Session 4aAAa

Architectural Acoustics, Speech Communication, and Noise: Room Acoustics Effects on Speech Comprehension and Recall I

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

David H. Griesinger, Cochair
Research, David Griesinger Acoustics, 221 Mt Auburn St #504, Cambridge, MA 02138

Chair's Introduction—8:40

Invited Papers

8:45

4aAAa1. Speech recognition in adverse conditions. Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

Speech recognition is highly sensitive to adverse conditions at all stages of the speech chain, i.e., the sequence of events that transmits a message from the mind/brain of a speaker through the acoustic medium to the mind/brain of a listener. Adverse conditions can originate from source degradations (e.g., disordered or foreign-accented speech), environmental disturbances (e.g., background sounds with or without energetic masking), and/or receiver (i.e., listener) limitations (e.g., impaired or incomplete language models, peripheral deficiencies, or tasks with high cognitive load). (For more on this classification system, see Mattys, Davis, Bradlow, & Scott, 2012, Language and Cognitive Processes, 27). This talk will present a series of studies focused on linguistic aspects of these various possible sources of adverse conditions for speech recognition. In particular, we will demonstrate separate and combined influences of the talker's language background (a possible source degradation), the presence of a background speech masker in either the same or a different language from that of the target speech (a possible environmental degradation), and the listener's experience with the language of the target and/or masking speech (a possible receiver limitation). Together, these studies demonstrate strong influences of language and linguistic experience on speech recognition in adverse conditions.

9:05

4aAAa2. Speech intelligibility and sentence recognition memory in noise. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu)

Much of daily communication occurs in adverse conditions impacting various levels of speech processing negatively. These adverse conditions may originate in talker- (fast, reduced speech), signal- (noise or degraded target signal), and listener- (impeded access or decoding of the target speech signal) oriented limitations, and may have consequences for perceptual processes, representations, attention, and memory functions (see Mattys *et al.*, 2012 for a review). In this talk, I first discuss a set of experiments that explore the extent to which listener-oriented clear speech and speech produced in response to noise (noise-adapted speech) by children, young adults and older adults contribute to enhanced word recognition in challenging listening conditions. Next, I discuss whether intelligibility-enhancing speaking style modifications impact speech processing beyond word recognition, namely recognition memory for sentences. The results show that effortful speech processing in challenging listening environments can be improved by speaking style adaptations on the part of the talker. In addition to enhanced intelligibility, a substantial improvement in sentence recognition memory can be achieved through speaker adaptations to the environment and to the listener when in adverse conditions. These results have implications for the quality of speech communication in a variety of environments, such as classrooms and hospitals.

9:25

4aAAa3. Reducing cognitive demands on listeners by speaking clearly in noisy places. Kristin Van Engen (Psych., Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu)

Listeners have more difficulty identifying spoken words in noisy environments when those words have many phonological neighbors (i.e., similar-sounding words in the lexicon) than when they have few phonological neighbors. This difficulty appears to be exacerbated in old age, where reductions in inhibitory control presumably make it more difficult to cope with competition from similar-sounding words. Fortunately, word recognition in noise can generally be improved for a wide range of listeners (e.g., younger and older adults, individuals with and without hearing impairment) when speakers adopt a clear speaking style. This study investigated whether clear speech, in addition to generally increasing speech intelligibility, also reduces the inhibitory demands associated with identifying lexically difficult words in noise for younger and older adults. The results show that, indeed, the difference between rates of identification

for words with many versus few neighbors was eliminated when those words were produced in clear speech. Data on the roles of individual differences (e.g., hearing, working memory, and inhibitory control) that may contribute to word identification in noise will also be presented.

9:45

4aAAa4. Improved speech understanding and amplitude modulation sensitivity in rooms: Wait a second!. Pavel Zahorik (Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville School of Medicine, Psychol. and Brain Sci., Life Sci. Bldg. 317, Louisville, KY 40292, pavel.zahorik@louisville.edu), Paul W. Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), Eugene Brandewie (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), and Nirmal K. Srinivasan (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., Portland, OR)

Sound transmission between source and receiver can be profoundly affected by room acoustics, yet under many circumstances, these acoustical effects have relatively minor perceptual consequences. This may be explained, in part, by listener adaptation to the acoustics of the listening environment. Here, evidence that room adaptation improves speech understanding is summarized. The adaptation is rapid (around 1 s), and observable for a variety of speech materials. It also appears to depend critically on the amplitude modulation characteristic of the signal reaching the ear, and as a result, similar room adaptation effects have been observed for measurements of amplitude modulation sensitivity. A better understanding of room adaptation effects will hopefully contribute to improved methods for speech transmission in rooms for both normally hearing and hearing-impaired listeners. [Work supported by NIDCD.]

10:05–10:20 Break

10:20

4aAAa5. The importance of attention, localization, and source separation to speech cognition and recall. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Acoustic standards for speech are based on word recognition. But for successful communication sound must be detected, separated from noise and other streams, phones, syllables, and words must be recognized and parsed into sentences, meaning must be found by relating the sentences to previous knowledge, and finally information must be stored in long term memory. All of these tasks require time and working memory. Acoustical conditions that increases the difficulty of any part of the task reduce recall. But attention is possibly the most important factor in successful communication. There is compelling anecdotal evidence that sound profoundly and involuntarily influences attention. Humans detect in fractions of a second whether a sound source is close, independent of its loudness and frequency content. When sound is perceived as close it demands a degree of attention that distant sound does not. The mechanism of detection relies on the phase relationships between harmonics of complex tones in the vocal formant range, properties of sound that also ease word recognition and source separation. We will present the physics of this process and the acoustic properties that enable it. Our goal is to increase attention and recall in venues of all types.

10:40

4aAAa6. Release from masking in simulated reverberant environments. Nirmal Kumar Srinivasan, Frederick J. Gallun, Sean D. Kampel, Kasey M. Jakien, Samuel Gordon, and Megan Stansell (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, nirmal.srinivasan@va.gov)

It is well documented that older listeners have more difficulty in understanding speech in complex listening environments. In two separate experiments, speech intelligibility enhancement due to prior exposure to listening environment and spatial release from masking (SRM) for small spatial separations were measured in simulated reverberant listening environments. Release from masking was measured by comparing threshold target-to-masker ratios (TMR) obtained with a speech target presented directly ahead of the listener and two speech maskers presented from the same location or in symmetrically displaced spatial configurations in an anechoic chamber. The results indicated that older listeners required much higher TMR at threshold and obtained decreased benefit from prior exposure to listening environments compared to younger listeners. For the small separation experiment, speech stimuli were presented over headphones and virtual acoustic techniques were used to simulate very small spatial separations (approx. 2 degrees) between target and maskers. Results reveal, for the first time, the minimum separation required between target and masker to achieve release from speech-on-speech masking in anechoic and reverberant conditions. The advantages of including small separations for understanding the functions relating spatial separation to release from masking will be discussed, as well as the value of including older listeners. [Work supported by NIH R01 DC011828.]

11:00

4aAAa7. Speech-on-speech masking for children and adults. Lauren Calandruccio, Lori J. Leibold (Allied Health Sci., Univ. of North Carolina, 301 S. Columbia St., Chapel Hill, NC 27599, Lauren_Calandruccio@med.unc.edu), and Emily Buss (Otolaryngology/Head and Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Children experience greater difficulty understanding speech in noise compared to adults. This age effect is pronounced when the noise causes both energetic and informational masking, for example, when listening to speech while other people are talking. As children acquire speech and language, they are faced with multi-speech environments all the time, for example, in the classroom. For adults, speech perception tends to be worse when the target and masker are matched in terms of talker sex and language, with mismatches improving performance. It is unknown, however, whether children are able to benefit from these (sex or language) target/masker mismatches. The goal of this project is to further our understanding of the speech-on-speech masking deficit children demonstrate throughout childhood, while specifically investigating whether children's speech recognition improves when the target and masker are spoken by talkers of the opposite sex, or when the target and masker speech are spoken in different languages. Normal-hearing children and adults were tested on word identification and sentence recognition tasks. Differences in SNR needed to equate performance between the two groups will be reported, as well as data reporting whether children are able to benefit from these target/masker mismatch cues.

11:20

4aAAa8. The neural basis of informational and energetic masking effects in the perception and production of speech. Samuel Evans (Inst. of Cognit. Neurosci., Univ. College London, 17 Queen Square, London, London WC1N 3AR, United Kingdom, samuel.evans@ucl.ac.uk), Carolyn McGettigan (Dept. of Psych., Royal Holloway, Egham, United Kingdom), Zarinah Agnew (Dept. of Otolaryngol., Univ. of California, San Francisco, San Francisco, CA), Stuart Rosen (Dept. of Speech, Hearing and Phonetic Sci., Univ. College London, London, United Kingdom), Lima Cesar (Ctr. for Psych., Univ. of Porto, Porto, Portugal), Dana Boebinger, Markus Ostarek, Sinead H. Chen, Angela Richards, Sophie Meekings, and Sophie K. Scott (Inst. of Cognit. Neurosci., Univ. College London, London, United Kingdom)

When we have spoken conversations, it is usually in the context of competing sounds within our environment. Speech can be masked by many different kinds of sounds, for example, machinery noise and the speech of others, and these different sounds place differing demands on cognitive resources. In this talk, I will present data from a series of functional magnetic resonance imaging (fMRI) studies in which the informational properties of background sounds have been manipulated to make them more or less similar to speech. I will demonstrate the neural effects associated with speaking over and listening to these sounds, and demonstrate how in perception these effects are modulated by the age of the listener. The results will be interpreted within a framework of auditory processing developed from primate neurophysiology and human functional imaging work (Rauschecker and Scott 2009).

THURSDAY MORNING, 30 OCTOBER 2014 SANTA FE, 10:35 A.M. TO 12:05 P.M.

Session 4aAAb

Architectural Acoustics: Uses, Measurements, and Advancements in the Use of Diffusion and Scattering Devices

David T. Bradley, Chair
Physics Astronomy, Vassar College, Poughkeepsie, NY 12604

Chair's Introduction—10:35

Invited Papers

10:40

4aAAb1. Effect of installed diffusers on sound field diffusivity in a real-world classroom. Ariana Sharma, David T. Bradley, and Mohammed Abdelaziz (Phys. + Astronomy, Vassar College, 124 Raymond Ave, Poughkeepsie, NY 12604, arsharma@vassar.edu)

An ideal diffuse sound field is both homogeneous (acoustic quantities are independent of position) and isotropic (acoustic quantities are invariant with respect to direction). Predicting and characterizing sound field diffusivity is essential to acousticians when designing and using acoustically sensitive spaces. Surfaces with a non-planar geometry, referred to as diffusers, can be installed in these spaces as a means of increasing and/or controlling the field diffusivity. Although some theoretical and computational modeling work has been carried out to better understand the relationship between these installed diffusers and the resulting field diffusivity, the current state-of-the-art does not include a systematic understanding of this relationship. Furthermore, very little work has been done to characterize this relationship in full scale and in the real world. In the current project, the effect of diffusers on field diffusivity has been studied in a full scale, real-world classroom. Field diffusivity has been measured for various configurations of the diffusers using two measurement techniques. The first technique uses a three-dimensional grid of receivers to characterize the field homogeneity. To characterize field isotropy, a spherical microphone array has also been used. Results and analysis will be presented and discussed.

11:00

4aAAb2. Effect of measurement conditions on sound scattered from a pyramid diffuser in a free field. Kimberly A. Riegel, David T. Bradley, Mallory Morgan, and Ian Kowalok (Phys. + Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604, kiriegel@vassar.edu)

A surface with a non-planar geometry, referred to as a diffuser, can be used in acoustically sensitive spaces to help control or eliminate unwanted effects from strong reflections by scattering the reflected sound. The scattering behavior of a diffuser can be measured in a free field, according to the standard ISO 17497-2. Many of the measurement conditions discussed in this standard can have an effect on the measured data; however, these conditions are often not well-specified and/or have not been substantiated. In the current study, a simple pyramid diffuser has been measured while varying several measurement conditions: surface material, orientation of the surface geometry, perimeter shape of the surface, and mounting depth of the surface. Reflected polar response and diffusion coefficient data have been collected and compared for each condition. Data have also been contrasted with those obtained by numerical simulation using boundary element method (BEM) techniques for an idealized pyramid diffuser. Results and analysis will be presented and discussed.

Contributed Papers

11:20

4aAAb3. Sound field diffusion by number of peak by continuous wavelet transform. Yongwon Cha, Muhammad Imran, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Hanyang University, Seoul 133-791, South Korea, chadyongwoncha@gmail.com)

The number of peak (Np) in the impulse response signal (IRs) captured for the real hall have been investigated and measured by using continuous wavelet transform (CWT). Np has a relationship with perceptual diffusion as an objective characteristic that is influenced by walls scattering elements. In addition, as measuring diffuse sound fields, the CWT coefficients are used for detecting the diffusive sound. Based on the absolute coefficient values calculated from CWT analysis, a practical method of counting reflections is considered. These reflections are specified as diffusive or specular base on their similarity with the mother wavelet. Temporal and spatial representation of absolute values of CWT is presented. Auditory experiments using a paired comparison method were conducted to gauge the relationship between the Np and perceptual sound field diffusion. It is revealed that a dominant factor influencing the subjective preference in the hall was the Np that varied with different wall surface treatments.

