions and seabed properties using non-linear time sampling of low-frequency bowhead whale calls on a vertical array. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Cedric Arisdakessian (Ecole Centrale de Lyon, Brest, France), and Julien Bonnel (ENSTA Bretagne, Brest, France)

Previous research has demonstrated that individual modal components of signals propagating in an acoustic waveguide can be isolated from a single hydrophone using non-linear time sampling (Bonnel *et al.*, 2010) and that mode shape functions can be directly extracted from vertical array data in a laboratory setting (Bonnel *et al.*, 2011). Here, we apply and extend the technique to experimental data collected in 2010 from a 15-element vertical array, covering only 63% of the 53 m water column off the continental slope of the Beaufort Sea, during the fall bowhead whale migration. We demonstrate how up to five distinct mode shapes can be extracted from frequency-modulated bowhead whale calls, along with the vertical array tilt and tilt direction, without resorting to assumptions about mode orthogonality across the array aperture. The extracted mode shapes can then be used to estimate whale call depth. Filtered modes (and mode cutoff information) can also be used to invert for whale call ranges and ocean bottom properties. [Work supported by the North Pacific Research Board and permitted by Shell Exploration and Production Company.]

3:50

2pAO6. Environmental inversion using bowhead whale calls in the Chukchi Sea. Graham A. Warner, Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. Ste. A405, Victoria, BC V8P 5C2, Canada, gwarner@uvic.ca), David Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper estimates environmental properties of a shallow-water site in the Chukchi Sea using bowhead whale calls recorded on an asynchronous cluster of ocean-bottom hydrophones. Frequency-modulated whale calls with energy in at least two (dispersive) modes were recorded on a cluster of seven hydrophones within a 5 km radius. The frequency-dependent mode arrival times for nine whale calls were used as data in a Bayesian focalization

inversion that considered the whale locations and range-independent environmental properties (sound-speed profile, water depth, and seabed geoaoustic profile) as unknown. A trans-dimensional inversion over the number of points defining the sound-speed profile and subbottom layers allows the data to determine the most appropriate environmental model parameterization. The whale-call instantaneous frequency, relative recorder clock drifts, and residual-error standard deviation are also unknown parameters in the inversion which provides uncertainty estimates for all model parameters and parameterizations. The sound-speed profile shows poor resolution but the thickness and sound speed for the upper sediment layer are reasonably well resolved. Results are compared to an inversion of controlled-source (airgun) dispersion data collected nearby which showed higher environmental resolution.

Invited Papers

4:05

2pAO7. Passive acoustics as a tool for assessment of seagrass ecosystems. Paulo Felisberto and Sergio Jesus (Univ. of Algarve, Campus de Gambelas, Faro 8005-139, Portugal, sjesus@ualg.pt)

Seagrass meadows are important coastal ecosystems, being the most productive biomes on Earth. The amount of produced oxygen is a key parameter for assessing the seagrass ecosystem metabolism and productivity. Several experiments conducted in *Posidonia oceanica* seagrass meadows have demonstrated that using active acoustic methods one can monitor the oxygen production. The acoustic method is sensitive to bubbles production and provides estimates at ecosystem level, what is an advantage to other methods. Healthy seagrass ecosystems are populated by several noisy marine taxa such as fishes and crustaceans, therefore the ambient noise can be used to assess the ecosystem metabolism and productivity. During two periods of one week in October 2011 and May 2013, acoustic and environmental data was gathered at several locations over a *Posidonia oceanica* bed in the Bay of Revellata in Corsica (France). The diurnal variability pattern of ambient noise characteristics (frequency band, power, directivity, and number of noisy sources) shows an evident correlation with oxygen production as measured by independent methods (optodes). The ambient noise data were dominated by impulse like waveforms associated with marine taxa. This work discusses the challenges faced on using these waveforms to invert for oxygen production.

4:30

2pAO8. Field calibration—Distributed vehicles and sensors and the use of their acoustic communications transmissions for probing the medium. Paul Hursky (HLS Res. Inc, 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

Underwater vehicles and float of various kinds are often used to provide distributed sensing or monitoring of the medium in which they are deployed. Such cooperating platforms typically communicate with each other using high-frequency, broadband acoustic communications. Such transmissions may provide much more information than is carried in their acomm payload, if they are used as ranging signals for navigation and probing signals for imaging the medium they travel through. We have collected data to explore this concept during two experiments, Makai 2005 in Hawaii and Radar 2007 in Portugal. We have previously shown how channel estimates gleaned from high-frequency probe signals from a moving platform and received on a receiving sensor array could be used to improve localization at lower frequency. We expand upon this previous work to demonstrate how such channel estimates at the relatively sparse locations where probes were transmitted by a vehicle can be interpolated to flesh out the channel impulse response throughout the localization search space. We also discuss how such transmissions can be used in aid of clock synchronization and navigation.

Session 2pBA

Biomedical Acoustics and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications II

Guillaume Haiat, Cochair

Multiscale Modeling and Simulation Laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Pierre Belanger, Cochair

*Mechanical Engineering, Ecole de Technologie Supérieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K1, Canada***Contributed Papers**

1:00

2pBA1. Three-dimensional and real-time two-dimensional topological imaging using parallel computing. Etienne Bachmann, Xavier Jacob (Lab. PHASE, 118 Rte. de Narbonne, Toulouse 31062, France, etiennebachmann@hotmail.fr), Samuel Rodriguez (I2M, Bordeaux, France), and Vincent Gibiat (Lab. PHASE, Toulouse, France)

We present the Fast Topological IMaging that has shown promising results to quickly process a picture by sending an ultrasonic plane wave within an unknown medium. This imaging algorithm is close to adjoint-based inversion methods but relies on a fast calculation of the direct and adjoint fields formulated in the frequency domain. The radiation pattern of a transducer array is computed once and for all, and then the direct and adjoint fields are obtained as a simple multiplication with the emitted or received signals, in Fourier domain. The resulting image represents the variations of acoustic impedance, and therefore highlights interfaces or flaws. Real-time imaging and high definition visualization both imply an expensive computation cost, that led us to implement this method on GPU (Graphics Processing Unit). Thanks to a massively parallel architecture, GPUs have become for ten years a new way to implement high performance algorithms. We used interoperability between OpenGL and CUDA to enable a real-time visualization. Experimental results in 2D/3D obtained with scalar waves are presented. At this time, the method has been implemented for acoustic

waves in fluids, with an initial homogeneous medium, but it can be extended to elastic media and more complex configurations.

1:15

2pBA2. Temperature dependence of shear wave speed in a viscoelastic wormlike micellar fluid. E. G. Sunethra K. Dayavansha, Cecille Labuda, and Joseph R. Gladden (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., P.O. Box 1848, University, MS 38677, sdayavan@go.olemiss.edu)

Wormlike micellar fluids are viscoelastic and can support shear waves. Phase transitions of the micellar aggregates are temperature dependent and can manifest as sharp changes in the shear wave speed as a function of temperature. In this work, the variation of shear speed with temperature of 200 mM CTAB/NaSal micellar fluid in a 5:3 ratio was studied. The dependence of shear wave speed on time between fluid synthesis and measurement was also investigated. Shear wave propagation through the fluid was observed as a time varying birefringence pattern by using a high speed camera and crossed polarizers and shear speed was calculated by edge tracking and wavelength measurement techniques. A gradual increase in shear wave speed was observed in the temperature range 20–40 °C. A phase transition was observed to occur between 6 and 7 °C. There was no evidence of variation of shear wave speed with time. The implications of the shear wave speed variation over a wide temperature range will be discussed.

Invited Papers

1:30

2pBA3. Optimized excitation for nonlinear wave propagation in complex media: From biomedical acoustic imaging to nondestructive testing of cultural heritage. Serge Dos Santos (Inserm U930 “Imaging and Brain”, INSA Ctr. Val de Loire, 3, Rue de la Chocolaterie, Blois, Centre-Val de Loire F-41034, France, serge.dossantos@insa-cvl.fr), Martin Lints (Inst. of Cybernetics, Tallinn Univ. of Technol., Tallinn, Estonia), Nathalie Poirot (GREMAN, IUT de Blois, Blois, France), and Andrus Salupere (Inst. of Cybernetics, Tallinn Univ. of Technol., Tallinn, Estonia)

The interaction between an acoustic wave and a complex media has an increase interest for both nondestructive testing (NDT) applications and for biomedical ultrasound. Today, new optimized excitations are generated thanks to the analysis of symmetry properties of the system such as reciprocity, nonlinear time reversal (TR) and other pulse-inverted (PI) techniques. Generalized TR based NEWS (Nonlinear Elastic Wave Spectroscopy) methods and their associate symmetry skeleton will be taken as an example. Among these family of “pulse coded excitation,” solitonic coding constitutes a new scheme in the sense that solitary waves are the best candidates for pulse propagation in nonlinear and layered media, such as tooth or skin. As another application of mixing properties in a wide frequency range, new broadband techniques are needed in the domain of the preservation of cultural heritage. The analysis of the composition of the stones is one of the key parameters in the study of aging historic buildings. The use of TR-NEWS based analysis combined with a FTIR-based system has shown a specific property of the tuffeau limestone where damaged sample contains calcite. In both domains, aging properties of complex medium are extracted thanks to the use of enhanced nonlinear signal processing tools.

1:50

2pBA4. Ultrasound molecular imaging of heterogeneous tumors to guide therapeutic ultrasound. Frederic Padilla (Labex DevWeCan, Lyon Univ., INSERM LabTAU Unit 1032, 151 Cours Albert Thomas, Lyon 69390, France, frederic.padilla@inserm.fr), Alexandre Helbert, Isabelle Tardy (Geneva Res. Ctr., Bracco Suisse SA, Plan-les-Ouates, Switzerland), Cyril Lafon, Jean-Yves Chapelon (Labex DevWeCan, Lyon Univ., INSERM LabTAU Unit 1032, Lyon, France), Mathew von Wronski, François Tranquart, and Jean-Marc Hyevelin (Geneva Res. Ctr., Bracco Suisse SA, Plan-les-Ouates, Switzerland)

Propagation through heterogeneous tissues may hamper the ability to correctly focus ultrasound, especially when targeting a tumor for drug delivery applications. To solve this issue, we propose to use ultrasound molecular imaging (USMI) for treatment planning of drug delivery triggered by focused ultrasound. In a model of orthotopic rat prostate tumor having heterogeneous B-mode appearance, we show that tumor extend can be precisely delineated with USMI targeting VEGFR-2 receptors. High intensity ultrasound bursts (Peak negative pressure -15 MPa) are then delivered within this delineated 3D volume to trigger the release of liposomal-encapsulated chemotherapy, by local initiation of inertial cavitation. Real time imaging show that cavitation is indeed restricted to the targeted area. In the animal group receiving both chemotherapy and cavitation ultrasound, a potentiation of the therapeutic effect of the drug is clearly observed. This study demonstrates experimentally that USMI is an effective imaging method to characterize heterogeneous tissues and to guide therapeutic ultrasound.