11:35

4aAAb4. In praise of smooth surfaces: Promoting a balance between specular and diffuse surfaces in performance space design. Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 53 W. Jackson Boulevard, Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com)

Diffusive surfaces are often presented as a panacea for achieving desirable listening conditions in performance spaces. While diffusive surfaces are a valuable and necessary part of the finish palette in any theater or concert hall, a significant number of specular surfaces are crucial to the success of many such spaces. Case studies will be presented in which excessive use of diffusion has resulted in losses of clarity and loudness, including comparisons to the results following the introduction of specular surfaces, either flat or gently curved. Aural examples will be presented to demonstrate the perceptual differences when specular surfaces are employed as compared to highly diffusive surfaces at key locations in spaces for music and drama.

11:50

4aAAb5. Scattershot: A look at designing, integrating, and measuring diffusion. Shane J. Kanter, John Strong, Carl Giegold, and Scott Pfeiffer (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

A primary goal of the small-scale performance venue is to provide the audience with supportive, well-timed reflections and to energize the space adequately without overpowering the room volume. The judicious use of sound-diffusive elements in such venues can lend a pleasing sense of body and space while avoiding undesirable reflections that disrupt the listener experience. However, while working with architects to develop a space that is pleasing to both the ear and the eye, it is often necessary to reconcile these needs with each other. Diffusive elements must integrate seamlessly within the space visually as well as architecturally. While developing interior room acoustics for three small spaces for performance/worship, with audience size ranging from 150 to 299, an exploration of diffusive elements was conducted. As each project required a different method and frequency range of diffusion, scale models were constructed and tested under varied conditions, using sometimes unorthodox methods to determine the acoustic effect. These efforts were focused on limiting coloration caused by the "picket fence effect," reducing harsh reflections without rendering a space excessively sound-absorptive, and maintaining coherent reflections from discrete sections of a prominent wall while leaving other sections diffusive. Methods, experiences, and results will be presented.

Session 4aAB

Animal Bioacoustics and Acoustical Oceanography: Use of Passive Acoustics for Estimation of Animal Population Density I

Tina M. Yack, Cochair
Bio-Waves, Inc., 364 2nd Street, Suite #3, Encinitas, CA 92024

Danielle Harris, Cochair
Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, The Observatory, Buchanan Gardens, St. Andrews KY16 9LZ, United Kingdom

Chair's Introduction—8:00

Invited Papers

8:05

4aAB1. Estimating density from passive acoustics: Are we there yet? Tiago A. Marques, Danielle Harris, and Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews, Fife KY16 9 LZ, United Kingdom, tiago.marques@st-andrews.ac.uk)

In the last few years, there have been a considerable number of papers describing methods or case studies involving passive acoustic density estimation. While this might be interpreted as evidence that density estimates might now be easily and routinely implemented, the truth is that so far these methods and applications have been essentially proof-of-concept in nature, based in areas and/or species particularly suited for the methods and also often involved assumptions hard to evaluate. We briefly review some of the existing work in this area concentrating on a few aspects we believe are key for the implementation of density estimation from passive acoustics in a broader context. These are (1) the development of fundamental research addressing the problem of sound production rate, fundamental as it allows to convert estimates of density of sounds into density of animals and (2) the development of hardware capable of providing cheap deployable units capable of ranging, allowing straightforward implementations of distance sampling based approaches. The perfect density estimate is out there waiting to happen, but we have not found it yet.

8:25

4aAB2. Use of passive acoustics for estimation of cetacean population density: Realizing the potential. Jay Barlow and Shannon Rankin (Marine Mammal and Turtle Div., NOAA-SWFSC, 8901 La Jolla Shores Dr., La Jolla, CA 92037, jay.barlow@noaa.gov)

The potential of passive acoustic methods to estimate cetacean population density has seldom been realized. It has been most successfully applied to species that consistently use echo-location during foraging, have very distinctive echo-location signals and forage a large fraction of the time, notably sperm whale, porpoise, and beaked whales. Research is needed to eliminate some of the impediments to applying acoustics to estimate the density of other species. For baleen whales, one of the greatest uncertainties is the lack of information on call rates. For delphinids, the greatest uncertainties are in estimating group size and in species recognition. For all species, there is a need to develop inexpensive recorders that can be distributed in large number at random locations in a study area. For towed hydrophone surveys, there is a need to better localize species in their 3-D environment and to instantaneously localize animals from a single signal received on multiple hydrophones. While improvements can be made, we may need to recognize that some of impediments cannot be overcome with any reasonable research budget. In these cases, efforts should be concentrated in improving acoustic methods to aid visual-based transect methods.

8:45

4aAB3. Acoustic capture-recapture methods for animal density estimation. David Borchers (Dept. of Mathematics & Statistics, Univ. of St. Andrews, CREEM, Buchannan Gdns, St. Andrews, Fife KY16 9LZ, United Kingdom, dlb@st-andrews.ac.uk)

Capture-recapture methods are one of the two most widely-used methods of estimating wildlife density and abundance. They can be used with passive acoustic detectors—in which case acoustic detection on a detector constitutes "capture" and detection on other detectors and/or at other times constitute "recaptures." Unbiased estimation of animal density from any capture-recapture survey requires that the effective area of the detectors be estimated, and information on detected animals' locations are essential for this. While locations are not observed, acoustic data contain information on location in a variety of guises, including time-difference-of arrival, signal strength, and sometimes directional information. This talk gives an overview of the use of such data with spatially explicit capture-recapture (SECR) methods, including consideration of some of the particular challenges that acoustic data present for SECR methods, ways of dealing with these, and an outline of some unresolved issues.

9:05

4aAB4. U.S. Navy application and interest in passive acoustics for estimation of marine mammal population density. Anu Kumar (Living Marine Resources, NAVFAC EXWC, 1000 23rd Ave., Code EV, Port Hueneme, CA 93043, anurag.kumar@navy.mil), Chip Johnson (Environ. Readiness, Command Pacific Fleet, Coronado, CA), Julie Rivers (Environ. Readiness, Command Pacific Fleet, Pearl Harbor, HI), Jene Nissen (Environ. Readiness, U.S. Fleet Forces, Norfolk, VA), and Joel Bell (Marine Resources, NAVFAC Atlantic, Norfolk, VA)

Marine species population density estimation from passive acoustic monitoring is an emergent topic of interest to the U.S. Navy. Density estimates are used by the Navy and other Federal partners in effects modeling for environmental compliance documentation. Current traditional methods of marine mammal density estimation via visual line transect surveys require expensive ship time and long days at-sea for an experienced crew to yield limited spatial and temporal coverage. While visual surveys remain an effective means of deriving density estimates, passive acoustic based density estimation methods have the unique ability to improve on visual density estimates for some key species by: (a) expanding spatial and temporal density coverage, (b) providing coverage in areas too remote or difficult for traditional visual surveys, (c) reduce the statistical uncertainty of a given density estimate, and (d) providing estimates for species that are difficult to survey visually (e.g., minke and beaked whales). The U.S. Navy has invested in research for the development, refinement, and scientific validation of passive acoustic methods for cost effective density estimates in the future. The value, importance, and current development in passive acoustic-based density estimation methods for Navy applications will be discussed.

9:25

4aAB5. Towing the line: Line-transect based density estimation of whales using towed hydrophone arrays. Thomas F. Norris and Tina M. Yack (Bio-Waves Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, thomas.f.norris@bio-waves.net)

Towed hydrophone arrays have been used to monitor marine mammals from research vessels since the 1980s. Although towed hydrophone arrays have now become a standard part of line-transect surveys of cetaceans, density estimation exclusively using passive acoustic has only been attempted for a few species. We use examples from four acoustic line-transect surveys that we conducted in the North Pacific Ocean to illustrate the steps involved, and issues inherent, in using data from towed hydrophone arrays to estimate densities of cetaceans. We will focus on two species of cetaceans, sperm whales and minke whales, with examples of beaked whales and other species as needed. Issues related to survey design, data-collection, and data analysis and interpretation will be discussed using examples from these studies. We provide recommendations to improve the survey design, data-collection methods, and analyses. We also suggest areas where additional research and methodological development are required in order to produce robust density estimates from acoustic based data.

Contributed Papers

9:45

4aAB6. From clicks to counts: Applying line-transect methods to passive acoustic monitoring of sperm whales in the Gulf of Alaska. Tina M. Yack, Thomas F. Norris, Elizabeth Ferguson (Bio-Waves Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, tina.yack@bio-waves.net), Brenda K. Rone (Cascadia Res. Collective, Seattle, WA), and Alexandre N. Zerbini (Alaska Fisheries Sci. Ctr., Seattle, WA)

A visual and acoustic line-transect survey of marine mammals was conducted in the central Gulf of Alaska (GoA) during the summer of 2013. The survey area was divided into four sub-strata to reflect four distinct habitats; "inshore," "slope," "offshore," and "seamount." Passive acoustic monitoring was conducted using a towed-hydrophone array system. One of the main objectives of the acoustic survey was to obtain an acoustic-based density estimate for sperm whales. A total of 241 acoustic encounters of sperm whales during 6,304 km of effort were obtained compared to 19 visual encounters during 4,155 km of effort. Line-transect analytical methods were used to estimate the abundance of sperm whales. To estimate the detection function, target motion analysis was used to obtain perpendicular distances to individual sperm whales. An acoustic-based density and abundance estimate was obtained for each stratum (N = 78; CV = 0.36 offshore; N = 16; CV = 0.55 seamount; N = 121; and CV = 0.18 slope) and for the entire survey area (N = 215; D = 0.0013; and CV = 0.18). These results will be compared to visual-based estimates. The advantages and disadvantages of acoustic-based density estimates as well as application of these methods to other species (e.g., beaked whales) and areas will be discussed.

10:00–10:15 Break

10:15

4aAB7. Studying the biosonar activities of deep diving odontocetes in Hawaii and other western Pacific locations. Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu) and Giacomo Giorli (Oceanogr. Dept., Univ. of Hawaii, Honolulu, HI)

Ecological acoustic recorders (EARs) have been deployed at several locations in Hawaii and in other western Pacific locations to study the foraging behavior of deep-diving odontocetes. EARs have been deployed at depths greater than 400 m at five locations around the island of Kauai, one at Ni'ihau, two around the island of Okinawa, and four in the Marianas (two close to island of Guam, one close to the island of Saipan and another close to the island of Tinian). The four groups of deep-diving odontocetes were blackfish (mainly pilot whales and false killer whales), sperm whales, beaked whales (Cuvier and Bainsville beaked whales) and Risso's dolphin. In all locations, the biosonar signals of blackfish were detected the most followed by either by sperm and beaked whales depending on specific locations with Risso's dolphin being detected the least. There was a strong tendency for these animals to forage at night in all locations. The detection rate indicate much lower populations of these four groups of odontocetes around Okinawa and in the Marianas then off Kauai in the main Hawaiian island chain by a factor of about 4–5.

10:30

4aAB8. Fin whale vocalization classification and abundance estimation. Wei Huang, Delin Wang (Elec. and Comput. Eng., Northeastern Univ., 006 Hayden Hall, 370 Huntington Ave., Boston, MA 02115, huang.wei1@husky.neu.edu), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Several thousand fin whale vocalizations from multiple fin individuals were passively recorded by a high-resolution coherent hydrophone array system in the Gulf of Maine in Fall 2006. The recorded fin whale vocalizations have relatively short durations roughly 0.4 s and frequencies ranging

from 15 to 40 Hz. Here we classify the fin whale vocalizations and apply the results to estimate the minimum number of vocalizing fin individuals detected by our hydrophone array. The horizontal azimuth or bearing of each fin whale vocalization is first determined by beamforming. Each beamformed fin whale vocalization spectrogram is next characterized by several features such as center frequency, upper and lower frequency limits, as well as amplitude-weighted mean frequency. The vocalizations are then classified using k-mean clustering into several distinct vocal types. The vocalization clustering result is then combined with the bearing-time trajectory information for a consecutive sequence of vocalizations to provide an estimate of the minimum number of vocalizing fin individuals detected.

10:45

4aAB9. Neglect of bandwidth of odontocetes echolocation clicks biases propagation loss and single hydrophone population estimates. Michael A. Ainslie, Alexander M. von Benda-Beckmann (Acoust. and Sonar, TNO, P.O. Box 96864, The Hague 2509JG, Netherlands, michael.ainslie@tno.nl), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St Andrews, United Kingdom), and Tyack L. Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, United Kingdom)

Passive acoustic monitoring with a single hydrophone has been suggested as a cost-effective method to monitor population density of echolocating marine mammals, by estimating the distance at which the hydrophone is able to distinguish the echolocation clicks from the background. To avoid a bias in the estimated population density, this method relies on an unbiased estimate of the propagation loss (PL). It is common practice to estimate PL at the center frequency of a broadband echolocation click and to assume this narrowband PL applies also to the broadband click. For a typical situation this narrowband approximation overestimates PL, underestimates the detection range and consequently overestimates the population density by an amount that for fixed center frequency increases with increasing pulse bandwidth and sonar figure of merit. We investigate the detection process for different marine mammal species and assess the magnitude of error on the estimated density due to various simplifying assumptions. Our main purposes are to quantify and, where possible and needed, correct the bias in the population density estimate for selected species and detectors due to use of the narrowband approximation, and to understand the factors affecting the magnitude of this bias to enable extrapolation to other species and detectors.

11:00

4aAB10. Instantaneous acoustical response of marine mammals to abrupt changes in ambient noise. John E. Joseph, Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd, Monterey, CA, jejoseph@nps.edu), and Ming-Jer Huang (National Kaohsiung Univ. of Appl. Sci., Kaohsiung, Taiwan)

Four months of passive acoustic data recorded at Thirtymile Bank in offshore southern California have been analyzed to describe instantaneous vocal response of marine mammals to abrupt changes in ambient noise. Main contributors to the distinctive regional soundscape are heavy commercial shipping, military activities in the naval training range, diverse marine life and natural sources including wind and tectonic activity. Many of these sources produce intense, irregular and short-term events shaped by local oceanographic conditions, bathymetry and bottom structure (Thirtymile Bank blind thrust). We seek to attribute detected changes in cetacean vocal behavior (loudness, calling rate, and pattern) to these events and differentiate the reaction by noise source, its intensity, frequency and/or duration. Main target species are blue and fin whales. Initial hypotheses formulated after data scanning are tested statistically (2D histograms and PCA). To quantify the vocal behavior variations, an innovative detection approach based on pattern recognition is applied, which allows for extraction of individual calls with low false alarm and high detection success comparable to those of a human analyst. Obtained results relate cetacean acoustic behavior to ambient noise variability and thus help refine existing cue-based formulae for estimation of whale population density from PAM data.

11:15

4aAB11. Measuring whale and dolphin call rates as a function of behavioral, social, and environmental context. Stacy L. DeRuiter, Catriona M. Harris (School of Mathematics & Statistics, Univ. of St. Andrews, CREEM, St. Andrews KY169LZ, United Kingdom, sldr@st-andrews.ac.uk), Nicola J. Quick (Duke University Marine Lab, Duke Univ., Beaufort, NC), Dina Sadykova, Lindesay A. Scott-Hayward (School of Mathematics & Statistics, Univ. of St. Andrews, St. Andrews, United Kingdom), Alison K. Stimpert (Moss Landing Marine Lab., California State Univ., Moss Landing, CA), Brandon L. Southall (Southall Environ. Assoc., Inc., Aptos, CA), Len Thomas (School of Mathematics & Statistics, Univ. of St. Andrews, St Andrews, United Kingdom), and Fleur Visser (Kelp Marine Res., Hoorn, Netherlands)

Cetacean sound-production rates are highly variable and patchy in time, depending upon individual behavior, social context, and environmental context. Better quantification of the drivers of this variability should allow more realistic estimates of expected call rates, improving our ability to convert between call counts and animal density, and also facilitating detection of sound-production changes due to acoustic disturbance. Here, we analyze digital acoustic tag (DTAG) records and visual observations collected during behavioral response studies (BRSs), which aim to assess normal cetacean behavior and measure changes in response to acoustic disturbance; data sources include SOCAL BRS, the 3S project, and Bahamas BRS, with statistical contributions from the MOCHA project (http://www.creem.st-and.ac.uk/mocha/links). We illustrate use of generalized linear models (and their extensions) as a flexible framework for sound-production-rate analysis. In the context of acoustic disturbance, we also detail use of two-dimensional spatially adaptive surfaces to jointly model effects of sound-source proximity and sound intensity. Specifically, we quantify variability in pilot whale group sound production rates in relation to behavior and environment, and individual fin whale call rates in relation to social and environmental context and dive behavior; with and without acoustic disturbance.

11:30

4aAB12. Estimating relative abundance of singing humpback whales in Los Cabos, México, using diffuse ambient noise. Kerri Seger, Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8880 Biological Grade, MESOM 161, La Jolla, CA 92093-0206, kseger@ucsd.edu), Diana C. López Arzate, and Jorge Urbán (Laboratorio de Mamíferos Marinos, Universidad Autónoma de Baja California Sur, La Paz, BCS, Mexico)

Previous research has speculated that diffuse ambient noise levels can be used to estimate relative cetacean abundance in certain locations when baleen whale vocal activity dominates the soundscape (Au *et al.*, 2000; Mellinger *et al.*, 2009). During the 2013 and 2014 humpback whale breeding seasons off Los Cabos, Mexico, visual point and line transects were conducted alongside two bottom-mounted acoustic deployments. As theorized, preliminary analysis of ambient noise between 100 and 1,000 Hz is dominated by humpback whale song. It also displays a diel cycle similar to that found in the West Indies, Australia, and Hawai'i, whereby peak levels occur near midnight and troughs occur soon after sunrise (Au *et al.*, 2000; McCauley *et al.*, 1996). Depending upon site and year, the median band-integrated levels fluctuated between 7 and 16 dB re 1 uPa when sampled in one hour increments. This presentation uses analytical models of wind-generated noise in an ocean waveguide to analyze potential relationships between singing whale density and diffuse ambient noise levels. It explores whether various diel cycle strengths (peak-to-peak measurements and Fourier analysis) correspond with trends observed from concurrent visual censuses. [Work sponsored by the Ocean Foundation.]

11:45

4aAB13. Large-scale static acoustic survey of a low-density population—Estimating the abundance of the Baltic Sea harbor porpoise. Jens C. Koblitz (German Oceanogr. Museum, Katharinenberg 14-20, Stralsund 18439, Germany, Jens.Koblitz@meeresmuseum.de), Mats Amundin (Kolmården Wildlife Park, Kolmarden, Sweden), Julia Carlström (AquaBiota Water Res., Stockholm, Sweden), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Ida Carlén (AquaBiota Water Res., Stockholm, Sweden), Jonas Teilmann (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Nick Tregenza (Chelonia Ltd., Long Rock, United Kingdom), Daniel Wennerberg (Kolmården Wildlife Park, Kolmarden, Sweden), Line Kyhn, Signe Svegaard (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Radek Koza, Monika Kosecka, Iwona Pawliczka (Univ. of Gdansk, Gdansk, Poland), Cinthia Tiberi Ljungqvist (Kolmården Wildlife Park, Kolmården, Sweden), Katharina Brundiers (German Oceanogr. Museum, Stralsund, Germany), Andrew Wright (George Mason Univ., Fairfax, VA), Lonnie Mikkelsen, Jakob Tougaard (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Olli Loisa (Turku Univ. of Appl. Sci., Turku, Finland), Anders Galatius (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Ivar Jüssi (ProMare NPO, Harjumaa, Estonia), and Harald Benke (German Oceanogr. Museum, Stralsund, Germany)

SAMBAH (Static Acoustic Monitoring of the Baltic Sea Harbor Porpoise) is an EU LIFE+-funded project with the primary goal of estimating the abundance and distribution of the critically endangered Baltic Sea harbor porpoise. From May 2011 to April 2013, project members in all EU countries around the Baltic Sea undertook a static acoustic survey using 304 porpoise detectors distributed in a randomly positioned systematic grid in waters 5–80 m deep. In the recorded data, click trains originating from porpoises have been identified automatically using an algorithm developed specifically for Baltic conditions. To determine the click train C-POD detection function, a series of experiments have been carried out, including acoustic tracking of wild free ranging porpoises using hydrophone arrays in an area with moored C-PODs and playbacks of porpoise-like signals at SAMBAH C-PODs during various hydrological conditions. Porpoise abundance has been estimated by counting the number of individuals detected in short time interval windows (snapshots), and then accounting for false positive detections, probability of animals being silent, and probability of detection of non-silent animals within a specified maximum range. We describe the method in detail, and how the auxiliary experiments have enabled us to estimate the required quantities.

THURSDAY MORNING, 30 OCTOBER 2014 — INDIANA A/B, 7:55 A.M. TO 12:00 NOON

Session 4aBA

Biomedical Acoustics: Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications I

Vera A. Khokhlova, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Jeffrey B. Fowlkes, Cochair
Univ. of Michigan Health System, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Chair's Introduction—7:55

Invited Papers

8:00

4aBA1. Histotripsy: An overview. Charles A. Cain (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., 2121 Gerstacker, Ann Arbor, MI 48105, cain@umich.edu)

Histotripsy produces non-thermal lesions by generating dense highly confined energetic bubble clouds that mechanically fractionate tissue. This nonlinear thresholding phenomenon has useful consequences. If only the tip of the waveform (P-) exceeds the intrinsic threshold*, small lesions less than the diffraction limit can be generated. This is called microtripsy (other presentations in this session). Moreover, side lobes from distorting aberrations can be "thresholded-out" wherein part of the main lobe exceeds the intrinsic threshold producing a clean bubble cloud (and lesion) conferring significant immunity to aberrations. If a high frequency probe (imaging) waveform intersects a low frequency pump waveform, the compounded waveform can momentarily exceed the intrinsic threshold producing a lesion with an imaging transducer. Multi-beam histotripsy (other presentations in this session) allows flexible placement of both pump and probe transducers. Very broadband P- "monopolar" pulses*, ideal for histotripsy, can be synthesized in a generalization of the multi-beam histotripsy (other presentations in this session) case wherein very short pulses from transducer elements of many different frequencies are added at the focus of what is called a frequency compounding transducer (other presentations in this session). Ultrasound image guidance works well with histotripsy. Bubble clouds are easily seen simplifying both lesion targeting and continuous validation of the ongoing process. Hypoechoic homogenized tissue allows real-time quantification of lesion formation.

8:20

4aBA2. Boiling histotripsy: A noninvasive method for mechanical tissue disintegration. Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Tatiana D. Khokhlova (Dept. of Gastroenterology, Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Yak-Nam Wang, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Petr Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Julianna C. Simon (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Navid Farr (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Ari Partanen (Clinical Sci., Philips Healthcare, Cleveland, OH), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joo Ha Hwang (Dept. of Gastroenterology, Univ. of Washington, Seattle, WA), Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Boiling histotripsy is an experimental noninvasive focused ultrasound therapy that applies shocked ms-length pulses to achieve mechanical disintegration of a targeted tissue. Localized delivery of high-amplitude shocks causes rapid heating, resulting in boiling of the tissue. The interaction of incident shocks with the boiling bubble results in tissue disruption and liquefaction without significant thermal injury. Simulations are utilized to design and characterize therapy sources, predicting focal waveforms, shock amplitudes, and boiling times. Transducers have been developed to generate focal shock amplitudes >70 MPa and achieve rapid boiling at depth in tissue. Therapy systems including ultrasound-guided single-element sources and clinical MRI-guided phased arrays have been successfully used to create *ex vivo* and *in vivo* lesions at ultrasound frequencies in the 1–3 MHz range. Histological and biochemical analyses show mechanical disruption of tissue architecture with minimal thermal effect, similar to cavitation-based histotripsy. Atomization as observed with acoustic fountains has been proposed as an underlying mechanism of tissue disintegration. This promising technology is being explored for several applications in tissue ablation, as well as new areas such as tissue engineering and biomarker detection. [Work supported by NIH 2T32DK007779-11A1, R01EB007643-05, 1K01EB015745, and NSBRI through NASA NCC 9-58.]

8:40

4aBA3. Bubbles in tissue: Yes or No? Charles C. Church (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu)

The question of whether bubbles exist in most or all biological tissues rather than being restricted to only a few well-known examples remains a mystery. When Apfel and Holland developed the theoretical background for the mechanical index (MI), they first assumed that such bubbles did exist and further assumed that some of those bubbles were of a size that would undergo inertial cavitation at the lowest possible rarefactional pressure. Comparison of cavitation thresholds determined experimentally in various mammalian tissues *in vivo* with the results of computational studies seems to provide a definitive answer to that question. No, optimally sized bubbles do not pre-exist in tissue, although very small bubbles, with radii on the order of nm, may be present. However, this answer is inextricably related to the accuracy of the theory used to study the question, in this case a form of the Keller-Miksis equation modified to include the viscoelastic properties of tissue. Previous analysis has focused on elasticity, assuming that viscosity is constant, but is it? Blood is known to be shear-thinning, and some soft tissues appear to be as well. The effect of shear rate on cavitation thresholds and implications for bubble populations in tissue will be discussed.