2:10

2pBA5. Structure-factor model for quantifying the ultrasound scattering from concentrated cell pellet biophantoms. Emilie Franceschini, Romain de Monchy (Laboratoire de Mécanique et d'Acoustique CNRS UPR 7051, Aix-Marseille Université, Centrale Marseille, LMA CNRS UPR 7051, 31 chemin Joseph Aiguier, Marseille 13009, France, franceschini@lma.cnrs-mrs.fr), and Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., New York, NY)

Ultrasonic backscatter coefficient (BSC) measurements were performed on K562 cell pellet biophantoms with cell concentrations ranging from 0.6% to 30% using ultrasound in the 10–42 MHz frequency band. The concentrated biophantoms mimic densely packed cells with known concentrations and are thus simplified versions of real tumor. Three scattering models, the fluid-filled-sphere model (FFSM), the Gaussian model (GM), and the structure factor model (SFM), were compared for modeling the scattering of the biophantoms. The GM and FFSM assume sparse, independently, and randomly distributed scatterers and are thus suitable for modeling dilute media; however, the SFM does not contain these assumptions and therefore can model dense media accurately. First, a parameter-estimation procedure was developed to estimate scatterer size and acoustic impedance contrast (assuming that cell concentrations were known *a priori*) and thereby compare theoretical with measured BSCs for all studied concentrations. The SFM yielded scatterer-radius estimates of $6.4 \mu\text{m}$, which were consistent with the cell radius measured by optical microscopy. Second, the ability of the three models to estimate the scatterer size and acoustic concentration was compared. These scatterer properties were predicted well by the SFM, whereas the GM and the FFSM underestimated cell size and overestimated acoustic concentration for the more-concentrated biophantoms.

2:30

2pBA6. Modified transfer function with a phase rotation parameter for ultrasound longitudinal waves in cancelous bone. Hirofumi Taki (Dept. of Electron. Eng., Graduate School of Eng., Tohoku Univ., 6-6-05, Aramaki-Aza-Aoba, Aoba-ku, Sendai, Miyagi 980-8579, Japan, taki@ecei.tohoku.ac.jp), Yoshiaki Nagatani (Dept. of Electronics, Kobe City College of Technol., Kobe, Japan), Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan), Katsunori Mizuno (Inst. of Industrial Sci., The Univ. of Tokyo, Tokyo, Japan), Toru Sato (Graduate School of Informatics, Kyoto Univ., Kyoto, Japan), and Hiroshi Kanai (Dept. of Electron. Eng., Graduate School of Eng., Tohoku Univ., Sendai, Japan)

The through-transmission ultrasound signal in cancelous bone consists of two longitudinal waves, called the fast and slow waves. The conventional propagation model assumes that the wavefront of an ultrasound wave passing through a bone specimen is flat and that each wave arrives simultaneously at all the points on a transducer surface. To compensate for the waveform change caused by the effect of the uneven wavefront received at a flat transducer, we propose a new transfer function modified with a phase rotation parameter. We also propose a fast decomposition method that requires only 5 seconds using a laptop personal computer. The proposed decomposition method using the modified transfer function succeeded to separate the two waves and characterize each of them accurately, where the normalized residual intensity power was less than -20 dB in the experimental study when the bone specimen thickness was from 6 to 15 mm. In the simulation study, the normalized residual power was less than -20 dB when the specimen thickness was from 3 to 8 mm. These results show that the transfer function with a phase rotation parameter is valid and that the proposed decomposition method has a great potential to provide good indicators of osteoporosis in real-time.

2:50

2pBA7. Simulation technique and time-frequency analysis of the acoustic wave inside cancelous bone. Yoshiaki Nagatani (Dept. of Electronics, Kobe City College of Technol., 8-3, Gakuen-higashi-machi, Nishi-ku, Kobe, Hyogo 651-2194, Japan, nagatani@ultrasonics.jp)

Since the cancelous bone inside long bones has porous structure, the acoustic wave propagating inside bone may bring us useful information for the diagnosis of osteoporosis or bone healing although it is difficult to be investigated. In this paper, simulation techniques for understanding the complex wave propagation will be presented. The three-dimensional finite-difference time-domain (FDTD) method is useful for the wave simulation inside the complicated network structure. Due to the alignment of trabeculae, the pulse wave propagating inside cancelous bone separates into two wave, called the fast wave and slow wave. Here, the effect of the elasticity on the two wave phenomenon will be shown as well as the effect of the viscoelasticity. In addition to the speed and amplitude of the two waves, the frequency of the waves also changes corresponding to the structure of the medium. However, the detailed properties of the two waves are hardly revealed stably because of the wave overlapping. Therefore, a stable technique for quantitative evaluation of the wave characteristics is important. In this paper, the procedure and examples of a robust method of time-frequency analysis, named multi-channel instantaneous frequency, will be presented. These techniques help us understand and analyze the detailed wave behavior.

2p TUE. PM

Contributed Paper

3:25

2pBA8. Quantitative ultrasound measurements in trabecular bone using the echographic response of a metallic pin: Application to spine surgery. Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr), Yoshiaki Nagatani (Kobe City College of Technol., Kobe, Japan), Seraphin Guipieri, and Didier Geiger (Multiscale modeling and simulation Lab., CNRS, Creteil, France)

Bone quality is an important parameter in spine surgery, but it remains difficult to be assessed clinically. The aim of this work is to demonstrate the feasibility of a QUS method to retrieve bone mechanical properties using an echographic technique taking advantage of the presence of a metallic pin

inserted in bone tissue. To do so, an experimental validation is performed and acoustical modeling is used in order to assess the influence of experimental errors. A metallic pin was inserted in bone tissue perpendicularly to the transducer axis. The echographic response of the bone sample was determined and the echoes of the pin inserted in bone tissue and in water were compared to determine speed of sound (SOS) and broadband ultrasonic attenuation (BUA), which was compared to bone volume fraction (BV/TV). A 2-D finite element model was developed to assess the effect of positioning errors. Moreover, the coupling of finite difference time domain simulation with high resolution imaging technique was used to understand the interaction between an ultrasonic wave and the bone microstructure. A significant correlation between SOS and BV/TV was found ($R^2 = 0.6$). The numerical results show the relative robustness of the measurement method, which could be useful to estimate bone quality intraoperatively.

Invited Papers

3:40

2pBA9. A combined modeling and experimental investigation of ultrasonic attenuation mechanisms in cancelous bone-mimicking aluminum foams. Lawrence H. Le (Radiology and Diagnostic Imaging, Univ. of AB, Edmonton, AB T6G 2B7, Canada, lawrence.le@ualberta.ca), Behzad Vafaiean (Civil and Environ. Eng., Univ. of AB, Edmonton, AB, Canada), Kim-Cuong T. Nguyen (School of Dentistry, Univ. of AB, Edmonton, AB, Canada), and Samer Adeeb (Civil and Environ. Eng., Univ. of AB, Edmonton, AB, Canada)

Bone is composed of cancelous and cortical bones. Cancelous bone, which is more complicated than cortical bone in terms of structure, is inhomogeneous, anisotropic, and porous. The trabeculae form interconnected network with viscous marrow filling the pores. The presence of trabeculae makes cancelous bone a highly scattering medium. Trabecular bone spacing is considered an important parameter to detect change in bone tissue microstructures. However, due to high porosity and heterogeneity of human cancelous bones, the underlying mechanisms of interactions between ultrasound and cancelous bones are still not fully understood. The water-saturated aluminum foams were previously studied for the suitability of cancelous bone-mimicking phantoms. The ligament thickness and pore size of the foam samples were very similar to those of human cancelous bones. Recently, we performed the micro-scale *elastic* modeling of broadband ultrasound traveling through the water-saturated aluminum foams using the standard Galerkin finite element method. The simulated results were compared with the experimental measurements using the derived broadband ultrasound attenuation coefficients. The results strongly suggested that wave scattering and mode conversion were the dominant attenuation mechanisms of ultrasound propagating in aluminum foams. The study further demonstrated the capability of the finite element method to effectively simulate the signatures of the ultrasound signals propagating in fluid-filled weakly absorptive porous structures.

4:00

2pBA10. Ultrasonic guided waves in bone system with degradation. Dhawal Thakare (Mech. Eng., Indian Inst. of Technol. Madras, 410 MDS, Chennai, TN 600036, India), Pierre Belanger (Mech. Eng., Ecole de Technologie Supérieure, Montreal, QC, Canada), and Prabhu Rajagopal (Mech. Eng., Indian Inst. of Technol. Madras, Chennai, TN, India, prajagopal@iitm.ac.in)

This paper investigates the feasibility of using ultrasonic guided waves for assessing the cortical bone and hence detect conditions such as osteoporosis. Guided wave propagation in bone systems modeled as multi-layered tubular structures consisting of anisotropic bone filled with viscous marrow and surrounded by tissue is studied using the Semi Analytical Finite Element (SAFE) method. Effects of changes to cortical bone thickness and mechanical properties are investigated. An attempt is also made to consider bone anisotropy in the models. The results, validated by experiments with bone phantoms, show that material and geometric condition strongly impacts the velocity of guided waves supported in the bone system. Identification of optimal guided wave modes for practical assessment is also discussed.

4:20

2pBA11. Axial transmission assessment of trabecular bone at distal radius using guided wave. Daniel Pereira, Alexandre Abid, Lucien Diotalevi (Dept. of Mech. Eng., Ecole de Technologie Supérieure, 6347 Christopher Colomb, Montreal, QC H2S2G5, Canada, pereira.ufrgs@gmail.com), Julio Fernandes (Ctr. de recherche l'Hôpital du Sacré-Coeur de Montréal, Montreal, QC, Canada), and Pierre Belanger (Dept. of Mech. Eng., Ecole de Technologie Supérieure, Montreal, QC, Canada)

The diagnosis of osteoporosis at skeletal sites composed mainly of trabecular bone, such as distal radius, can be considered clinically more relevant than cortical bone regions. Thus, the possibility to merge the potential of guided waves method to the clinical relevance of trabecular bone assessment is extremely motivating and has not yet been explored in details. Therefore, the objective of this paper is to investigate the feasibility of guided waves method to detect variation in the mechanical properties of trabecular bone at the distal radius using axial transmission. Due to the complexity of the distal region, three-dimensional finite elements simulations were performed using a bone model built from a computed tomography image of a human radius. The accuracy of the numerical model was experimentally validated using a bone phantom. The simulated excitation was transmitted to the radius using a longitudinal load applied to a small circular region at the extremity of distal radius. The identification of the guided waves was evaluated using 2DFFT transform. The results showed that the excitation imposed at the extremity of the distal radius creates guided waves that propagate along the long bone. Furthermore, the identified modes showed high sensitivity to the trabecular bone properties.

4:35

2pBA12. Guided wave velocity sensitivity to bone mechanical property evolution in cortical-marrow and cortical-trabecular phantoms. Alexandre Abid (Health Technology, Ecole de Technologie Supérieure (ETS), 3860 rue saint hubert appartement no. 1, Montreal, QC H2L 4A5, Canada, alexandreabid@hotmail.com), Dhawal Thakare (Indian Inst. of Technol. Madras, Chennai, India), Daniel Pereira (Mechanical, Ecole de Technologie Supérieure (ETS), Montreal, QC, Canada), Prabu Rajagopal (Indian Inst. of Technol. Madras, Chennai, India), Julio Fernandes (Res. Ctr. of Sacre Coeur Hospital, Montréal, QC, Canada), and Pierre Bélanger (Mechanical, Ecole de Technologie Supérieure (ETS), Montréal, QC, Canada)

Guided waves methods are sensitive to the mechanical properties of the material into which they are propagating. Guided waves methods are promising to detect osteoporosis; they are cost effective and do not expose the patient to radiation. Numerous studies have focused on the first arriving signal using a plate approximation. The aim of this study is to identify the best frequency/mode combinations for cortical-marrow and cortical-trabecular cylinder phantoms based on the sensitivity of the mode to mechanical property changes. Two different setups were used, axial transmission in a small diameter cortical-marrow phantom and circumferential propagation in a larger diameter cortical-trabecular phantom. Experiments for each method were carried out with in-plane and out-of-plane excitation, using frequencies from 50 to 200 kHz and were in good agreement with a 3D finite element model and dispersion curves extracted using semi-analytical finite element modeling. The modes' velocity was identified using either first zero

crossing, short time Fourier transform or 2D Fourier transform. The evolution of the mechanical properties from a healthy bone to an osteoporotic bone was simulated using the finite element models. The results of this study will be used to further develop the technique using highly sensitive mode and frequency combinations.

4:50

2pBA13. An experimental study on the effect of temperature on the acoustic properties of cranial bone. Alec Hughes and Kullervo Hynynen (Dept. of Medical Biophys., Univ. of Toronto, 101 College St., Rm. 15-701, Toronto, ON M5G 1L7, Canada, ahughes@sri.utoronto.ca)

Transcranial focused ultrasound is increasingly being used as an alternative non-invasive treatment for various brain disorders, including essential tremor, Parkinson's disease, and neuropathic pain. These applications necessitate an understanding of the complex relationship between temperature and acoustic properties of cranial bone. In particular, the longitudinal speed of sound and attenuation coefficients will be investigated. In this study, *ex vivo* skull caps were heated to temperatures ranging from 20 to 50 °C, and ultrasound pulses were transmitted through the skull caps using a spherical transducer of 5 cm diameter and 10 cm focal length, at clinically relevant frequencies of 0.836 and 1.402 MHz. A thin Mylar film was placed at the focus, and a laser vibrometer was used to receive the ultrasound pulse transmitted through the skull. It was found that there was a measurable change in the phase and amplitude of the received signal, implying a change in both the speed of sound and attenuation of the bone at different temperatures. It was also found that these changes were completely reversible. These results imply that at sufficiently high cranial bone temperatures, the assumption of temperature-independent acoustic properties of bone may become invalid.

5:05

2pBA14. Assessing dental implant stability using quantitative ultrasound methods: Experimental approach and numerical validation. Romain Vayron and Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr)

Dental implants are widely used for oral rehabilitation. However, there remain risks of failure that are difficult to anticipate. The objective of this study is to investigate the potentiality of quantitative ultrasound (QUS) to assess dental implant stability. To do so, the implant is initially completely inserted in the proximal part of a bovine humeral bone sample. The 10 MHz ultrasonic response of the implant is then measured and a quantitative indicator I is derived based on the rf signal obtained. Then, the implant is unscrewed by 2π radians and the measurement is realized again. The procedure is repeated seven times and the indicator is derived after each rotation of the implant. Analysis of variance (ANOVA) ($p < 10^{-5}$) tests revealed a significant effect of the amount of bone in contact with the implant on the distribution of I . In parallel, a finite element model is developed in order to model the ultrasonic propagation in the implant surrounded by bone tissue. The results show the feasibility of our QUS device to assess implant primary stability. A good agreement is obtained between experimental and numerical results. This study paves the way for the development of a new approach in oral implantology.

Session 2pEA

Engineering Acoustics: Analysis of Sound Sources, Receivers, and Attenuators

Kenneth M. Walsh, Chair

K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

1:30

2pEA1. Experimental and computational multi-port characterization of a circular orifice plate. Stefan Sack and Mats Åbom (The Marcus Wallenberg Lab., The Royal Inst. of Technol., Teknikringen 8, Stockholm 100 44, Sweden, ssack@kth.se)

The generation and scattering behavior of fluid machines in duct systems is of great interest to minimize sound emission, for instance of air condition systems. Such systems may be described as linear, time-invariant multi-port networks containing passive elements, that scatter existing sound fields and active elements, that emit noise themselves. The aim of any multi-port analysis is to determine the direction-dependent transmission and reflection coefficients for the propagating wave modes and the sound generation. This parameters may be ascertained either numerically or experimentally. In a first step, external sound fields dominating the source are applied to determine the system scattering. In a second step, the source strength can be computed. Once these characteristic data are determined for all elements of interest, the sound scattering and emission of every considerable combination of multi-ports can be calculated easily. This paper shows an experimental study as well as a numerical approach to determine both, the passive and the active characterization of an orifice plate in a circular duct in presence of higher order acoustic modes. An enhanced measurement procedure is presented, and the results of this procedure are compared with data extracted from a full compressible DDES flow-computation.

1:45

2pEA2. Optimization of wave surface for planar loudspeaker array reproduction. Akio Ando (Faculty of Eng., Univ. of Toyama, 1-10-11 Kinuta Setagaya, Tokyo 157-8510, Japan, andio@a.memail.jp), Takuya Yamoto (Graduate School of Sci. and Eng., Univ. of Toyama, Toyama, Japan), and Daisuke Yamabayashi (Faculty of Eng., Univ. of Toyama, Toyama, Japan)

Over the past few decades, a considerable number of studies have been made on the sound field reproduction by loudspeaker array. It is well known that the reproduced sound field suffers from various errors caused by the finite and discrete loudspeaker array. The error by the finite array is a so-called truncation error, which is a kind of diffraction effect resulting from the absence of secondary sources outside the finite area. Such error sometimes brings the degradation of reproduced wave surface and frequency response. In this study, the weighting coefficient vector for loudspeaker input is introduced, meaning that the input signal fed into each loudspeaker is multiplied by an element of the vector. The input signal is calculated according to the Rayleigh I integral. The vector is solved such that the integration of square error between the ideal and synthesized wave surface over the target area is minimized under the constraint that all elements of the vector have non-negative values. Various optimization methods are compared in this problem. It is shown that the shape of wave surface is improved by the optimization and the improvement of wave surface also brings the improvement of frequency response.

2:00

2pEA3. A comparative study of acoustics and vibration analysis for wearing bearings. Bethany Snow, Chris Dunn, Lin Lin, and James Smith (Univ. of Southern Maine, 37 college Ave., Gorham, ME 04038, bethany.snow@maine.edu)

Monitoring of progressive wear and early diagnosis of damage are critical to a rotating machine to avoid dangerous, costly, time consuming failures. A common symptom of wear in rotating machines is characterized by an increase in noise and/or vibration levels. It is usually monitored by either accelerometers or acoustic emission sensors, both of which are required to be mounted on the machine. Acoustics measurement using microphones has its unique non-contact advantage of monitoring the sound that caused by the wearing. Some work has been done on investigating the power of the acoustic signals of wearing machines; however, the principal indicator of failure onset and the relation between the spectra of the vibration signal and the non-contact acoustic signal is unclear. In this project, both acoustic and vibration signals were measured to investigate a machine with bearings of different degrees of wear. Acoustic signal were recorded at various azimuth settings and compared to the vibration signal. Frequency analysis and signal processing techniques were applied to compare the information contents and sensitivities of those two measurements.

2:15

2pEA4. Nonlinear acoustic response of stand-off tubes used in acoustic pressure measurements—Part I: Experimental study. Anthony Rehl, John M. Quinlan, David Scarborough, and Ben Zinn (Aerosp. Eng., Georgia Inst. of Technol., 635 Strong St., Atlanta, GA 30332, ajrehl@gatech.edu)

Rocket engine combustion instabilities, which lead to rapid engine failure through enhanced heat transfer rates and high-cycle fatigue, continue to be the most serious concern facing engine designers. Experimental testing and pressure measurements remains the best approach to determine the susceptibility of an engine design to acoustically coupled combustion instabilities. But, the harsh, high-temperature environment requires that pressure transducers be remotely mounted to the engine's main chamber using "sense-tubes," thereby creating an area-contraction at the connection point between the sense-tube and combustor. Preliminary measurements showed large discrepancies between sense-tube measured and engine acoustic pressure amplitudes. To elucidate these discrepancies, this experimental study measured the nonlinear response of the area-contraction/sense-tube geometry. Specifically, the sense-tube was attached to a two-microphone impedance tube, allowing the measurement of the acoustic impedance of the combined area-contraction and sense-tube; the acoustic pressure was also measured at the sense-tube termination, allowing the the direct measurement of the frequency response function. Measurements were performed over a range of frequencies, area-contraction ratios, acoustic velocity amplitudes, and sense-tube length-to-diameter ratios. These measurements reveal that the acoustic response of the sense-tube was highly nonlinear—even for low amplitude forcing.

2:45

2pEA5. Nonlinear acoustic response of stand-off tubes used in acoustic pressure measurements—Part II: Analysis. John M. Quinlan, David Scarborough, Anthony Rehl, and Ben Zinn (Aerosp., Georgia Inst. of Technol., 635 Strong St., Atlanta, GA 30332, jmquinlan@gmail.com)

Rocket engine combustion instabilities, which lead to rapid engine failure through enhanced heat transfer rates and high-cycle fatigue, continue to be the most serious concern facing engine designers. Experimental testing and pressure measurements remains the best approach to determine the susceptibility of an engine design to acoustically coupled combustion instabilities. But, the harsh, high-temperature environment requires remotely mounting dynamic pressure transducers to the engine's main chamber using "sense-tubes." Experiments reveal that the acoustic response of the sense-tube is highly nonlinear due to the area-contraction at the connection point between the engine and sense-tube. This nonlinearity leads to large discrepancies between the combustor's actual and the remotely measured acoustic amplitudes. Our aim was to develop an accurate, nonlinear sense-tube acoustic response model including steady flow effects. Therefore, the governing equation for the area-contraction pressure drop was approximated using a Fourier based technique to develop expressions for the steady and acoustic pressure drop across the area change. The acoustic pressure drop model was then incorporated into a response model for the tube. Measurements of the acoustic response of the area-contraction without mean flow agree well with predictions of the developed model.