9:00

4aBA4. Benefits and challenges of employing elevated acoustic output in diagnostic imaging. Kathryn Nightingale (Biomedical Eng., Duke Univ., PO Box 90281, Durham, NC 27708-0281, kathy.nightingale@duke.edu) and Charles C. Church (National Ctr. for Acoust., Univ. of MS, University, MS)

The acoustic output levels used in diagnostic ultrasonic imaging in the US have been subject to a de facto limitation by guidelines established by the USFDA in 1976, for which no known bioeffects had been reported. These track-3 guidelines link the Mechanical Index (MI) and the Thermal Index (TI) to the maximum outputs as of May 28, 1976, through a linear derating process. Subsequently, new imaging technologies have been developed that employ unique beam sequences (e.g., harmonic imaging and ARFI imaging) which were not well developed when the current regulatory scheme was put in place, so neither the MI nor the TI takes them into account in an optimal manner. Additionally, there appears to be a large separation between the maxima in the track-3 guidelines and the acoustic output levels for which cavitation-based bioeffects are observed in tissues not known to contain gas bodies. In this presentation, we summarize the history of and the scientific basis for the MI, define an output regime and specify clinical applications under consideration for conditionally increased output (CIO), review the potential risks of CIO in this regime based upon existing scientific evidence, and summarize the evidence for the potential clinical benefits of CIO.

9:20

4aBA5. Standards for characterizing highly nonlinear acoustic output from therapeutic ultrasound devices: Current methods and future challenges. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

One of the major challenges of characterizing the acoustic fields and power from diagnostic and high intensity or high pressure therapeutic devices is addressing the impact of amplitude-dependent nonlinear propagation effects. The destructive capabilities of high intensity therapeutic devices (HITU) make acoustic output measurements with conventional fragile sensors used for diagnostic ultrasound difficult. Different approaches involving more robust measurement devices, scaling and simulation are described in two recent IEC documents, IEC TS 62556 for the specification and measurement of HITU fields and IEC 62555 for the measurement of acoustic power from HITU devices. Existing and proposed applications include even higher pressure levels and use of cavitation effects. Promising hybrid approaches involve a combination of measurement and simulation. In order to meet the challenges of design, verification, and measurement, standards and consensus are needed to couple the measurements to the prediction of acoustic output in realistic tissue models as well as associated effects such as acoustic radiation force and temperature elevation.

9:40

4aBA6. Uncertainties in characterization of high-intensity, nonlinear pressure fields for therapeutic applications. Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Petr V. Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Tatiana D. Khokhlova (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Michael R. Bailey (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov, and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

A fundamental aspect of developing therapeutic ultrasound applications is the need to quantitatively characterize the acoustic fields delivered by transducers. A typical approach is to make direct pressure measurements in water. With very high intensities and potentially shocks, executing this approach is problematic because of the strict requirements imposed on hydrophone bandwidth, robustness, and size. To overcome these issues, a method has been proposed that relies on acoustic holography and simulations of nonlinear propagation based on the 3D Westervelt model. This approach has been applied to several therapy transducers including a multi-element phased array. Uncertainties in the approach can be evaluated for both model boundary conditions determined from linear holography and the nonlinear focusing gain achieved at high power levels. Neglecting hydrophone calibration uncertainties, errors associated with the holography technique remain less than about 10% in practice. To assess the accuracy of nonlinear simulations, results were compared to independent measurements of focal waveforms using a fiber optic probe hydrophone (FOPH). When relative calibration uncertainties between the capsule hydrophone and FOPH are mitigated, simulations and FOPH measurements agree within about 15% for peak pressures at the focus. [Work supported by NIH grants EB016118, EB007643, T32 DK007779, DK43881, and NSBRI through NASA NCC 9-58.]

10:00–10:20 Break

10:20

4aBA7. Cavitation characteristics in High Intensity Focused Ultrasound lesions. Gail ter Haar and Ian Rivens (Phys., Inst. of Cancer Res., Phys. Dept., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, United Kingdom, gail.terhaar@icr.ac.uk)

The acoustic emissions recorded during HIFU lesions fall into three broad categories: those associated with non-inertial cavitation, those associated with inertial cavitation, and those linked with tissue water boiling. These three mechanisms can be linked with different lesion shapes, and with characteristic histological appearance. By careful choice of acoustic driving parameters, these effects may be studied individually.

10:40

4aBA8. The role of tissue mechanical properties in histotripsy tissue fractionation. Eli Vlaisavljevich, Charles Cain, and Zhen Xu (Univ. of Michigan, 1111 Nielsen Ct. Apt. 1, Ann Arbor, MI 48105, evlaisav@umich.edu)

Histotripsy is a therapeutic ultrasound technique that controls cavitation to fractionate tissue using short, high-pressure ultrasound pulses. Histotripsy has been demonstrated to successfully fractionate many different tissues, though stiffer tissues such as cartilage or tendon (Young's moduli >1 MPa) are more resistant to histotripsy-induced damage than softer tissues such as liver (Young's moduli ~9 kPa). In this work, we investigate the effects of tissue mechanical properties on various aspects of the histotripsy process including the pressure threshold required to generate a cavitation cloud, the bubble dynamics, and the stress–strain applied to tissue structures. Ultrasound pulses of 1–2 acoustic cycles at varying frequencies (345 kHz, 500 kHz, 1.5 MHz, and 3 MHz) were applied to agarose tissue phantoms and ex vivo bovine tissues with varying mechanical properties. Results demonstrate that the intrinsic threshold to initiate a cavitation cloud is independent of tissue stiffness and frequency. The bubble expansion is suppressed in stiffer tissues, leading to a decrease in strain to surrounding tissue and an increase in damage resistance. Finally, we investigate strategies to optimize histotripsy therapy for the treatment of tissues with specific mechanical properties. Overall, this work improves our understanding of how tissue properties affect histotripsy and will guide parameter optimization for histotripsy tissue fractionation.

11:00

4aBA9. Technical advances for histotripsy: Strategic ultrasound pulsing methods for precise histotripsy lesion formation. Kuang-Wei Lin, Timothy L. Hall, Zhen Xu, and Charles A. Cain (Univ. of Michigan, 2200 Bonisteel Blvd., Gerstacker, Rm. 1107, Ann Arbor, MI 48109, kwlin@umich.edu)

Conventional histotripsy uses ultrasound pulses longer than three cycles wherein the bubble cloud formation relies on the pressure-release scattering of the positive shock fronts from sparsely distributed single cavitation bubbles, making the cavitation event unpredictable and sometimes chaotic. Recently, we have developed three new strategic histotripsy pulsing techniques to further increase the precision of cavitation cloud and lesion formation. (1) Microtripsy: When applying histotripsy pulses shorter than three cycles, the formation of a dense bubble cloud only depends on the applied peak negative pressure (P-) exceeding an intrinsic threshold of the medium. With a P- not significantly higher than this, very precise sub-vocal-volume lesions can be generated. (2) Dual-beam histotripsy: A sub-threshold high-frequency pulse (perhaps from an imaging transducer) is enabled by a sub-threshold low-frequency pump pulse to exceed the intrinsic threshold and produces very precise lesions. (3) Frequency compounding: a near monopolar pulse can be synthesized using a frequency-compounding transducer (an array transducer consisting of elements with various resonant frequencies). By adjusting time delays for individual frequency components and allowing their principal negative peaks to arrive at the focus concurrently, a near monopolar pulse with a dominant negative phase can be generated (no complicating high peak positive shock fronts).

11:20

4aBA10. Histotripsy: Urologic applications and translational progress. William W. Roberts (Urology, Univ. of Michigan, 3879 Taubman Ctr., 1500 East Medical Ctr. Dr., Ann Arbor, MI 48109-5330, willrobe@umich.edu), Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), J. B. Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), Zhen Xu, and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy is an extracorporeal ablative technology based on initiation and control of acoustic cavitation within a target volume. This mechanical form of tissue homogenization differs from the ablative processes employed by conventional thermoablative modalities and exhibits a number of unique features (non-thermal, high precision, real-time monitoring/feedback, and tissue liquefaction), which are potentially advantageous characteristics for ablative applications in a variety of organs and disease processes. Histotripsy has been applied to the prostate in canine models for tissue debulking as a therapy for benign prostatic hyperplasia and for ablation of ACE-1 tumors, a canine prostate cancer model. Homogenization of normal renal tissue as well as implanted VX-2 renal tumors has been demonstrated with histotripsy. Initial studies assessing tumor metastases in this model did not reveal metastatic potentiation following mechanical homogenization by histotripsy. Enhanced understanding of cavitation and methods for acoustic control of the target volume are being refined in tank studies for treatment of urinary calculi. Development of novel acoustic pulsing strategies, refinement of technology, and enhanced understanding of cavitational bioeffects are driving pre-clinical translation of histotripsy for a number of applications. A human pilot trial is underway to assess the safety of histotripsy as a treatment for benign prostatic hyperplasia.

11:40

4aBA11. Boiling histotripsy of the kidney: Preliminary studies and predictors of treatment effectiveness. George R. Schade, Adam D. Maxwell (Dept. of Urology, Univ. of Washington, 5555 14th Ave. NW, Apt 342, Seattle, WA 98107, grschade@uw.edu), Tatiana Khokhlova (Dept. of Gastroenterology, Univ. of Washington, Seattle, WA), Yak-Nam Wang, Oleg Sapoznikov, Michael R. Bailey, and Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Boiling histotripsy (BH), an ultrasound technique to mechanically homogenize tissue, has been described in *ex vivo* liver and myocardium. As a noninvasive, non-thermal based approach, BH may have advantages over clinically available thermal ablative technologies for renal masses. We aimed to characterize BH exposures in human and porcine *ex vivo* kidneys using a 7-element 1 MHz transducer (duty factor 1–3%, 5–10 ms pulses, 98 MPa *in situ* shock amplitude, 17 MPa peak negative). Lesions were successfully created in both species, demonstrating focally homogenized tissue above treatment thresholds (pulse number) with stark transition between treated and untreated cells on histologic assessment. Human tissue generally required more pulses to produce similar effect compared to porcine. Similarly, kidneys displayed tissue specific resistance to BH with increasing resistance from cortex to medulla to the collecting system. Tissue properties that predict resistance to renal BH were evaluated demonstrating correlation between tissue collagen content and tissue resistance. Subsequently, the impact of intervening abdominal wall and ribs on lesion generation *ex vivo* was evaluated. "Transabdominal" and "transcostal" treatment required approximately 5- and 20-fold greater acoustic power, respectively, to elicit boiling vs. no intervening tissue. [Work supported by NIH T32DK007779, R01EB007643, K01EB015745 and NSBRI through NASA NCC 9-58.]

THURSDAY MORNING, 30 OCTOBER 2014 MARRIOTT 9/10, 8:30 A.M. TO 11:15 A.M.

Session 4aEA

Engineering Acoustics: Acoustic Transduction: Theory and Practice I

Richard D. Costley, Chair
Geotechnical and Structures Lab., U.S. Army Engineer Research & Development Center, 3909 Halls Ferry Rd, Vicksburg, MS 39180

Contributed Papers

8:30

4aEA1. Historic transducers: Balanced armature receiver (BAR). Jont B. Allen (ECE, Univ. of Illinois, Urbana-Champaign, Urbana, IL) and Noori Kim (ECE, Univ. of Illinois, Urbana-Champaign, 1085 Baytowner df 11, Champaign, IL 61822, nkim13@illinois.edu)

The oldest telephone receiver is the Balanced Armature Receiver (BAR) type, and it is still in use. The original technology goes back to the invention of telephone receiver by A. G. Bell in 1876. Attraction and release of the armature are controlled by the current from the coils, which generates electromagnetic fields [Hunt (1954) Chapter 7, and Beranek and Mellow (2014)]. As the electrical current goes into the electric terminal of the receiver, it generates an AC magnetic field which direction is perpendicular to the current. Due to the polarity between the permanent (DC) magnet field and the generated AC magnetic field, an armature (which sits within the core of the coil and the magnet) feels a force. The very basic principles for explaining this movement in a gyrator, a fifth circuit element introduced by Tellegen in 1948, along with an inductor, a capacitor, a resistor, and a transformer. This component represents the anti-reciprocal characteristic of the system. This study is starting from comparing the BAR type receiver to the moving-coil loud speaker. We believe that this work will provide a fundamental and clear insight into this type of BAR system.

8:45

4aEA2. Radiation from wedges of a power law profile. Marcel C. Remillieux, Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, mcr1@lanl.gov)

The large impedance contrast between bulk piezoelectric disks and air does not allow for efficient coupling of sound radiation from the piezoelectric into air. Here, we present the idea of using wedges of power law profiles to more efficiently radiate sound into air. Wedges of power law profiles have been used to provide absorption of vibrational energy in plates, but their efficient radiation of sound into air has not been demonstrated. We present numerical modeling and experimental results to demonstrate the concept. The wedge shape provides a gradual impedance contrast as the wave travels down the tapering of the wedge, while the wave speed also continually slows down. For an ideal wedge that tapers down to zero thickness, the waves become trapped at the tip and the vibrational energy can only radiate into the surrounding air. [This work was supported by institutional support [Laboratory Directed Research and Development (LDRD)] at Los Alamos National Laboratory.]