3:00

2pEA6. Finite element method assisted development of an analytic model to describe the acoustic response of tee-junctions. Dan Fries and David E. Scarborough (Aerosp. Eng., Georgia Inst. of Technol., 635 Strong St., Atlanta, GA 30318, dfries@gatech.edu)

Tee-junctions are a central element of almost any duct system. The acoustics of such are altered by the transversal branch defining the tee. Resulting phenomena have been studied experimentally, numerically and analytically in the past. However, a simple practical tool for prediction and fundamental understanding is still missing. Applying the Finite Element Method, this work attempted to analyze the tee-junction's behavior with higher-order methods. To this end the commercial software COMSOL-Multiphysics was used. Thereafter, a low-order plane wave network model was developed and compared with simulation results. An error analysis of the power transmission coefficient computed by the model and the FEM simulation showed the limits of this low-order approximation. Thus, a guideline is provided for when the plane wave approximation delivers meaningful results. Moreover, it allowed for adjustments to the model increasing the model's accuracy, using a semi-empirical expression for a frequency dependent length correction of the side-branch. This had been predicted by other authors but never given explicitly. For side branches longer than two main-duct diameters and a thickness equal to or smaller than the main duct, the model gives good results up to 85% of cut-on of the first higher order mode. The proposed model can easily be implemented to estimate the characteristic quantities of a tee-junction.

3:15

2pEA7. A micromachined microphone based on a field effect transistor and an electret for low frequency acoustic detection. Kumjae Shin, Min Sung, and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technol., KIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk 790-784, South Korea, forhim13@postech.ac.kr)

Miniaturized microphone has sensitivity degradation in low-frequency region because most microphones use a capacitive-type transduction

mechanism which has low-frequency cut-off. Therefore, detection of low-frequency sound is carried out using a microphone with large membrane and a large back chamber. To overcome this limitation, a micro-machined microphone based on a FET (Field effect transistor) and electret has been reported, and the feasibility of this system was demonstrated. The electric field due to the electret modulates the channel of the FET embedded in the membrane. The acoustic signal causes the FET mounted on the membrane to vibrate, which changes the separation between the channel of the FET and the electret. The resulting change in the electric field modulates the conductivity of the channel. The use of an integrated FET as a sensing mechanism in response to the electric field from the electret makes it possible to detect the displacement of the membrane directly. We describe a theoretical model for the low-frequency operation of this system and provide a comparison with experimentally measured results. A sensitivity analysis for the transduction mechanism is carried out, and the frequency response of the microphone is characterized using an acoustic measurement setup developed for low-frequency sound.

3:30

2pEA8. System and method for determining the level of a substance in a container based on measurement of resonance from an acoustic circuit that includes unfilled space within the container that changes size as substance is added or removed from the container. Robert H. Cameron (Icosahedron Corp., 714 Winter Dr, El Paso, TX 79902-2129, rcameron@elp.rr.com)

This abstract is for a poster session to describe a patent application made to the patent office in November 2012. The patent describes a system and method for determining the level of a substance in a container, based on measurement of resonance from an acoustic circuit that includes unfilled space within the container that changes size as substance is added or removed from the container. In particular, one application of this device is to measure the unfilled space in the fuel tanks of vehicles such as cars and trucks. For over 100 years, this measurement has been done by a simple float mechanism but, because of the development of tank design for vehicles that involve irregular shapes this method is increasingly less accurate. The proposed device will overcome these limitations and should provide a much more accurate reading of the unfilled space and therefore the amount of fuel in the tank since the total volume of the tank is known.

3:45

2pEA9. A stepped-plate radiator as a parametric array loudspeaker. Yonghwan Hwang and Wonkyu Moon (Postech, PIRO416, Postech, Hyoja-dong, Nam gu, Po hang KS010, South Korea, serenius@postech.ac.kr)

A parametric array loudspeaker can generate a high directional audible sound beam in the air by the nonlinear acoustic interaction widely known as a "Parametric Array." The advantage of the parametric array phenomenon is that it creates a narrow beam using the small aperture of an acoustic radiator. It can be used in many applications, including high directivity communication systems. However, the sound pressure level generated by the parametric array is very low, which necessitates a high efficiency and intensity of sound. This paper describes the application of a stepped-plate radiator in a parametric array loudspeaker. The stepped-plate radiator has two resonance frequencies ($f_1 = 75$ kHz, $f_2 = 85$ kHz) in the design frequency region, and three different steps that compensate for the phase difference in the flexural vibrating plate. This allows the generation of high directivity audible sound in the wide flat radiation bandwidth of the audible frequencies range (100 Hz to 16 kHz, difference frequency waves with equalization). [Work supported by ADD (UD130007DD).]

TUESDAY AFTERNOON, 3 NOVEMBER 2015

DAYTONA, 1:30 P.M. TO 3:00 P.M.

Session 2pED

Education in Acoustics: Take 5's

Jack Dostal, Chair

Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

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 begins. If you have more than one demo, sign-up for two consecutive slots.

TUESDAY AFTERNOON, 3 NOVEMBER 2015

DAYTONA, 3:15 P.M. TO 4:45 P.M.

Session 2pID

Interdisciplinary: Guidance From the Experts: Applying for Grants and Fellowships

Caleb F. Sieck, Cochair

*Applied Research Laboratories and Department of Electrical & Computer Engineering, The University of Texas at Austin,
4021 Steck Ave #115, Austin, TX 78759*

Anna C. Diedesch, Cochair

Hearing & Speech Sciences, Vanderbilt Univ., Nashville, TN 37209

Michaela Warnecke, Cochair

*Psychological and Brain Sciences, Johns Hopkins University, 3400 N Charles St, Dept Psychological & Brain Sciences,
Baltimore, MD 21218*

Christopher Jasinski, Cochair

University of Notre Dame, 54162 Ironwood Road, South Bend, IN 46635

A panel of successful fellowship winners, selection committee members, and fellowship agency members organized by the Student Council.

This panel will consist of Laura Kloepper, post-doc at Brown University and recipient of an NSF Postdoctoral Fellowship, Alberto Rivera-Rentas, Research Training Officer for NIH/NIDCD, Andrew Oxenham, Professor at the University of Minnesota and NIH reviewer, and Jason Summers, Chief Scientist at Applied Research in Acoustics and recipient of grants from government institutions and private research foundations.

The panelists will provide a brief introduction about themselves and answer questions regarding grant and fellowship opportunities and application advice.

Session 2pNS**Noise: Damage Risk Criteria for Noise Exposure II**

Richard L. McKinley, Cochair

Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Hilary L. Gallagher, Cochair

*Air Force Research Lab., 2610 Seventh St. Bldg. 441, Wright-Patterson AFB, OH 45433-7901***Chair's Introduction—1:00*****Contributed Paper*****1:05**

2pNS1. Military Standard 1474E: Design criteria for noise limits vs. operational effectiveness. Bruce E. Amrein (Human Res. & Eng. Directorate, Army Res. Lab., ATTN: RDRL-HRS-D, 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005, amrein@gmail.com)

In April 2015, the U.S. Department of Defense (DoD) published a significant revision to Military Standard 1474 (MIL-STD-1474): Design Criteria—Noise Limits. Through the efforts of a DoD multi-service working group, every aspect of MIL-STD-1474 has been revised to improve readability; reduce conflicting guidance; and consolidate requirements common to steady-state and impulsive noise produced by weapons systems, and ground-, air- and water-borne platforms. Noise requirements in military

environments differ significantly from typical industrial or occupational situations, and mission success requires offensive equipment and weapons to be more lethal and survivable than those used by the adversary. Producing material suitable for military operations requires unique design criteria often exceeding civilian national or international standards. MIL-STD-1474E provides the design tools and measurement techniques necessary to satisfy these unique and often contradictory requirements. For the first time in MIL-STD-1474E, a computer-based, electro-acoustic model of the human auditory system is used to evaluate hazard from impulsive noise events typical of weapon firing. This presentation describes the salient requirements necessary to produce and deploy military systems maximizing Warfighter effectiveness, while minimizing hearing damage caused by their use.

Invited Papers**1:20**

2pNS2. Demonstrating compliance of noise exposure experienced by British military aircrew with noise legislation in the United Kingdom. Susan H. James (Aircrew Systems, QinetiQ, Rm. 2022 A5 Bldg QinetiQ, Cody Technol. Park, Farnborough, Hampshire GU14 0LX, United Kingdom, shjames@qinetiq.com)

In 2005, the Control of Noise at Work Regulations were introduced into UK law and imposed two Exposure Limits Values, one for continuous noise and one for impulse noise. The Ministry of Defence (MOD) has mandated that the legislation will apply in full throughout the military, and hence, there has been a requirement to understand the types of exposure military operators are exposed to. Military aircrew have historically been exposed to high levels of continuous noise and the more stringent legislation has necessitated enhanced noise mitigation, including the development of new hearing protection technologies. However, in more recent years, military aircrew have become increasingly exposed to weapon firing and the MOD are now addressing the implications of the combined effects of the two different exposures and their impact on compliance with the legislation. In-flight measurement of continuous and impulse noise exposure has been conducted in a range of aircraft and the two exposures assessed for compliance with the legislative criteria. To meet the protection requirements in all platforms, enhanced hearing protection technologies are being developed, including Adaptive Digital Active Noise Reduction and comms processing techniques.

1:40

2pNS3. Noise exposure criteria for continuous noise: A case for reducing the 8 hour open ear exposure level. Hilary Gallagher, Richard L. McKinley (Air Force Res. Lab., Battlespace Acoust. Branch, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, hilary.gallagher.1@us.af.mil), Melissa A. Theis (Air Force Res. Lab., ORISE, Wright-Patterson AFB, OH), and Elizabeth A. McKenna (Air Force Res. Lab., Henry M Jackson Foundation, Wright-Patterson AFB, OH)

Personnel working in hazardous noise environments can be protected from noise induced hearing loss (NIHL) or other hearing related disabilities when their exposures are limited. Limiting noise exposures can be accomplished by engineering controls, administrative controls, and/or with the use of personal protective equipment. Damage risk criteria (DRC) were developed as a guide to limit personnel noise exposure in order to reduce the risk of NIHL. With the exception of the US Occupational Safety and Health Administration

(OSHA), the US and many European countries accepted the DRC of 85 dB for 8 hours with a 3 dB per doubling exchange rate. Recent studies of the auditory response to noise dose, conducted at the Air Force Research Laboratory, found that A-weighted open ear exposures may be more hazardous (i.e., have a higher effective noise dose) than equal A-weighted level protected ear exposures. A potential conclusion of these studies is that hearing protectors are more effective than previously thought and reducing the open ear exposure criteria may be needed in order to minimize NIHL. This presentation will describe the pros and cons of reducing the 85 dBA DRC.

2:00

2pNS4. Calculating total daily noise exposure using aircrew flight equipment noise attenuation and flight segment noise data. Daniel A. Gross (US Navy, 48110 Shaw Rd., Bldg. 2187, Patuxent River, MD 20670, Daniel.A.Gross@navy.mil)

The workplace of military aircrew is louder and more varied than most industrial environments. In the recent revision of MIL-STD-1474E, Noise Limits, octave band level design limits for helicopters have been replaced by a broader requirement based on noise exposure at the ear. This change brings 1474E into alignment with DoD hearing conservation policies which require that total daily noise exposure does not exceed an Leq of 85 dBA during an 8 hour period. Although these policies also require use of hearing protection whenever levels exceed 85 dBA, there is currently little written guidance for how to use hearing protector attenuation data to estimate exposure at the ear over the course of a typical mission profile. This paper will explain how attenuation data for flight helmets and communication headsets has been used by the Naval Air Systems Command to estimate noise exposure for crew and passengers in various types of aircraft.

2:20

2pNS5. Effects of noise exposure in combination with exposures to JP-8 jet fuel. David R. Mattie (711 HPW/RHDJ, 711 HPW/RHDJ, 2729 R St., Wright-Patterson Air Force Base, OH 45433-5707, david.mattie@us.af.mil), Larry D. Fechter (Loma Linda Veterans Affairs Medical Ctr., Loma Linda, CA), Jeff W. Fisher (National Ctr. for Toxicological Res., Little Rock, AR), John E. Stubbs (509 MDOS/SOGJ, Wright-Patterson Air Force Base, OH), and O'neil W. Guthrie (Northern Arizona Univ., Flagstaff, AZ)

The objective was to evaluate the potency of JP-8 jet fuel to enhance noise-induced hearing loss using inhalation exposure to fuel and simultaneous exposure to noise with male & female Fischer 344 (F344) rats for 6 h/d, 5 d/wk for 4 wk (200, 750, or 1500 mg/m³). Parallel groups of rats also received nondamaging noise (85 dB) in combination with fuel, noise alone (75, 85, or 95 dB), or no exposure to fuel or noise. Computer software was used to generate a pure and precisely filtered white noise file of one octave band, centered at 8 kHz. The filtered file was then played through electrodynamic shakers that induced vibration from the outside in the metal plenum at the bottom of each exposure chamber to produce noise. Significant concentration-related impairment of auditory function measured by distortion product otoacoustic emissions (DPOAE) and compound action potential (CAP) threshold was seen in rats exposed to combined JP-8 (1500 mg/m³) plus noise (85 dB) with trends toward impairment at 750 mg/m³ JP-8 plus noise (85 dB). JP-8 alone exerted no effect on auditory function. Noise was able to disrupt DPOAE and increase auditory thresholds when noise exposure was at 95 dB. Two additional studies with JP-8 (1000 mg/m³) and noise (85 dB), one with F344 rats and one with Long Evans rats did not affect auditory function. However, a pilot assessment indicated a central auditory processing dysfunction (i.e., impaired brainstem encoding of stimulus intensity) among F344 and LE rats exposed to JP-8 alone and JP-8 with noise.

Contributed Paper

2:40

2pNS6. The effects of military training vibration on rock art. Timothy Lavalley (LPES, Inc., 14053 Lawnes Creek Rd., Smithfield, VA 23430, tlavalley@lpesinc.com) and Jannie Loubser (Stratum UnLtd. LLC, Alpharetta, GA)

This presentation will provide an overview of potential impacts of vibration due to military training on Native American rock art. Military training is often conducted in areas near culturally sensitive landmarks or structures, and information regarding the effects of vibration on these types of structures is limited. An extensive literature review was conducted to determine the current state of information regarding vibration effects on rock art and

other cultural resources. The condition of rock art can vary from the extremely stable to fragile; and based on the existing studies, thresholds for effects were defined in term of stability class for different rock art structures. Vibrations and peak sound levels from maneuver and live fire training activities were estimated and compared to these effects thresholds. Both airborne and groundborne vibrations decreased with distance and at some point attenuate below the threshold of adverse effects. Critical distances for effects were estimated for a variety of training activities—including demolition activities, ground maneuvers, mortar fire, and rotorcraft operations. Results will be provided in both peak particle velocity and peak sound levels. These results as well as potential best management practices will be discussed.

Session 2pPA

Physical Acoustics: Acoustic Characterization of Critical Phenomena

Josh R. Gladden, Cochair

Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Veerle M. Keppens, Cochair

Univ. of Tennessee, Dept. Materials Science and Engineering, Knoxville, TN 37996

Chair's Introduction—1:10

Invited Papers

1:15

2pPA1. Variations in elastic and anelastic properties as indicators of static and dynamic strain relaxation phenomena associated with phase transitions. Michael A. Carpenter (Dept. of Earth Sci., Univ. of Cambridge, Downing St., Cambridge CB2 3EQ, United Kingdom, mc43@esc.cam.ac.uk)

Almost all phase transitions are accompanied by some degree of lattice strain, typically varying between a few per mil to a few per cent, but extending to 5–10% for martensitic transitions. It is inevitable, therefore, that there will also be changes in elastic properties and, because the elastic moduli are susceptibilities, these are typically in the range of 10's of per cent. At the same time, the associated transformation microstructures may be mobile under the action of external stress and therefore give rise to anelastic losses in a dynamical mechanical measurement. It has turned out that Resonant Ultrasound Spectroscopy, in a frequency window ~ 0.1 –2 MHz, is a particularly powerful method for characterizing elastic and anelastic behavior as functions of temperature and magnetic field for ferroic and multiferroic phase transitions in a wide range of materials. Examples of recent collaborative studies, relating to magnetoelastic effects in EuTiO_3 , multiferroic transitions in $\text{Pb}(\text{Fe}_{0.5}\text{Nb}_{0.5})\text{O}_3$, the influence of grain size on Jahn-Teller \times charge ordering in $\text{La}_{0.5}\text{Ca}_{0.5}\text{MnO}_3$, ferroelectric and ferroelastic transitions in metal organic frameworks, and martensitic transitions in Heusler alloys, will be used to demonstrate specific mechanisms and kinetics of static and dynamic strain relaxation phenomena.

1:45

2pPA2. A key test of the theory of critical phenomena using acoustics in liquid helium. Julian D. Maynard (Physics, Penn State Univ., 104 Davey Lab, Box 231, University Park, PA 16802, maynard@phys.psu.edu)

A significant success of modern theoretical physics has been in the study of critical phenomena, where a system displays singular properties while undergoing a transition between different phases of matter. The modern theory (recognized by a Wolf Prize and a Nobel Prize) involves the notions of length scale invariance, power law singularities, the renormalization group etc. A significant experimental test of the theory has been the study of the transition between normal liquid and superfluid behavior of liquid helium-4 (recognized with a London Prize). The singular properties which characterize a critical transition typically involve a divergence in a susceptibility, such as a heat capacity, compressibility, or other property which may couple to a local configuration of atoms or molecules. Since these latter properties may be probed with sound waves, acoustics can be a useful probe of critical behavior. In the case of superfluid helium there are five fundamentally different modes of sound propagation, and this acoustic abundance contributed to the success of the critical phenomenon theory test. This talk will review the theory of critical phenomena, describe the different modes of sound propagation in superfluid helium, and summarize the results of the key test of the critical phenomenon theory.

2:15

2pPA3. Structural and magnetic phase transitions in $\text{EuTi}_{1-x}\text{Nb}_x\text{O}_3$: A resonant ultrasound spectroscopy study. Ling Li (Mater. Sci. and Eng., Univ. of Tennessee, 1508 Middle Dr., Knoxville, TN 37909-2879, liling822@gmail.com), James R. Morris (BES Mater. Sci. and Eng. Program, Oak Ridge National Lab., Knoxville, TN), Michael R. Koehler (Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN), Zhiling Dun, Haidong Zhou (Phys. and Astronomy, Univ. of Tennessee, Knoxville, TN), Jiaqiang Yan, David G. Mandrus, and Veerle M. Keppens (Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN)

We have investigated the structural and magnetic phase transitions in $\text{EuTi}_{1-x}\text{Nb}_x\text{O}_3$ ($0 \leq x \leq 0.3$) with synchrotron powder X-ray diffraction (XRD), resonant ultrasound spectroscopy (RUS), and magnetization measurements. The $\text{Pm-3m} \leftrightarrow \text{I4/mcm}$ structural transition in pure and doped compounds is marked by a pronounced step-like softening of the elastic moduli near T_S , which resembles that of SrTiO_3 and can be adequately modeled using the Landau free energy model employing the same coupling between strain and octahedral tilting order parameter as previously used to model SrTiO_3 . Upon Nb doping, the cubic-to-tetragonal structural transition shifts to higher temperatures and the room temperature lattice parameter increases while the magnitude of the octahedral tilting decreases. In addition, Nb substitution for Ti destabilizes the antiferromagnetic ground state of the parent compound and long range ferromagnetic order is observed in the samples with $x \geq 0.1$.

2pPA4. Fluctuation-driven attenuation and dispersion of sound near the critical points of fluids. Keith A. Gillis and Michael R. Moldover (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov)

We discuss the extraordinary growth in the attenuation of sound α_λ and in the dispersion in the speed of sound c , which occurs near all liquid-vapor critical points. The attenuation and dispersion have been measured over 5 orders of magnitude in frequency. Remarkably, the data collapse onto universal, theoretically predicted curves. The theory considers equilibrium density fluctuations near the critical point where the fluctuations are large compared with the particle spacing. These density fluctuations have a distribution of sizes characterized by the correlation length ξ and a distribution of lifetimes characterized by the relaxation time τ . As the critical point is approached, ξ and τ diverge with the universal power laws $\xi \propto r^{-0.63}$ and $\tau \propto r^{-1.93}$, where r is a measure of the distance from the critical point. [At the critical density ρ_c , $r \equiv (T - T_c)/T_c$.] In the low-frequency limit ($\omega\tau \ll 1$), the attenuation grows as $\alpha_\lambda \propto r^{-1.93}$ and the speed of sound approaches zero as $c \propto r^{0.055}$. When $\omega\tau \gg 1$, α_λ approaches a maximum, non-universal limit and strong dispersion is present. Low-frequency sound waves reach the condition $\omega\tau = 1$ closer to T_c and deeper into the asymptotic critical regime than do high-frequency sound waves. In Earth's gravity, we show that stirring a near-critical fluid reduces stratification and enables measurements closer to the critical point. We compare the attenuation and dispersion for pure fluids near the liquid-vapor critical point with that of binary liquid mixtures near the consolute point.

Contributed Papers

3:45

2pPA5. Simple homodyne ultrasound interferometer for solid state physical acoustics. Oleksiy Svitelskiy, David Lee (Physics, Gordon College, 255 Grapevine Rd., Gordon College, Wenham, MA 01984, oleksiy.svitelskiy@gordon.edu), John Grossmann (Columbia Univ., New York, NY), Lynn A. Boatner (Oak Ridge National Lab., Oak Ridge, TN), Grace J. Yong (Towson Univ., Baltimore, MD), and Alexey Suslov (National High Magnetic Field Lab., Tallahassee, FL)

Ultrasonic pulse-echo technique is a valuable and non-destructive tool to explore elastic properties of materials. We propose a new instrument based on mass-produced microchips. In our design, the signal is processed by an AD8302 RF gain and phase detector (www.analog.com). Its phase output is linearly proportional to the phase difference between the exciting and response signals. The gain output is proportional to the log of the ratio of amplitudes of the received to the exciting signals. To exclude the non-linear fragments and to enable exploring large phase changes, we employ a parallel connection of two detectors, fed by in-phase and quadrature reference signals, respectively. The interferometer was tested by measuring the temperature dependences of both sound speed and attenuation in metallic glasses as well as in ferroelectric KTaNbO_3 (KTN) single crystals. The instrument allows for exploring phase transitions with precision of $\Delta V/V \sim 10^{-7}$ (V is ultrasound speed) in the broad dynamic range from -60 to $\times 20$ dBm. These qualities allowed us to detect the theoretically predicted, but not observed previously velocity kink at the KTN phase transition from tetragonal to orthorhombic symmetry, whereas the attenuation curve showed new features in the development of the low-temperature structure of the KTN crystal.