9:00

4aEA3. The self-sustained oscillator as an underwater low frequency projector: Progress report. Andrew A. Acquaviva and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., c/o Steve Thompson, N-249 Millennium Sci. Complex, University Park, PA, acquavaa@gmail.com)

Wind musical instruments are examples of pressure operated self-sustained oscillators that act as acoustic projectors. Recent studies have shown that this type of self-sustained oscillator can also be implemented underwater as a low frequency projector. However, the results of the early feasibility studies were complicated by the existence of cavitation in the high pressure region of the resonator. A redesign has eliminated the cavitation and allows better comparison with analytical calculations.

9:15

4aEA4. Design and testing of an underwater acoustic Fresnel zone plate diffractive lens. David C. Calvo, Abel L. Thangawng, Michael Nicholas, and Christopher N. Layman, Jr. (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil)

Fresnel zone plate (FZP) lenses offer a means of focusing sound based on diffraction in cases where the thickness of conventional lenses may be impractical. A binary-profile FZP for underwater use featuring a center acoustically opaque disk with alternating transparent and opaque annular regions was fabricated to operate nominally at 200 kHz. The overall diameter of the lens was 13 in. and consisted of 13 opaque annuli. The opaque regions were 3 mm thick and made from silicone rubber with a high concentration of gas voids. These regions were bonded to an acoustically transparent silicone rubber substrate film that was 1 mm thick. The FZP was situated in a frame and tested in a 5 x 4 x 4 cu. ft. ultrasonic tank using a piston source for insonification. The measured focal distance for normal incidence of 12.5 cm agreed with finite-element predictions taking into account the wavefront curvature of the incident field which had to be included given the finite dimensions of tank. The focal gain was measured to be 20 dB. The radius to the first null at the focal plane was approximately 4 mm, which agreed with theoretical resolution predictions. [Work sponsored by the Office of Naval Research.]

9:30

4aEA5. Acoustical transduction in two-dimensional piezoelectric array. Ola Nusierat, Lucien Cremaldi (Phys. and Astronomy, Univ. of MS, Oxford, MS), and Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu)

The acoustical transduction in an array of ferroelectric domains with alternating piezoelectric coefficients is characterized by multi-frequency resonances, which occur at the boundary of the acoustic Brillouin zone (ABZ). The resonances correspond to two successive domain excitations in the first and second ABZ correspondingly, where the speed of ultrasound is somewhat different. An important parameter for acoustical transduction is the electric impedance Z. The results of the theoretical and experimental investigations of Z in a periodically poled LiNbO3 are presented. The magnitude and phase of Z depend on the array parameters including domain resonance frequency and domain number; Z of arrays consisting of up to 88 0.45-mm-long domains in the zx-cut crystal are investigated. The strong changes in Z-magnitude and phase are observed in the range of 3–4 MHz. The two resonance zones are within 3.33 ± 0.05 MHz and 3.67± 0.05 MHz. The change in domain number influences Z and its phase. By varying the number of inversely poled domains and resonance frequencies, one can significantly control/change the electrical impedance of the multidomain array. The findings may be used for developing new acoustic sensors and transducers.

9:45

4aEA6. A non-conventional acoustic transduction method using fluidic laminar proportional amplifiers. Michael V. Scanlon (RDRL-SES-P, Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Pressure sensing using fluidic laminar proportional amplifiers (LPAs) was developed at Harry Diamond Laboratories in the late 1970s and was applied to acoustic detection and amplification. LPAs use a partially constrained laminar jet of low-pressure air as the sensing medium, which is deflected by the incoming acoustic signal. LPA geometries enable pressure gain by focusing incoming pressure fluctuations at the jet's nozzle exit, thereby applying leverage to create jet deflection over its short transit toward a splitter. With no input signal, the jet is not deflected and downstream pressures on both sides of the splitter are equal. A differential input signal of magnitude one, referenced to ambient pressure balancing the opposite side of the jet, produces an differential output signal of magnitude ten. This amplified signal can be differentially fed into the inputs on both sides of the next LPA for additional gain. By cascading LPAs together, very small signals can be amplified a large amount. Originally, a DC pressure amplifier, LPAs have exceptional infrasound response, and excellent sensitivity since there is no mass or stiffness associated with a diaphragm, and is matched to the environment. Standard microphones at the output ports can take advantage of increased sensitivity and gain.

10:00–10:15 Break

10:15

4aEA7. Investigation of piezoelectric bimorph bender transducers to generate and receive shear waves. Andrew R. McNeese, Kevin M. Lee, Megan S. Ballard, Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), and R. Daniel Costley (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

This paper further demonstrates the ability of piezoceramic bimorph bender elements to preferentially generate and receive near-surface shear waves for in situ sediment characterization measurements, in terrestrial as well as marine clay soils. The bimorph elements are housed in probe transducers that can manually be inserted into the sediment and are based on the work of Shirley [J. Acoust. Soc. Am. 63(5), 1643–1645 (1978)] and of Richardson *et al*. [Geo.—Marine Letts. 196–203 (1997)]. The transducers can discretely generate and receive horizontally polarized shear waves, within their bimorph directivity patterns. The use of multiple probes allows one to measure the shear wave velocity and attenuation parameters in the sediment of interest. Measured shear wave data on a hard clay terrestrial soil, as well as on soft marine sediments, are presented. These parameters along with density and compressional wave velocity define the elastic moduli (Poisson's ratio, shear modulus, and bulk modulus) of the sediment, which are of interest in various areas of geophysics, underwater acoustics, and geotechnical engineering. Discussion will focus on use of the probes in both terrestrial and marine sediment environments. [Work supported by ARL:UT Austin.]

10:30

4aEA8. Multi-mode seismic source for underground application. abderrhamane ounadjela (sonic, Schlumberger, 2-2-1 Fuchinobe, Sagamihara, Sagamihara, Kanagawa 252-0206, Japan, ounadjela1@slb.com), Henri Pierre Valero, Jean christophe Auchere (sonic, Schlumberger, Sagamihara-Shi, Japan), and Olivier Moyal (sonic, Schlumberger, Clamart, France)

A new multi-mode downhole acoustic source has been designed to fulfill requirements of oil business. Three acoustic modes of radiation, i.e., monopole, dipole, and quadruple modes, respectively, are considered to assess the properties of the oil reservoir. Because of the geometry of the well it is challenging to design an efficient and effective powerful device. This new source uses an apparatus to convert the axial motion of the four motors distributed on the azimuth into a radial one. In order to make this conversion effective, the axial motion transformation into a radial one is performed thanks to a rod rolling on a cone; this conversion minimizes the loss by friction and is very effective. The conversion apparatus is also exploited to match the acoustic impedance of the surrounding medium. This new design is described in this paper as well as intensive modeling which allowed optimizing this multi-mode source device. Experimental data is in a good agreement with numerical modeling.

10:45

4aEA9. Sound characteristics of the caxirola when used by different uninstructed users. Talita Pozzer and Stephan Paul (UFSM, Tuiuti, 925. Apto 21, Santa Maria, RS 97015661, Brazil, talita.pozzer@eac.ufsm.br)

While originally developed to be the official musical instrument of the 2014 Soccer World Cup the caxirola was banned form the stadiums as could be thrown into the field by angry spectators. Nevertheless, outside the stadiums the caxirola was still used, thus an already started investigation into the acoustics of the caxirola was concluded. At a previous ASA meeting we presented the sound power level (SWL) of the caxirola only for the two most typical ways of use. Now we present data on the sound pressure level close to the user's ears (SPLcue) and the SWL, both measured in a reverberation room, from 30 subjects that used the caxirola according their understanding. It was found that the total SPLcue vary from 78 dB(A) up to 95 dB(A) and the global SWL of the caxirola varies from 72 dB until 84 dB. The distribution is not normal, them the SWL has 79 dB(A) as median, that is very similar of the result obtained at the previous study. The SPLcue and the SPL measured for calculating the SWL are differents. This probably due to the distance variation between the source and the ear of user causing a near field some times.

11:00

4aEA10. A micro-machined hydrophone using the piezoelectric-gate-of-field-effect-transistor for low frequency sounds. Min Sung, Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 416, POSTECH, San31, Hyoja-dong, Nam-gu, Pohang, Kyungbuk 790784, South Korea, smmath2@postech.ac.kr), Cheeyoung Joh (Underwater sensor Lab., Agency for Defense Development, Changwon, Kyungnam, South Korea,), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), Pohang, Kyungbuk, South Korea)

The micro-sized piezoelectric body for the miniaturized hydrophone is known to have the limits in low frequencies due to its high impedance and low sensitivity. In this study, a new transduction mechanism named as PiGoFET (piezoelectric gate of field effect transistor) is devised so that its application for the miniaturized hydrophone could overcome the limits of the micro-sized piezoelectric body. The PiGoFET transduction can be realized by combining a field effect transistor and a small piezoelectric body on its gate. A micro-machined silicon membrane of 2 mm diameter was connected to the small piezoelectric body so that acoustic pressure can apply appropriate forces on the body on the FET gate. The electric field from the deformed piezoelectric body modulates the channel current of FET directly, thus the transduction makes the sound pressure transferred to the source–drain current effectively at very low frequencies with micro-sized piezoelectric body. Under the described concept, a hydrophone was fabricated by micro-machining and calibrated using the comparison method in low frequencies to investigate its performance. [Research funded by MRCnD.]

Session 4aPAa

Physical Acoustics, Underwater Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, and Noise: Borehole Acoustic Logging and Micro-Seismics for Hydrocarbon Reservoir Characterization

Said Assous, Cochair
Geoscience, Weatherford, East Leake, Loughborough LE126JX, United Kingdom

David Eccles, Cochair
Weatherford, Geoscience, Loughborough, United Kingdom

Chair's Introduction—8:00

Invited Papers

8:05

4aPAa1. Generalized collar waves and their characteristics. Xiuming Wang, Xiao He, and Xiumei Zhang (State Key Lab. of Acoust., Inst. of Acoust. , 21 4th Northwestern Ring Rd., Hadian District, Beijing 100190, China, wangxm@mail.ioa.ac.cn)

A good acoustic logging while drilling (ALWD) tool is difficult to be designed because of collar waves that propagate along the tool. There always exist such acoustic waves in ALWD. The collar wave arrivals can strongly interfere with formation compressional waves in wave slowness picking up. In the past years, a considerable research work has been seen in suppressing collar waves in order to accurately pick up p- and s-wave slowness, and the obtained p- and s-wave slowness accuracy is still a problem. In this work, numerical and physical experiments are conducted to tackle collar wave propagation problems. And the collar wave propagation physics is elaborated and a generalized collar wave concept is proposed. It is shown that collar waves are much complex, and they consist of two kinds of collar waves, i.e., the direct collar waves and indirect collar waves. Both of these two collar waves make the ALWD data difficult to process for formation wave slowness picking up. Because of drilling string structures, the complicated collar waves cannot be effectively suppressed only with a groove isolator.

8:20

4aPAa2. Characterizing the nonlinear interaction of S (shear) and P (longitudinal) waves in reservoir rocks. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu), Thomas Gallot (Sci. Inst., Univ. of the Republic, Montevideo, Uruguay), Alison Malcolm, Stephen Brown, Dan Burns, and Michael Fehler (Earth Resources Lab., Massachusetts Inst. of Technol., Cambridge, MA)

The nonlinear elastic response of rocks is known to be caused by internal microstructure, particularly cracks and fluids. In order to quantify this nonlinearity, this paper presents a method for characterizing the interaction of two nonresonant traveling waves: a low-amplitude P-wave probe and a high-amplitude lower frequency S-wave pump with their particle motions aligned. We measure changes in the arrival time of the P-wave probe as it passes through the perturbation created by a traveling S-wave pump in a sample of room-dry Berea sandstone ($15 \times 15 \times 3$ cm). The velocity measurements are made at times too short for the shear wave to reflect back from the bottom of the sample and interfere with the measurement. The S-wave pump induces strains of 0.3—2.2×10^{-6}, and we observe changes in the P-wave probe arrival time of up to 100 ns, corresponding to a change in elastic properties of 0.2%. By changing the relative time delay between the probe and pump signals, we record the measured changes in travel time of the P-wave probe to recover the nonlinear parameters $\beta\sim -10^2$ and $\delta \sim\!-10^9$ at room-temperature. This work significantly broadens the applicability of dynamic acousto-elastic testing by utilizing both S and P traveling waves.

8:35

4aPAa3. A case study of multipole acoustic logging in heavy oil sand reservoirs. Peng Liu, Wenxiao Qiao, Xiaohua Che, Ruijia Wang, Xiaodong Ju, and Junqiang Lu (State Key Lab. of Petroleum Resources and Prospecting, China Univ. of Petroleum (Beijing), No. 18, Fuxue Rd., Changping District, Beijing, Beijing 102249, China, liupeng198712@126.com)

The multipole acoustic logging tool (MPAL) was tested in the heavy oil sand reservoirs of Canada. Compared with near shales, the P-wave slowness of heavy oil sands does not change obviously, with the value of about 125μs/ft; the dipole shear slowness decreases significantly to 275μs/ft. The heavy oil sands have a Vp/Vs value of less than 2.4. The slowness and amplitude of dipole shear wave are good lithology discriminators that have great differences between heavy oil sands and shales. The heavy oil sand reservoirs are anisotropic. The crossover phenomenon in the fast and slow dipole shear wave dispersion curves indicates that the anisotropy is induced by unbalanced horizontal stress in the region.

8:50

4aPAa4. Borehole sonic imaging applications. Jennifer Market (Weatherford, 19819 Hampton Wood Dr, Tomball, TX 77377, jennifer.market@weatherford.com)

The advent of azimuthal logging-while-drilling (LWD) sonic tools has opened up a surfeit of near-real time applications. Azimuthal images of compressional and shear velocities allow for geosteering, fracture identification, stress profiling, production enhancement, and 3D wellbore stability analysis. Combining borehole sonic images with electrical, gamma ray, and density images yields a detailed picture of the near- and far-wellbore nature of the stress field and resultant fracturing. A brief review of the physics of azimuthal sonic logging will be presented, paying particular attention to azimuthal resolution and depth of investigation. Examples of combined interpretations of sonic, density, and electrical images will be shown to illustrate fracture characterization, unconventional reservoir completion planning, and geosteering. Finally, recommendations for the optimized acquisition of borehole sonic images will be discussed.

Contributed Papers

9:05

4aPAa5. Numerical simulations of an electromagnetic actuator in a low-frequency range for dipole acoustic wave logging. Yinqiu Zhou, Penglai Xin, and Xiuming Wang (Inst. of Acoust., Chinese Acad. of Sci., 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhouyinqiu@mail.ioa.ac.cn)

In dipole acoustic logging, transducers are required to work in a low frequency range, such as 0.5–5 kHz, to measure shear wave velocities so as to accurately analyze the anisotropy parameters of formations. In this paper, an electromagnetic actuator is designed for more effective low-frequency excitations than conventional piezoelectric bender-bar transducers. A numerical model has been set up to simulate electromagnetic actuators to generate flexural waves. The Finite Element Method (FEM) has been applied to simulating the radiation modes and harmonic responses of the actuator in a fluid, such as air and water. In the frequency range of 0–5 kHz, the first ten vibration modes are simulated and analyzed. The simulation results of 3-D harmonic responses of the sound field, such as the deformation, acoustic sound pressure, and directivity pattern, have been conducted to evaluate the radiation performance. From the simulation results, it is concluded that the second asymmetric mode at 670 Hz could be excited more easily than the others. This oscillated-vibration mode is useful to be applied in a dipole source. The frequency response curve is broad and flat and the electromagnetic actuator is beneficial to generate the wideband signal in a required low frequency range, especially below 1 kHz.

9:20

4aPAa6. Phase moveout method for extracting flexural mode dispersion and borehole properties. Said Assous, David Eccles, and Peter Elkington (GeoSci., Weatherford, Weatherford, East Leake, Loughborough, United Kingdom, david.eccles@eu.weatherford.com)

Among the dispersive modes encountered in acoustic well logging applications is the flexural mode associated with dipole source excitations whose low frequency asymptote provides the only reliable means of determining shear velocity in slow rock formations. We have developed a phase moveout method for extracting flexural mode dispersion curves from with excellent velocity resolution; the method is entirely data-driven, but in combination with a forward model able to generate theoretical dispersion curves, we are able to address the inverse problem and extract formation and borehole properties in addition to the rock shear velocity. The concept is demonstrated using data from isotropic and anisotropic formations.

9:35

4aPAa7. Borehole acoustic array processing methods: A review. Said Assous and Peter Elkington (GeoSci., Weatherford, East Leake, Loughborough LE126JX, United Kingdom, said.assous@eu.weatherford.com)

In this talk, we review the different borehole acoustic array methods and compare their effectiveness with simulated and real waveform examples: Starting from the slowness time coherence (STC) method, weighted semblance method (WSS), and many other common dispersive processing approaches including: Prony's method, maximum entropy (ARMA) methods, and predictive array processing and Matrix pencil technique. We also discuss the Methods include phase minimization or coherency maximization and phase-based approaches.

9:50

4aPAa8. Classifying and removing monopole mode propagating through drill collar. Naoki Sakiyama (Schlumberger K.K., 2-18-3-406, Bessho, Hachio-ji 192-0363, Japan, NSakiyama@slb.com), Alain Dumont (Schlumberger K.K., Kawasaki, Japan), Wataru Izuhara (Schlumberger K.K., Inagi, Japan), Hiroaki Yamamoto (Schlumberger K.K., Kamakura, Japan), Makito Katayama (Schlumberger K.K., Yamato, Japan), and Takeshi Fukushima (Schlumberger K.K., Hachio-ji, Japan)

Understanding characteristics of the acoustic wave propagating through drill collars is important for formation evaluation with logging-while-drilling (LWD) sonic tools. Knowing the frequency-slowness information of different types of the wave propagating through the collar, we can minimize the unwanted wave propagating through the collar by processing and robustly identify formation compressional and shear arrivals. Extensional modes of the steel drill collar are generally dispersive and range from 180 μs/m to 400 μs/m depending on the frequency band. A fundamental torsional mode of the drill collar is nondispersive, but its slowness is sensitive to the geometry of the drill collar. Depending on the geometry and shear modulus of the material, the slowness of the torsional mode can be slower than 330 μs/m. For identifying slowness of the formation arrivals, those different slownesses of the wave propagating through the collar need to be identified separately from those of the wave propagating through formations. Examining various types of the acoustic wave propagating through a drill collar, we determined that the waves can be properly muted by processing for the semblance of waveforms acquired with LWD sonic tools.

10:05–10:20 Panel Discussion

Session 4aPAb

Physical Acoustics: Topics in Physical Acoustics I

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

William Slaton, Cochair
Physics & Astronomy, The University of Central Arkansas, 201 Donaghey Ave, Conway, AR 72034

Contributed Papers

10:30

4aPAb1. Faraday waves on a two-dimensional periodic substrate. C. T. Maki (Phys., Hampden-Sydney College, 1 College Rd., Hampden-Sydney, VA 23943, MakiC15@hsc.edu), Peter Rodriguez, Purity Dele-Oni, Pei-Chuan Fu, and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

A vertically oscillating body of liquid will exhibit Faraday waves when forced above a threshold interface acceleration amplitude. The patterns and their wavelengths at driving frequencies of order 100 Hz are well known in the literature. However, wave interactions influenced by periodic structures on a driving substrate are less well-studied. We report results of a Faraday experiment with a specific periodically structured substrate in the strong coupling regime where the liquid depth is of the order of the structure height. We observe patterns and pattern wavelengths versus driving frequency over the range of 50–350 Hz. These observations may be of interest in situations where Faraday waves appear or are applied.

10:45

4aPAb2. Substrate interaction in ranged photoacoustic spectroscopy of layered samples. Logan S. Marcus, Ellen L. Holthoff, and Paul M. Pellegrino (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-SEE-E, Adelphi, MD 20783, loganmarcus@gmail.com)

Photoacoustic spectroscopy (PAS) is a useful monitoring technique that is well suited for ranged detection of condensed materials. Ranged PAS has been demonstrated using an interferometer as the sensor. Interferometric measurement of photoacoustic phenomena focuses on the measurement of changes in path length of a probe laser beam. That probe beam measures, without discrimination, the acoustic, thermal, and physical changes to the excited sample and the layer of gas adjacent to the surface of the solid sample. For layered samples, the photoacoustic response of the system is influenced by the physical properties of the substrate as well as the sample under investigation. We will discuss the affect that substrate absorption of the excitation source has on the spectra collected in PAS. We also discuss the role that the vibrational modes of the substrate have in photoacoustic signal generation.

11:00

4aPAb3. Difference frequency scattered waves from nonlinear interactions of a solid sphere. Chrisna Nguon (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu), Max Denis (Mayo Clinic, Rochester, MN), Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MA)

In this work, the generation of difference frequency waves arising from the interaction of dual-incident beams on a solid sphere is considered. The high-frequency incident beams induce a radiation force onto the fluid-saturated sphere causing the scatterer to vibrate. An analysis on the contribution between the difference frequency sound and radiation force pressure is of particular interest. The scattered pressure due to the two primary waves are obtained as solutions to the Kirchhoff–Helmholtz integral equation for the fluid–solid boundary. Due to the contrasting material properties between the host fluid and solid sphere, high-order approximations are used to evaluate the integral equation.

11:15

4aPAb4. Effect of surface irregularities on the stability of Stokes boundary. Katherine Aho, Jenny Au, Charles Thompson, and Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave, Lowell, MA 01854, katherine_aho@student.uml.edu)

In this work, we examine that impact that wall surface roughness plays on the stability of an oscillatory Stokes boundary layer. The temporal growth of three-dimensional disturbances excited by wall height variations is of particular interest. Floquet theory is used to identify linearly unstable region in parameter space. It is shown that disturbances become unstable at critical value of the Taylor number for a given surface curvature. The case of oscillatory flow in a two-dimensional rigid walled channel is considered in detail.

11:30

4aPAb5. Novel optoacoustic source for arbitrarily shaped acoustic wavefronts. Weiwei Chan, Yuanxiang Yang, Manish Arora, and Claus-Dieter Ohl (Phys. and Appl. Phys., Nanyang Technolog. Univ., Nanyang Link 21 School of Physical and Mathematical Sci. Nanyang Technolog. University, Singapore 637371, Singapore, chan0700@e.ntu.edu.sg)

We present a novel approach to generate arbitrary acoustic wavefronts using the optoacoustic effect on custom designed PDMS substrates. PDMS blocks are casted into the desired shape with a 3D-printed mold and coated with a layer of an optical absorber. Acoustic wavefront corresponding to the geometry of coated surface is generated by exposing this structure to nano-second laser pulse (Nd:YAG, $\lambda = 532$ nm). For a spherical shell design, pressure pulses of amplitude up to 6.1 bar peak to peak and frequency >30 MHz could be generated. By utilizing other geometries, we focus the acoustic waves from different sections of the transmitter onto a single focal point at different time delay, thus permitting generation of double-peak acoustic pulse from a single laser pulse. Further modification of the structure permits designing of multi-foci, multi-peak acoustic pulses from a single optical pulse.

11:45

4aPAb6. Accuracy of local Kramers–Kronig relations between material damping and dynamic elastic properties. Tamas Pritz (Budapest Univ. of Technol. and Economics, Apostol u 23, Budapest 1025, Hungary, tampri@eik.bme.hu)

The local Kramers–Kronig (KK) relations are the differential form approximations of the general KK integral equations linking the damping properties (loss modulus or loss factor) and dynamic modulus of elasticity (shear, bulk, etc.) of linear solid viscoelastic materials. The local KK

relations are not exact; therefore, their accuracy is known to depend on the rate of frequency variations of material dynamic properties. The accuracy of the local KK relations is investigated in this paper under the assumption that the frequency dependence of the loss modulus obeys a simple power law. It is shown by analytic calculations that the accuracy of prediction of the local KK relations is better than 10% if the exponent in the loss modulus-frequency function is smaller than 0.35. This conclusion supports the result of an earlier numerical study. Some experimental data verifying the theoretical results will be presented. The conclusions drawn in the paper can easily be extended to acoustic wave propagation, namely to the accuracy of local KK relations between attenuation and dispersion of phase velocity.