4:00

2pPA6. Temperature and pressure effects on elastic properties of lead magnesium niobate-lead titanate relaxorferroelectric material. Sumudu P. Tennakoon and Joseph R. Gladden (Phys. and Astronomy & NCPA, Univ. of MS, 145 Hill Dr., NCPA, Rm. 1077, University, MS 38677, sptennak@go.olemiss.edu)

Lead magnesium niobate—lead titanate (PMN-PT) exhibits exceptional electromechanical properties, considered as a highly efficient transduction

material for vibration energy harvesting and acoustic sensing applications. It is reported in the literature that the PMN-PT undergoes structural phase transitions with changes in temperature and the chemical composition. We seek to gain insight into the phase diagram of PMN-PT using temperature and pressure dependence of the elastic properties. Single crystal PMN-PT with chemical composition close to the morphotropic phase boundary (MPB) was used in a resonant ultrasound spectroscopy (RUS) study performed in the temperature range from room temperature to 773 K and the pressure range from near vacuum to 3.4 MPa. At atmospheric pressure, significantly high acoustic attenuation of the PMN-PT material is observed at the temperatures below 400 K. Strong stiffening is observed in the temperature range of 400 K–673 K, followed by a gradual softening at higher temperatures. With the varying pressure, we observed an increased pressure sensitivity of elastic properties of the PMN-PT material that can be localized to the temperature regime where the strong stiffening is observed. As time allows, the behavior of PMN-PT upon cooling below room temperature will also be discussed.

4:15

2pPA7. Acousto-optic investigation of acoustic anisotropy in paratellurite crystals. Farkhad R. Akhmedzhanov and Ulugbek Saidvaliev (Samarkand State Univ., 15 University Blvd., Samarkand 140104, Uzbekistan, farkhad2@yahoo.com)

Propagation velocity and attenuation coefficient of acoustics waves in paratellurite crystals were measured by Bragg diffraction of light in the frequency range of 0.4–1.6 GHz. These results were used for calculation of real and imaginary components of the complex tensor of elastic constants. The analysis of the anisotropy of attenuation was carried out for the acoustic waves of different polarization propagating in the crystallographic planes, which are orthogonal to symmetry axes of second and fourth order. The strongest anisotropy of acoustic attenuation and phase velocity is observed for the transverse acoustic waves propagating in the plane (1-10). It is shown that the attenuation reduces the integral efficiency of diffraction in several times in the acousto-optic deflector in which is used an oblique cut of paratellurite and the transverse acoustic wave propagating at an angle of 6 degrees to the [110] axis in the plane (1-10).

Session 2pSA**Structural Acoustics and Vibration, Noise, and Architectural Acoustics: Structural Acoustics and Vibration in Buildings**

James E. Phillips, Cochair

Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Benjamin Shafer, Cochair

*Technical Services, PABCO Gypsum, 3905 N 10th St, Tacoma, WA 98406***Chair's Introduction—1:00*****Invited Papers*****1:05****2pSA1. A preview of the second edition of the AISC Design Guide 11 “Vibrations of Steel Framed Structural Systems due to Human Activity”.** Eric E. Ungar (Acentech, 15 Considine Rd., Newton Ctr., MA 02459, eungar@acentech.com)

The Steel Design Guide 11, “Floor Vibrations due to Human Activity,” which originally was published in 1997, has been refurbished, updated, and broadened. At the time of the writing of this abstract, a draft is being reviewed by its sponsor, the American Institute of Steel Construction. Like the original, the second edition presents easily used means for predicting the vibrations of floors of steel construction due to typical walking and for assessing the acceptability of these vibrations in relation to human comfort. The new edition relies on more recent data on footfall forces and includes prediction methods (analytically derived and empirically adjusted on the basis of comparison with experimental data) for evaluating the acceptability of walking-induced vibrations on equipment and facilities whose limits are expressed in terms of a variety of measures. Analyses of stairs and pedestrian bridges are included, and modeling of structures with steel joists is discussed. An extensive chapter is devoted to the application of finite-element analysis to floor structures.

1:30**2pSA2. The relationship of vibration and groundborne/structureborne noise in buildings relative to vibration in the ground from exterior sources.** James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphilips@wiai.com)

Ground vibration generated by roadway traffic, rail, and transit systems can be transmitted into buildings where it can either be perceived as feelable vibration or as sound radiated from the interior building surfaces, i.e., floors, walls, and ceilings. The amount of vibration that is transmitted into a building is affected by a reduction as the vibration is transmitted from the ground to the foundation of the building (i.e., coupling loss) and amplification due to resonances within the building structure. This paper will discuss these effects and measurements that have been conducted to quantify the effects for common residential building construction adjacent to rail lines. A brief discussion regarding the relationship of groundborne/structureborne noise to interior vibration from exterior sources of ground vibration will be included.

1:55**2pSA3. Control of amplification of ground vibration in joist-framed buildings.** John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

For multifamily buildings near railroads, it is possible to directly measure the ground vibration caused by passing trains. The indoor vibration levels are determined by predictions of the transfer of vibration from ground to foundation, propagation up the building, and amplification by floor resonances. Large amounts of amplification have been observed in lightweight joist-framed structures. One mitigation method is to stiffen the structural system, both to raise the resonant frequency to a less objectionable range, and to reduce the magnitude of the resultant vibration. However, the benefit of this mitigation has rarely been quantified or presented. Measurements were performed in a multifamily building adjacent to railroad tracks that had been stiffened over a portion of the building. Vibration propagation and amplification were measured in the stiffened and non-stiffened portions, and the effect of mitigation is evaluated in the context of anticipated results and actual measured levels.

Contributed Papers

2:20

2pSA4. Modeling, designing, and testing low frequency impact isolation solutions for building structures with fitness centers above grade. Norman D. Varney (Kinetics Noise Control, 6300 Irelan Pl., Dublin, OH 43017, nvarney@kineticsnoise.com)

There is a trend for fitness centers to find a quick, inexpensive occupancy and ideal location in a vacant office or retail space, often above grade. The tenants sharing the building structure are then subjected to sudden airborne noise and structureborne vibration from the impact energy generated by heavy weights being dropped onto the floor. Because of this trend there is a need to analyze the large, low frequency, shock generated energies that travel through the structure. This paper will discuss the process of designing an experimental floor assembly and apparatus to test weight drops, developing a modeling program to simulate weight drops using three degrees of freedom, which can be tailored to varying isolation solutions, and the means of physically testing the results.

2:35

2pSA5. Noise control within building water supply lines. Elliott Gruber and Kenneth Cunefare (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, e_gruber@gatech.edu)

Successful treatment of fluid-borne noise in a building's plumbing system reduces noise propagation throughout the building, which improves occupant comfort and extends component lifetimes. Circulation pumps and quick-closing valves contribute to noise in water systems. Pump noise is generally untreated. Quick-closing valves may cause water hammer, which can be treated with a water hammer arrestor (WHA). Common WHA's functions by adding a compliant gaseous volume to the system; the gas volume is sealed from the system by a free piston. Control of both water-borne noise and water hammer may be achieved by a flow-through device integrating a compliant, voided polymer. The performance of current WHAs diminish over time as (1) mineral deposits degrade sealing effectiveness and (2) the gas permeates the seals; a voided polymer WHA will not suffer from these drawbacks. Prior work has demonstrated that a voided polymer is an effective source of compliance for noise control in oil systems at the anticipated pressure and temperature range of water systems. Furthermore, the acoustic impedance of oil is similar to water. Basic modeling and performance data for a prototype device will be presented.

TUESDAY AFTERNOON, 3 NOVEMBER 2015

GRAND BALLROOM 6, 1:30 P.M. TO 3:00 P.M.

Session 2pSCa

Speech Communication: Forensic Acoustics, Speaker Identification, and Voice

Robert A. Fox, Chair

Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002

Contributed Papers

1:30

2pSCa1. Reconsideration of forensic acoustics. Harry Hollien (Linguist & IASCP, Univ. of Florida, IASCP, Univ. Fla, Gainesville, FL 32611, hollien@ufl.edu)

The high talent level of in our Society members is unquestioned. However, it appears that some of us have not yet reached full maturity relative to Forensic Acoustics. While the adversarial aspect of Forensics does not parallel our investigational rigor, the number of variables they face is much greater than ours. Accordingly, development of acoustic systems for forensic use requires different approaches than we typically use. Speaker identification (SI) constitutes a good example here. Fifty years ago, it was agreed that a valid SI system could be quickly created but that has not happened. In response, the undersigned adapted the scientific method (and its standards) as a model for SI system development. The resulting research was modestly successful but the approach was not considered "useful". Later, the Daubert test confirmed this model (but ordinarily was circumvented). Now, the National Academy of Science has verified these models and is insisting on proper developmental protocol. It is no longer permissible to develop an algorithm-based device, and apply it, without proper demonstration of its validity. The impact of these research-based standards—now required for creating Forensic systems—will be discussed, as will possible responses.

1:45

2pSCa2. Voice disguise and speaker identification. Grace S. Didla (Post Doctoral Researcher, Dept. of Linguist, Univ. of Florida, Gainesville, FL 32611, didlagrace@gmail.com) and Dr. Harry Hollien (Professor Emeritus, Linguist, Univ. of Florida, Gainesville, FL)

Voice disguise involves deliberately changing one's voice to conceal their identity. It is most often associated with crimes such as kidnapping, fraud, etc. In addition to being a general problem in the world of crime, it is a particular one in the area of Speaker Identification (SI) and for the following reasons: (1) There appears to be no limit on human ingenuity in the types of disguise that can be employed. (2) While there are unlimited ways to disguise speech, there seem to be far too few studies to provide useful information about it. (3) Further these studies are narrow in scope. This can be attributed to the extensive number of variables (type of disguise, familiarity with the speaker, language, dialect, etc.) and the need for their control. (4) Because of these limitations, a great deal more needs to be known about the major types of disguise and the ones detrimental to SI. Thus, the numerous issues plaguing voice disguise must be addressed systematically through an integrated research program. A review of their nature, the research needed and the reasons for conducting the program will be discussed. Data from a sample experiment of this type will be presented.