THURSDAY MORNING, 30 OCTOBER 2014 MARRIOTT 1/2, 8:30 A.M. TO 12:00 NOON

Session 4aPP

Psychological and Physiological Acoustics: Physiological and Psychological Aspects of Central Auditory Processing Dysfunction I

Frederick J. Gallun, Cochair
National Center for Rehabilitative Auditory Research, Portland VA Medical Center, 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Adrian KC Lee, Cochair
Box 357988, University of Washington, Seattle, WA 98195

Chair's Introduction—8:30

Invited Papers

8:35

4aPP1. Auditory processing disorder: Clinical and international perspective. David R. Moore (Commun. Sci. Res. Ctr., Cincinnati Children's Hospital, 240 Albert Sabin Way, Rm. S1.603, Cincinnati, OH 45229, david.moore2@cchmc.org)

APD may be considered a developmental, hearing or neurological disorder, depending on etiology, but in all cases, it is a listening difficulty without an abnormality of pure tone sensitivity. It has been variously defined as a disorder of the central auditory system associated with impaired spatial hearing, auditory discrimination, temporal processing, and performance with competing or degraded sounds. Clinical testing typically examines perception, intelligibility and ordering of both speech and non-speech sounds. While deficits in higher-order cognitive, communicative, and language functions are excluded in some definitions, recent consensus accepts that these functions may be inseparable from active listening. Some believe that APD in children is predominantly or exclusively cognitive in origin, while others insist that true APD has its origins within the auditory brainstem. However, children or their carers presenting at clinics typically complain of difficulty hearing speech in noise, remembering or understanding instructions, and attending to sounds. APD usually occurs alongside other developmental disorders (e.g., language impairment) and may be indistinguishable from them. Consequently, clinicians are uncertain how to diagnose or manage APD; both test procedures and interventions vary widely, even within a single clinic. Effective remediation primarily consists of improving the listening environment and providing communication devices.

9:05

4aPP2. Caught in the middle: The central problem in diagnosing auditory-processing disorders in adults. Larry E. Humes (Indiana Univ., Dept. Speech & Hearing Sci., Bloomington, IN 47405-7002, humes@indiana.edu)

It is challenging to establish the existence of higher-level auditory-processing disorders in military veterans with mild Traumatic Brain Injury (TBI). Yet, mild TBI appears to be a highly prevalent disorder among U.S. veterans returning from recent military conflicts in Iraq and Afghanistan. Recent prevalence estimates for mild TBI, for example, among these military veterans have suggest a rate of 7–9% [Carlson, K.F. *et al*. (2011), "Prevalence, assessment and treatment of mild Traumatic Brain Injury an Posttraumatic Stress Disorder: a systematic review of the evidence," J. Head Trauma Rehabil., 26, 103–115]. A key factor in diagnosing central components for auditory-processing disorders may lie in the potentially confounding influences of concomitant peripheral auditory and cognitive dysfunction in many veterans with TBI. This situation is strikingly similar to that observed in many older adults. Many older adults, for example, exhibit peripheral hearing loss and typical cognitive-processing deficits often associated with healthy aging. These concomitant problems make the diagnosis of centrally located auditory-processing problems in older adults extremely difficult. After building a case for many similarities between young veterans with mild TBI and older adults with presbycusis, this presentation will focus on several of the lessons learned from research with older adults. [Work supported, in part, by research grant R01 AG008293 from the NIA.]

9:35

4aPP3. Lack of a coherent theory limits the diagnosis and prognostic value of the central auditory processing disorder. Anthony T. Cacace (Commun. Sci. & Disord., Wayne State Univ., 207 Rackham, 60 Farnsworth, Detroit, MI 48202, cacacea@wayne.edu) and Dennis J. McFarland (Lab. of Neural Injury and Repair, Wadsworth Labs, NYS Health Dept., Albany, NY)

Spanning almost 6 decades, CAPD, defined as a modality specific perceptual dysfunction not due to peripheral hearing loss, still remains controversial and requires further development if it is to become a useful clinical entity. Early attempts to quantify the effects of central auditory nervous system lesions based on the use of filtered-speech material, dichotic presentation of digits, and various non-speech tests have generally been abandoned due to lack-of-success. Site-of-lesion approaches have given way to functional considerations whereby attempts to understand underlying processes, improve specificity-of-diagnosis, and delineate modality-specific (auditory) disorders from "non-specific supramodal dysfunctions" like those related to attention and memory have begun to fill the gap. Furthermore, because previous work was generally limited to auditory tasks alone, functional dissociations could not be established and consequently, the need to show the modality-specific nature of the observed deficits has been compromised; further limiting progress in this area. When viewed as a whole, including information from consensus conferences, organizational guidelines, representative studies, etc., what is conspicuously absent is a well-defined theory that permeates all areas of this domain, including the neural substrates of auditory processing. We will discuss the implications of this shortcoming and propose ways to move forward in a meaningful manner.

10:05–10:30 Break

10:30

4aPP4. Cochlear synaptopathy and neurodegeneration in noise and aging: Peripheral contributions to auditory dysfunction with normal thresholds. Sharon G. Kujawa (Dept. of Otology and Laryngology, Harvard Med. School and Massachusetts Eye and Ear Infirmary, Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, sharon_kujawa@meei.harvard.edu)

Declining auditory performance in listeners with normal audiometric thresholds is often attributed to changes in central circuits, based on the widespread view that normal thresholds indicate a lack of cochlear involvement. Recent work in animal models of noise and aging, however, demonstrates that there can be functionally important loss of sensory inner hair cell—afferent fiber communications that go undetected by conventional threshold metrics. We have described a progressive cochlear synaptopathy that leads to proportional neural loss with age, well before loss of hair cells or age-related changes in threshold sensitivity. Similar synaptic and neural losses occur after noise, even when thresholds return to normal. Since the IHC-afferent fiber synapse is the primary conduit for information to flow from the cochlea to the brain, and since each of these cochlear nerve fibers makes synaptic contact with one inner hair cell only, these losses should have significant perceptual consequences, even if thresholds are preserved. The prevalence of such pathology in the human is likely to be high, underscoring the importance of considering peripheral status when studying central contributions to auditory performance declines. [Research supported by R01 DC 008577 and P30 DC 05029.]

11:00

4aPP5. Quantifying supra-threshold sensory deficits in listeners with normal hearing thresholds. Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu), Hari Bharadwaj, Inyong Choi, Hannah Goldberg (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), Salwa Masud, and Golbarg Mehraei (Speech and Hearing BioSci. and Technol., Harvard/MIT, Boston, MA)

There is growing suspicion that some listeners with normal-hearing thresholds may be suffering from a specific form of sensory deficit—a loss of afferent auditory nerve fibers. We believe such deficits manifest behaviorally in conditions where perception depends upon precise spectro-temporal coding of supra-threshold sound. In our lab, we find striking inter-subject differences in perceptual ability even among listeners with normal hearing thresholds who have no complaints of hearing difficulty and have never sought clinical intervention. Among such ordinary listeners, those who perform relatively poorly on selective attention tasks (requiring the listener to focus on one sound stream presented amidst competing sound streams) also exhibit relatively weak temporal coding in subcortical responses and have poor thresholds for detecting fine temporal cues in supra-threshold sound. Here, we review the evidence for supra-threshold hearing deficits and describe measures that reveal this sensory loss. Given our findings in ordinary adult listeners, it stands to reason that at least a portion of the listeners who are diagnosed with central auditory processing dysfunction may suffer from similar sensory deficits, explaining why they have trouble communicating in many everyday social settings.

11:30

4aPP6. Neural correlates of central auditory processing deficits in the auditory midbrain in an animal model of age-related hearing loss. Joseph P. Walton (Commun. Sci. and Disord., Univ. of South Florida, 4202 Fowler Ave., PCD 1017, Tampa, FL 33620, jwalton1@usf.edu)

Age-related hearing loss (ARHL), clinically referred to as presbycusis, affects over 10 million Americans and is considered to be the most common communication disorder in the elderly. Presbycusis can be associated with at least two underlying etiologies, a decline in cochlear function resulting in sensorineural hearing loss, and deficits in auditory processing within the central auditory system. Previous psychoacoustic studies have revealed that aged human listeners display deficits in temporal acuity that worsen with the addition of background noise. Spectral and temporal acuity is essential for following the rapid changes in frequency and intensity that comprise most natural sounds including speech. The perceptual analysis of complex sounds depends to a large extent on the ability of the auditory system to follow and even sharpen neural encoding of rapidly changing acoustic signals, and the inferior colliculus (IC) is a key auditory nucleus involved in temporal and spectral processing. In this talk, I will review neural correlates of temporal and signal-in-noise processing at the level of the auditory midbrain in an animal model of ARHL. Understanding the neural substrate of these perceptual deficits will assist in its diagnosis and rehabilitation, and be crucial to further advances in the design of hearing aids and therapeutic interventions.

Session 4aSCa

Speech Communication: Subglottal Resonances in Speech Production and Perception

Abeer Alwan, Cochair
Dept. of Electrical Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095

Steven M. Lulich, Cochair
Speech and Hearing Sciences, Indiana University, 4789 N White River Drive, Bloomington, IN 47404

Mitchell Sommers, Cochair
Psychology, Washington University, Campus Box 1125, 1 Brookings Drive, Saint Louis, MO 63130

Chair's Introduction—8:00

Invited Papers

8:05

4aSCa1. The role of subglottal acoustics in speech production and perception. Mitchell Sommers (Indiana Univ., Saint Louis, MS), Abeer Alwan (Psych., Washington Univ., Los Angeles, CA), and Steven Lulich (Psych., Washington Univ., Dept. of Speech and Hearing Sci., Indiana University, Bloomington, IN, slulich@indiana.edu)

In this talk, we present an overview of subglottal acoustics, with emphasis on the significant anatomical structures that define subglottal resonances, and we present results from our experiments incorporating subglottal resonances into automatic speaker normalization and speech recognition technologies. Speech samples used in the modeling and perception studies were obtained from a new speech corpus (the UCLA-WashU subglottal database) of simultaneous microphone and (subglottal) accelerometer recordings of 50 adult speakers of American English (AE). We will discuss new findings about the Young's Modulus of tracheal soft tissue, the viscosity of tracheal cartilage, and the effect of going from a circular cross-section to a rectangular cross-section in the conus elasticus. We also present results from studies demonstrating a small, but significant, role of subglottal resonances in discriminating speaker height and of the interaction between subglottal resonances and formants in height discrimination.

8:25

4aSCa2. The effect of subglottal acoustics on vocal fold vibration. Ingo R. Titze (National Ctr. for Voice and Speech, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@ncvs2.org) and Ingo R. Titze (Dept. of Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

Acoustic pressures above and below the vocal folds produce a push-pull action on the vocal folds which can either help or hinder vocal fold vibration. The key variable is acoustic reactance, the energy-storage part of the complex acoustic impedance. For the subglottal airway, inertive (positive) reactance does not help vocal fold vibration, but helps to skew the glottal airflow waveform for high frequency harmonic excitation. Compliant (negative) reactance, on the contrary, helps vocal fold vibration but does not skew the waveform. Thus, the benefit of subglottal reactance is mixed. For supraglottal reactance, the benefit is additive. Inertive supraglottal reactance helps vocal fold vibration and skews the waveform, whereas compliant supraglottal reactance does neither. The effects will be demonstrated with source-filter interactive simulation.

8:45

4aSCa3. Impact of subglottal resonances on bifurcations and register changes in laboratory models of phonation. David Berry, Juergen Neubauer, and Zhaoyan Zhang (Surgery, UCLA, 31-24 Rehab, Los Angeles, CA 90095-1794, daberry@ucla.edu)

Many laboratory studies of phonation have failed to fully specify the subglottal system employed during research. Many of these same studies have reported a variety of nonlinear phenomena, such as bifurcations and vocal register changes. While such phenomena are often presumed to result from changes in the biomechanical properties of the larynx, such phenomena may also be a manifestation of coupling between the voice source and the subglottal tract. Using laboratory models of phonation, a variety of examples will be given of nonlinear phenomena induced by both laryngeal and subglottal mechanisms. Moreover, using tracheal tube lengths commonly reported in the literature, it will be shown that most of the nonlinear phenomena commonly reported in voice production may be replicated solely on the acoustical resonances of the subglottal system. Finally, recommendations will be given regarding the experimental design of laboratory experiments which may allow laryngeally induced bifurcations to be distinguished from subglottally induced bifurcations.

9:05–9:25 Break

9:25

4aSCa4. Subglottal ambulatory monitoring of vocal function to improve voice disorder assessment. Robert E. Hillman, Daryush Mehta, Jarrad H. Van Stan (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, One Bowdoin Square, 11th Fl., Boston, MA 02114, daryush.mehta@alum.mit.edu), Matias Zanartu (Dept. of Electron. Eng., Universidad Tecnica Federico Santa Mariıa, Valparaiso, Chile), Marzyeh Ghassemi, and John V. Guttag (Comput. Sci. and Artificial Intelligence Lab., Massachusetts Inst. of Technol., Cambridge, MA)

Many common voice disorders are chronic or recurring conditions that are likely to result from inefficient and/or abusive patterns of vocal behavior, referred to as vocal hyperfunction. The clinical management of hyperfunctional disorders would be greatly enhanced by the ability to monitor and quantify detrimental vocal behaviors during an individual's activities of daily life. This presentation will provide an update about ongoing work that is using a miniature accelerometer on the subglottal neck surface to collect a large set of ambulatory data on patients with hyperfunctional voice disorders (before and after treatment) and matched control subjects. Three types of analysis approaches are being employed in an effort to identify the best set of measures for differentiating among hyperfunctional and normal patterns of vocal behavior: (1) previously developed ambulatory measures of vocal function that include vocal dosages; (2) measures based on estimates of glottal airflow that are extracted from the accelerometer signal using a vocal system model, and (3) classification based on machine learning approaches that have been used successfully in analyzing long-term recordings of other physiologic signals (e.g., electrocardiograms).