2:00

2pSCa3. Vibrational and acoustic consequences of changes in subglottal pressure, vocal fold stiffness, vocal fold approximation, and vocal fold thickness. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Using a three-dimensional continuum model of phonation, this study investigates the effects of changes in subglottal pressure, vocal fold approximation and stiffening, and vocal fold medial surface thickness on vocal fold vibration and acoustics. The results show that increasing subglottal pressure leads to more or less uniform increase in harmonic energy across a very large frequency range as well as significant increase in noise production. Reduced noise production can be achieved by increasing medial surface thickness, vocal fold approximation, or vocal fold stiffening. Increasing vocal fold thickness and stiffness also lead to increased production of high-frequency harmonics. The closed quotient of vocal fold vibration depends primarily on the medial surface thickness of the vocal fold, with the closed quotient increasing with increasing thickness. The closed quotient also slightly decreases with increasing subglottal pressure, but increases with increasing vocal fold approximation and stiffening. These results suggest that, in addition to increasing subglottal pressure, vocal loudness can be also increased by increasing vocal fold thickness, approximation and/or stiffening to increase production of higher-order harmonics and reduce noise production, which leads to a perceived increase in vocal intensity. [Work supported by NIH.]

2:15

2pSCa4. Behavioral and computational estimates of breathiness and roughness over a wide range of dysphonic severity. David A. Eddins, Arianna Vera-Rodriguez, Mark D. Skowronski (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu), and Rahul Shrivastav (Univ. of Georgia, Athens, GA)

Perceptual evaluations of dysphonic voices frequently involve evaluation of the breathy and rough qualities. It is important to know how well one can disambiguate such VQ percepts in a perceptual evaluation so that accurate assessments can be made, focused treatment targets can be set, and outcomes can be quantified. In this study, 10 voices that varied over a wide range of breathiness and 10 voices that varied over wide range in roughness were selected from the University of Florida Disordered Voice Database (UFDVD). A single-variable matching task designed specifically for breathiness or roughness evaluation was used to index the perceived breathiness and roughness of the set of 20 stimuli. Because computational estimates of pitch salience (pitch strength) have been strongly associated with perceived breathiness, we also included a perceptual evaluation of pitch strength using an anchored magnitude estimation task. In an effort to better understand the interactions among perceived roughness and breathiness, a series of computational algorithms was used to estimate perceived breathiness and perceived roughness. Together, these results will help us understand the natural covariance of voice qualities and the ability to evaluate separate voice qualities independently.

2:30

2pSCa5. The acoustics of perceived creaky voice in American English. Sameer ud Dowla Khan, Kara Becker (Linguist, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, sameerudowlakhan@gmail.com), and Lal Zimman (Linguist, Univ. of California Santa Barbara, Santa Barbara, CA)

We compared auditory impressions of creaky voice in English to acoustic measures identified as correlates of contrastive voice qualities in other languages (e.g., Khmer, Chong, Zapotec, Gujarati, Hmong, Trique, and Yi). Sixteen trained linguistics undergraduates listened to the IP-final word 'bows' produced five times each by five American English speakers reading the Rainbow Passage, and gave a rating from 0 (no creak) to 5 (very creaky). Results show that stronger auditory impressions of creak are significantly correlated with lower f_0 , lower cepstral peak prominence (CPP), lower harmonics-to-noise ratios (HNR), and higher subharmonics-to-harmonics ratio (SHR). This suggests that listeners perceive greater creakiness as the voice becomes lower pitched, less periodic, and more audibly interspersed with subharmonic frequencies (i.e., diplophonia). Notably, none of the spectral amplitude measures proposed as acoustic correlates of glottal configurations for creaky voice in other languages (e.g., lower H1-H2 for smaller open quotient, lower H1-A1 for smaller posterior aperture, lower H1-A3 for more abrupt closure, etc.) was significantly correlated with these judgments in any expected direction. Taken together, these results suggest that while listeners consistently use pitch and periodicity as cues to creak, speakers might be varying in their articulatory strategies to achieve those acoustic effects.

2:45

2pSCa6. Evaluating acoustic measurements of creaky voice: A Vietnamese case study. Nadya Pincus, Angeliki Athanasopoulou, Taylor L. Miller, and Irene Vogel (Linguist & Cognit. Sci., Univ. of Delaware, 125 E Main St., Newark, DE 19716, npincus@udel.edu)

It has been proposed that there are two broad categories of creaky voice (CV), laryngealized and aperiodic. Moreover, several subdivisions have been proposed for both categories (Keating & Garellek, 2015), and various combinations of acoustic properties have been associated with each. It remains unclear, however, how to determine which type of CV a language has and which acoustic measurements to rely on. We address this problem with two rising tones in Vietnamese differing in phonation. All of the phonation measurements we tested with ANOVA were statistically significant ($p < .01$) in the distinction between the two tones, and thus inconclusive as to the type of CV. We propose that an additional binary logistic regression analysis be applied to the various measurements to determine the extent to which each one contributes to classifying creaky vs. modal vowels; and this, in turn, can inform us about the nature of the CV in the language. Specifically in Vietnamese, we found that HNR yields the strongest classification result (84%); the others were closer to chance (58–68%). We can thus conclude that the CV used in Vietnamese is the aperiodic type, as evidenced by the role of irregular F_0 as opposed to the other phonation properties.

2p TUE. PM

Session 2pSCb

Speech Communication: Speech Perception Potpourri (Poster Session)

Tuuli Morrill, Chair

Michigan State University, 4400 University Drive, 3E4, Fairfax, VA 22030

Authors will be at their posters from 3:30 p.m. to 5:00 p.m. To allow authors an opportunity to see other posters in their session, all posters will be on display from 1:00 p.m. to 5:00 p.m.

Contributed Papers

2pSCb1. Detailing vowel development in infancy using cortical auditory evoked potentials and multidimensional scaling. Kathleen McCarthy (Speech, Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, kathleen.mccarthy@ucl.ac.uk), Katrin Skoruppa (Universität Basel, Basel, Switzerland), and Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, London, United Kingdom)

The present study used an efficient measure of perceptual sensitivity to map perception across the British English vowel space for 80 monolingual English infants (4–5, 8–9, and 10–11 months old). Auditory evoked potentials were measured for spectral changes between concatenated vowels, which, for infants, typically evokes a positivity about 150–200 ms after each spectral change. These were measured for 28 pairs of seven monophthongal vowels (/i/, /ɪ/, /ɛ/, /a/, /ɑ/, /ɔ/, /u/) that were presented in a random concatenated sequence with changes every 300–400 ms. ERPs were averaged across epochs following each spectral change, with the magnitude of the response for each vowel pair used as similarity measure for multidimensional scaling. The 4–5 month old infants had two-dimensional perceptual maps that closely matched the F1 and F2 acoustic differences between vowels. In contrast, the older infant response was less related to acoustic differences and they had selectively larger responses for neighbors around the vowel quadrilateral (e.g., /i/-/ɪ/); suggesting a shift to a more phonologically driven processing. These results provide a more detailed picture of phonetic development than has been shown before, and demonstrate an efficient procedure to map speech processing in infancy.

2pSCb2. Speech perception capabilities in children several years after initial diagnosis of auditory processing disorders. Rachel Crum (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, 125 East Main St., Newark, DE 19716, rachelc@udel.edu), Jennifer Padilla (Dept. of Psychol. and Brain Sci., Univ. of Delaware, Newark, DE), Thierry Morlet (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE), L. A. Greenwood (Pediatrix Audiol. Services, Falls Church, VA), Jessica Loson, Sarah Zavala (Audiol., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE), and Kyoko Nagao (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE)

This study focuses on the progress of speech perception capabilities in children with auditory processing disorder (APD) measured several years after the initial diagnosis. A recent longitudinal study showed listening and communication difficulties of children diagnosed with APD persist into adulthood (Del Zoppo, Sanchez, and Lind, 2015). In the current study, we examined the speech perception progress of 21 children selected from the Auditory Processing Disorder database of 255 school-aged children in the Mid-Atlantic region as having an initial diagnosis of APD and an APD reassessment at our facility several years later. Average age for initial assessment was 7.9 (7–10) years with the most recent APD evaluation ages 11.5 (9–16) years. Results show that 81% of the children still had auditory processing deficits in their most recent evaluations, 9% had only associated area

issues, and 10% exhibited typical auditory processing performance. We found that average standardized scores for all tests, except auditory figure ground, collectively increased in the second assessment. These tests include: competing words, competing sentences, filtered words, composite scores, and phonemic synthesis. While the auditory processing skills of some children with APD are improving over time, some children still show impairment in several processing areas.

2pSCb3. Adults' perceptual voicing boundaries of 2-year-olds' citation form speech. Elaine R. Hitchcock (Commun. Sci. and Disord., Montclair State Univ., 116 West End Ave., Pompton Plains, NJ 07444, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Haskins Labs., New Haven, CT)

One way that speakers distinguish between phonemic categories is through voicing, frequently measured using voice onset time (VOT). Much perceptual research on voicing identification and discrimination has used synthetic speech stimuli varying in VOT. Results from adult listeners typically show stable crossover regions in the 20–35 ms range. Subsequent work, however, reveals that listeners' VOT boundaries vary with speech rate; further, an extensive history of research into vowel perception indicates that listeners normalize across vocal tract sizes. These considerations lead to the possibility that adult voicing boundaries may differ between the speech of adults vs. children, since children have slower speech rates and smaller vocal tracts. The present study obtained adult discrimination data for natural productions of bilabial and alveolar cognate pairs produced by 2–3-year-old monolingual, English-speaking children. Randomized stimuli were presented twice to 20 listeners resulting in 4,000 rated stimuli per category. The findings show 50% crossover points for VOT values at 28 ms for bilabials and 32 ms for alveolar phonemes. Such outcomes are consistent with past work based on adult data and suggest that mature listeners do not use substantially different perceptual criteria for judging voicing in children's speech. **Declaration of Interest Statement** The authors have no financial or nonfinancial disclosures to report.

2pSCb4. Top-down influences in perception with spectrally degraded resynthesized natural speech. Jane Smart and Stefan Frisch (Commun. Sci. and Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD 1017, Tampa, FL 33620, jbsmart@mail.usf.edu)

Lexical processes can influence perception of ambiguous phonemes (e.g., Ganong, 1980). To date, research has focused on these influences in quiet conditions with stimuli that have not been degraded. This project examines the interplay between lexical and acoustic information in speech perception, with stimuli in non-degraded and spectrally degraded conditions. Continua of /t, k/ onsets were developed by wavelet resynthesis of natural speech using TandemSTRAIGHT software, concatenated to the vowel × coda portion of words/nonwords, and distorted using AngelSim (TigerCIS) vocoding software. Normal hearing adult participants identified the onset phonemes in non-degraded and spectrally degraded conditions. Identification functions and the effects of lexical status in phoneme perception with spectrally degraded stimuli will be discussed.

2pSCb5. Brain activity for sound symbolism in visual size judgment: Combinations of voiced and voiceless plosives with a vowel “o” or “i”. Sachi Itagaki, Shota Murai, Shizuko Hiryu, Kohta I. Kobayasi (Graduate School of Life and Medical Sci., Doshisha, 1-3, Tatara Miyakodani, Kyotanabe, Kyoto 6100394, Japan, dmp1006@mail4.doshisha.ac.jp), Jan Auracher (Konan, Kobe, Japan), and Hiroshi Riquimaroux (Graduate school of Life and Medical Sci., Doshisha, Kyoto, Japan)

Sound symbolism is an idea that sounds itself has the impression. In most of the previous psychology and linguistics researches, stimuli were presented visually with alphabets, and subjects directly answered the impression of the sound. The purpose of this study is that establishing a behavioral paradigm applicable to functional magnetic resonance imaging (fMRI) research when the sound stimulus was presented aurally. In this experiment, we focused on sound symbolism in visual size. Subjects were required to answer visual size difference between standard and target stimulus. Visual stimuli were a gray circle on black background LCD screen. Sound stimuli were /bobo/ and /pipi/, and were assumed to have impression of “bigger” and “smaller,” respectively, according to previous researches. Currently, brain activity of the sound symbolism is examined in MRI using our previous behavior paradigm. The result suggested that our paradigm is able to use for fMRI study. However, MRI imaging results shows that the location and amount of activity is different by subjects, suggesting the neural substrate of the sound symbolism could vary between individuals. Relationships between the brain differences and individual behavioral differences will be discussed.

2pSCb6. Perceptual load of divided attention modulates auditory-motor integration of voice control. Hanjun Liu, Ying Liu, and Zhiqiang Guo (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

The present study sought to examine how the auditory-motor processing of feedback errors during vocal pitch regulation is modulated as a function of divided attention. Subjects were exposed to pitch perturbations in voice auditory feedback and flashing lights on the computer screen, during which they were asked to divide their attention to both auditory and visual stimuli by counting the number of both of them. The presentation rate of the visual stimuli (inter-stimulus interval) was manipulated to produce a low, intermediate, and high attentional load. The results revealed that, as compared to the low and intermediate attentional load conditions, the high attentional load condition elicited significantly smaller magnitudes of vocal responses but larger amplitudes of N1 and P2 responses to pitch feedback perturbations. These findings demonstrate the effect of attentional load on the auditory-motor processing of pitch feedback errors during divided attention, suggesting that perceptual load of visual attention interferes the attentional modulation of auditory-motor integration in voice control.

2pSCb7. Speaker identification using auditory modeling and vector quantization. Konstantina Iliadi and Stefan Bleack (Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, konstiliadi@gmail.com)

Speaker identification (SID) aims to identify the underlying speaker(s) given a speech utterance. SID systems can perform well under matched training and test conditions but their performance degrades significantly because of the mismatch caused by background noise in real-world environments. Achieving robustness to the SID systems depends very much on the front-end (or feature extractor), which is the first component in an automatic speaker recognition system. Feature extraction transforms the speech signal into a compact representation that is more discriminative than the original signal. We present on our poster a new system where the parametrization of the speech is based on an auditory model called Auditory Image Model (AIM). Two experiments were performed for two different sets of speakers. Experiment 1 identified the most informative regions of the auditory image that can indicate speaker recognition. Experiment 2 consisted of training 10 and 60 speakers using clean speech and testing those two groups using speech in the presence of babble noise of eight speakers for 5 SNRs. The results suggest that the extracted auditory feature vectors led to much better performance, i.e., higher SID accuracy, compared to the MFCC-based recognition system especially for low SNRs.

2pSCb8. Digit recognition with phonological features. Vipul Arora, Aditi Lahiri (Faculty of Linguist, Philology & Phonet., Univ. of Oxford, University of Oxford, Oxford OX1 2HG, United Kingdom, vipul.arora@ling-phil.ox.ac.uk), and Henning Reetz (Dpt. of Empirical Linguist, Goethe-Univ. Frankfurt, Frankfurt, Germany)

Our project aims to identify English digits, employing an approach to speech recognition built on principles derived from phonological knowledge and neurolinguistic experiments on how the humans perceive and process speech. This is in contrast to current ASR systems, which rely upon statistical machine learning using thousands of hours of speech training data and vast amount of computations, with little reliance on any aspect of phonological features. We focus on digits to test our approach with a circumscribed set of elements, which have considerably different phonemes (e.g., the sounds in ‘five’, ‘two’ and ‘six’ are all different). Our system converts the acoustic signal to a set of phonological features whose combinations are used to access words. The features are speaker and language independent, with the intention of building a system easily adaptable to other languages without re-training the acoustic front-end. A number of acoustic parameters (e.g. LPC, spectral and energy differences) were investigated and their relative importance compared for a robust estimation of phonological features; e.g., spectral slope below and above 2500 Hz disambiguates sibilants. However, distinguishing sonorant sounds requires a range of other parameters. We tested our system on the American English TIDIGIT database achieving unexpected success rates.

2pSCb9. Links between the perception of speaker age and sex in children’s voices. Peter F. Assmann, Michelle R. Kapolowicz, David A. Massey (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), Santiago Barreda (Dept. of Linguist, Univ. of California, Davis, Davis, CA), and Terrence M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

At a recent meeting [Assmann *et al.*, *J. Acoust. Soc. Am.* **135**, 2424 (2014)] we reported two experiments on the perception of speaker age and sex in children’s voices, along with two models to predict listeners’ judgments. The stimuli were vocoded /hVd/ syllables produced by 140 speakers, ages 5 through 18, processed to simulate a change in the sex of the speaker. Experimental conditions involved swapping the fundamental frequency (F0) contour and/or the formant frequencies (FF) to the opposite-sex average within each age group. The present study extended the original experiments by requiring each listener to judge both age and sex on each trial to investigate the relationship between the two perceptual responses. Results revealed that age estimation error is systematically linked to sex misclassification, particularly in older children. In the unswapped condition, age estimates tended to be lower if the voice was identified as male, relative to the same voice heard as female. The condition with both F0 and FF swapped approached the opposite pattern of results; however, the remaining discrepancy indicates these are not the only cues for the perception of age and sex in children’s voices.

2pSCb10. Understanding southern British, Glaswegian, and Spanish English accents: Speech in noise and evoked potentials. Petra Hödl, Melanie Pinet, Bronwen Evans, and Paul Iverson (Univ. College London, University College London, 2 Wakefield St, London, United Kingdom, p.hodl@ucl.ac.uk)

Speech recognition in noise is affected by the accents of the speakers and listeners, but it is not clear how overall accuracy is linked to the underlying perceptual and lexical processes. The present study investigated speech recognition for two native-accent groups (Southern British English and Glaswegian) and one non-native group (Spanish learners of English). Listeners were tested behaviorally on speech-recognition in noise, and using EEG measures of vowel perception (cortical evoked potentials to vowel spectra change) and lexical processing (N400). As expected, southern British English listeners were most accurate for southern British speech, Glaswegians were accurate for both Glaswegian and southern British English speech, and Spanish speakers had particular difficulty with Glaswegian. The EEG results demonstrated differences between groups in terms of both vowel and lexical processing. In particular, Glaswegian listeners differed in

their lexical processing for the two native accents despite having similar speech-in-noise accuracy, and Spanish speakers appeared to use contextual information less than the other two groups. The results begin to demonstrate how differences at a perceptual level can be compensated for during lexical processing, in ways that are not apparent purely from recognition accuracy scores.

2pSCb11. Effect of menstrual phase on dichotic listening. Richard J. Morris and Alissa N. Smith (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu)

Women using birth control and those not using it completing weekly dichotic listening sessions for nine weeks. Results indicated no differences between the two groups for right ear advantage.

2pSCb12. Listener characteristics and the perception of speech in noise. Noah Silbert and Lina Motlagh Zadeh (Commun. Sci. and Disord., Univ. of Cincinnati, 3239 Bishop St. Apt #4, Cincinnati, OH 45220, motlagla@mail.uc.edu)

Speech communication is often made difficult by the presence of background noise. Much research on the perception of noise-masked speech has focused on the masking of phonetic information by different types of noise (e.g., white noise, speech-shaped noise, temporally modulated noise, multi-talker babble). The present work focuses on the relationships between some cognitive characteristics of listeners and accuracy in the identification of noise-masked consonants. 37 listeners identified numerous tokens of each of 4 consonants (p, b, f, v) in CV syllables produced by 8 talkers (4 male, 4 female) masked by 10-talker babble. Listeners also completed a number of tasks designed to measure selective attention: two dichotic listening tasks and two non-speech discrimination tasks. On each trial of the dichotic listening tasks, one or the other ear was cued visually (i.e., “right ear” or “left

ear”), after which the listener indicated the talker sex or the consonant in the target ear, depending on task. In the two non-speech tasks, listeners discriminated either the frequency or the duration of broadband target noise bursts embedded in temporally modulated background noise. Analyses indicate a positive relationship between noise-masked speech accuracy and performance on the dichotic consonant identification and complex non-speech discrimination tasks.

2pSCb13. The effect of semantic cues on intelligibility: A comparison between spectrally sparse speech and natural speech in noise. Bahar Shahsavarani, Thomas Carrell (Commun. Disord., Univ. of Nebraska - Lincoln, 352 Barkley Memorial Ctr., Lincoln, NE 68583, sshahsav@cse.unl.edu), and Ashok Samal (Comput. Sci., Univ. of Nebraska - Lincoln, Lincoln, NE)

Listeners are known to make use of contextual cues when perceiving speech. Using contextual information is even more important when listening to distorted or difficult speech signals. For example, listeners benefit from contextual cues to compensate for the absence of fine acoustic-phonetic information when listening to natural speech in noise. In the present investigation, the effect of context was compared between four- and eight-channel spectrally degraded speech (Shannon *et al.*, 1995) versus natural speech-in-noise (0 dB SNR). Spectrally degraded speech simulates the primary information transmitted by cochlear implant devices. The results demonstrated that eight-band signals received benefit from context to the same extent as natural speech in noise. In contrast, four-band signals provided significantly less benefit from context than natural speech in noise. The most parsimonious explanation of this pattern of results is that listeners need a threshold amount of acoustic information to make use of context equally, regardless of the type of distortion. Alternatively, the results could be explained by assuming that listeners employ different strategies depending on the overall acoustic characteristics of the speech signal. Future experiments will distinguish between these alternatives.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

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, 3 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Orlando
Acoustical Oceanography	7:30 p.m.	River Terrace 2
Animal Bioacoustics	7:30 p.m.	City Terrace 9
Architectural Acoustics	7:30 p.m.	Grand Ballroom 3
Physical Acoustics	7:30 p.m.	St. Johns
Psychological and Physiological Acoustics	7:30 p.m.	Grand Ballroom 7
Structural Acoustics and Vibration	7:30 p.m.	Daytona

Committees meeting on Wednesday, 4 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Clearwater
Signal Processing in Acoustics	8:00 p.m.	City Terrace 7

Committees meeting on Thursday, 5 November

Committee	Start Time	Room
Musical Acoustics	7:30 p.m.	Grand Ballroom 1
Noise	7:30 p.m.	Grand Ballroom 2
Underwater Acoustics	7:30 p.m.	River Terrace 2

2p TUE. PM