9:45

4aSCa5. Do subglottal resonances lead to quantal effects resulting in the features [back] and [low]?: A review. Helen Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu) and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

A question of general interest is why languages have the sound categories that they do. K. N. Stevens proposed the Quantal Theory of phonological contrasts, suggesting that regions of discontinuity in the articulatory-acoustic mapping serve as category boundaries. H. M. Hanson and K. N. Stevens [Proc. ICPhS, 182–185, 1995] modeled the interaction of subglottal resonances with the vocal-tract filter, showing that when a changing supraglottal formant strays into the territory of a stationary tracheal formant, a discontinuity in supraglottal formant frequency and attenuation of the formant peak occurs. They suggested that vowel space and quality could thus be affected. K. N. Stevens [*Acoustic Phonetics*, MIT Press, 1998] went further, musing that because the first and second subglottal resonances lead to instabilities in supraglottal formant frequency and amplitude, vowel systems would benefit by avoiding vowels with formants at these frequencies. Avoiding the first subglottal resonance would naturally lead to the division of vowels into those with a low vs. non-low tongue body; avoiding the second would lead to the division of vowels into those having a back vs. front tongue body. We will review subsequent research that offers substantial support for this hypothesis, justifying inclusion of the effects of subglottal resonances in phonological models.

Contributed Paper

10:05

4aSCa6. Relationship between lung volumes and subglottal resonances. Natalie E. Duvanenko (Speech and Hearing Sci., Indiana Univ., 2416 Cibuta Court, West Lafayette, IN 47906, nduvanen@umail.iu.edu) and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Subglottal resonances are dependent the anatomical structure of the lungs, but efforts to detect changes in subglottal resonances throughout an utterance have failed to show any effect of lung volume. In this study, we present the results of an experiment investigating the relationship between lung volumes and subglottal resonances. The pulmonary subdivisions for several speakers were established using a whole-body plethysmograph. Subsequently, lung volume and subglottal resonances were recorded simultaneously using a spirometer and an accelerometer while the speakers produced long sustained vowels.

Session 4aSCb

Speech Communication: Learning and Acquisition of Speech (Poster Session)

Maria V. Kondaurova, Chair
Otolaryngology – Head & Neck Surgery, Indiana University School of Medicine, 699 Riley Hospital Drive – RR044, Indianapolis, IN 46202

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, the contributors of odd-numbered papers will be at their posters from 8:00 a.m. and 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

8:00

4aSCb1. Labels facilitate the learning of competing abstract perceptual mappings. Shannon L. Heald, Nina Bartram, Brendan Colson, and Howard C. Nusbaum (Psych., Univ. of Chicago, 5848 S. University Ave., B406, Chicago, IL 60637, smbowdre@uchicago.edu)

Listeners are able to quickly adapt to synthetic speech, even though it contains misleading and degraded acoustic information. Previous research has shown that testing and training on a given synthesizer using only novel words leads listeners to form abstract or generalized knowledge for how that particular synthesizer maps different acoustic patterns onto their pre-existing phonological categories. Prior to consolidation, this knowledge has been shown to be susceptible to interference. Given that labels have been argued to stabilize abstract ideas in working memory and to help learners form category representations that are robust against interference, we examined how learning for a given synthesizer is affected by labeled or unlabeled immediate training on an additional synthesizer, which uses a different acoustic to phonetic mapping. We demonstrated that the learning of an additional synthesizer interferes with the retention of a previously learned synthesizer but that this is ameliorated if the additional synthesizer is labeled. Our findings indicate that labeling may be important in facilitating daytime learning for competing abstract perceptual mappings prior to consolidation and suggests that speech perception may be best understood through the lens of perceptual categorization.

4aSCb2. When more is not better: Variable input in the formation of robust word representations. Andrea K. Davis (Linguist, Univ. of Arizona, 1076 Palomino Rd., Cloverdale, CA 95425, davisak@email.arizona.edu) and LouAnn Gerken (Linguist, Univ. of Arizona, Tucson, AZ)

A number of studies with infants and with young children suggest that hearing words produced by multiple talkers helps learners to develop more robust word representations (Richtsmeier *et al.*, 2009; Rost & McMurray, 2009, 2010). Native adult learners, however, do not seem to derive the same benefit from multiple talkers. A word-learning study with native adults was conducted, and a second study with second language learners will have been completed by this fall. Native-speaking participants learned four new minimal English-like minimal pair words either from a single talker or from multiple talkers. They were then tested with (a) a perceptual task, in which they saw the two pictures corresponding to a minimal pair, heard one of the pair, and had to choose the picture corresponding to the word they heard; (b) a speeded production task, in which they had to repeat the words they had just learned as quickly as possible. Unlike infants, the two groups did not differ significantly in perceptual accuracy. However, the single talker group had significantly higher variance in the speeded production task. It is hypothesized that this greater variance is due to individual differences in learning strategies, which are masked when learning from multiple talkers.

4aSCb3. A comparison of acoustic and perceptual changes in children's productions of American English /r/. Sarah Hamilton (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Casey Keck (Commun. Sci. and Disord., Univ. of Cincinnati, 408 Glengarry Way, Fort Wright, KY 41011, stewarce@mail.uc.edu), and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Speech-language pathologists rely primarily on their perceptual judgments when evaluating whether children have made progress in speech sound therapy. Speech sound perception in normal listeners has been characterized as largely categorical, such that slight articulatory changes may go unnoticed unless they reach a specific acoustic signature assigned to a different category. While perception may be categorical, acoustic phenomena are largely measured in continuous units, meaning that there is a potential mismatch between the two methods of recording change. Clinicians, using perceptual categorization, commonly report that some children make no progress in therapy, yet acoustically, the children's productions may be shifting toward acceptable acoustic characteristics. Using subtle changes in the acoustic signal during therapy could potentially prevent these clients from being discharged due to a perceived lack of progress. This poster evaluates acoustic changes compared to perceptual changes in children's productions of the American English phoneme /r/ after receiving speech therapy using ultrasound supplemented with telepractice home practice. Preliminary data indicate that there are significant differences between participants' acoustic values of /r/ and perceptual ratings by clinicians.

4aSCb4. Perceptual categorization of /r/ for children with residual sound errors. Sarah M. Hamilton, Suzanne Boyce, and Lindsay Mullins (Commun. Sci. and Disord., Univ. of Cincinnati, 3433 Clifton Ave., Cincinnati, OH 45220, hamilsm@mail.uc.edu)

Many studies have found that children with resistant speech sound errors (RSSD) show (1) atypical category boundaries, and (2) difficulty identifying whether their own productions are correct or misarticulated. Historically, perceptual category discrimination tests use synthesized speech representing incremental change along an acoustic continuum, while tests of a child's self-perception are confined to categorical correct vs. error choices. Thus, it has not been possible to explore the boundaries of RSSD children's categorical self-perception in any detail or to customize perceptual training for therapeutic purposes. Following an observation of Hagiwara (1995), who noted that typical speakers show F3 values for /r/ between 80% and 60% of their average vowel F3, Hamilton et al (2014) found that this threshold largely replicates adult listener judgments, such that productions above and below the 80% threshold sounded consistently "incorrect" or "correct," but that productions closest to the 80% threshold were given more ambiguous judgments. In this study, we apply this notion of an F3 threshold to investigate whether children with RSD respond like adult listeners when presented with natural-speech stimuli along a continuum of correct and incorrect /r/. Preliminary results indicate that children with RSD do not make adult-like decisions when categorizing /r/ productions.

4aSCb5. A child-specific compensatory mechanism in the acquisition of English /s/. Hye-Young Bang, Meghan Clayards, and Heather Goad (Linguist, McGill Univ., 1085 Dr. Penfield, Montreal, QC H3A 1A7, Canada, hye-young.bang@mail.mcgill.ca)

This study examines corpus data involving word-initial [sV] productions from 79 children aged 2–5 (Edwards & Beckman 2008) in comparison with a corpus of word-initial [sV] syllables produced by 13 adults. We quantified target-like /s/ production using spectral moment analysis on the frication portion (high center of gravity, low SD, and low skewness). In adults, we found that higher vowels (low F1 after normalization) were associated with more target-like /s/ productions, likely reflecting a tighter constriction. In children, older subjects produced more target-like outputs overall. However, unlike adults, children's outputs before low vowels were more target-like, regardless of age. This is unexpected given the articulatory challenges of producing /s/ in low vowel contexts. Further investigation found that high F1 (low vowels) was associated with louder /s/ (relative to V) and more encroachment of sibilant noise on the following vowel (high harmonics-to-noise ratio). This finding suggests that young children may be increasing airflow during /s/ production to compensate for a less tight constriction when the jaw must lower for the following vowel. Thus, children may adopt a more accessible mechanism, different from adults, to compensate for their immature lingual gestures, possibly in an attempt to maximize phonological contrasts in word-initial position.

4aSCb6. Moving targets and unsteady states: "Shifting" productions of sibilant fricatives by young children. Patrick Reidy (Dept. of Linguist, The Ohio State Univ., 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, patrick.francis.reidy@gmail.com)

The English voiceless sibilant /s/–/ʃ/ contrast is one that many children do not acquire until their adolescent years. This protracted acquisition may be due to the high level of articulatory control that is necessary to the successful production of an adult-like sibilant, which involves the coordination of lingual, mandibular, and pulmonic gestures. Poor coordination among these gestures can result in the acoustic properties of the noise source or the vocal tract filter changing throughout the timecourse of the frication, to the extent that the phonetic percept of the frication noise changes across its duration. The present study examined such "shifting" productions of sibilant fricatives by native English-acquiring two- through five-year-old children, which were identified from the Paidologos corpus as those productions where the interval of frication was transcribed phonetically as a sequence of fricative sounds. There were two types of shift in frication quality: (1) a gradual change in the resonant frequencies in the spectrogram, suggesting a repositioning of the oral constriction; and (2) an abrupt change in the level of the frication, suggesting a switch in the noise source. Work is underway to develop measures that differentiate these two types of shift, and that suggest their underlying articulatory causes.

4aSCb7. Effects of spectral smearing on sentence recognition by adults and children. Joanna H. Lowenstein (Otolaryngology-Head & Neck Surgery, Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, lowenstein.6@osu.edu), Eric Tarr (Audio Eng. Technol., Belmont Univ., Nashville, TN), and Susan Nittrouer (Otolaryngology-Head & Neck Surgery, Ohio State Univ., Columbus, OH)

Children's speech perception depends on dynamic formant patterns more than that of adults. Spectral smearing of formants, as found with the broadened auditory filters associated with hearing loss, should disproportionately affect children because of this greater dependence on formant patterns. Making formants more prominent, on the other hand, may result in improved recognition. Adults (40) and children age 5 and 7 (20 of each age) listened to 75 four-word syntactically correct, semantically anomalous sentences processed so that excursions around the mean spectral slope were sharpened by 50% (making individual formants more prominent), flattened by 50% (smearing individual formants), or left unchanged. These sentences were presented to children and to half of the adults in speech-shaped noise at 0 dB SNR. The rest of the adults listened to the sentences at -3 dB SNR. Results indicate that all listeners did more poorly with the smeared formants, with 5-year-olds showing the largest decrement in performance at 0 dB SNR. However, adults at -3 dB SNR showed an even greater decrement in performance. Making formants more prominent did not improve recognition, perhaps due to harmonic-formant mismatches. Thus, there is reason to explore processing strategies that might enhance formant prominence for listeners with hearing loss.

4aSCb8. Acoustic-phonetic characteristics of older children's spontaneous speech in interactions in conversational and clear speaking styles. Valerie Hazan, Michèle Pettinato, Outi Tuomainen, and Sonia Granlund (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

This study investigated (a) the acoustic-phonetic characteristics of spontaneous speech produced by talkers aged 9–14 years in an interactive (diapix) task with an interlocutor of the same age and gender (NB condition) and (b) the adaptations these talkers made to clarify their speech when speech intelligibility was artificially degraded for their interlocutor (VOC condition). Recordings were made for 96 child talkers (50 F, 46 M); the adult reference values came from the LUCID corpus recorded under the same conditions [Baker and Hazan, J. Acoustic. Soc. Am. 130, 2139–2152 (2011)]. Articulation rate, pause frequency, fundamental frequency, vowel area, and mean intensity (1–3 kHz range) were analyzed to establish whether they had reached adult-like values and whether young talkers showed similar clear speech strategies as adults in difficult communicative situations. In the NB condition, children (including the 13–14 year group) differed from adults in terms of their articulation rate, vowel area, median F0, and intensity. Child talkers made adaptations to their speech in the VOC condition, but adults and children differed in their use of F0 range, vowel hyperarticulation, and pause frequency as clear speech strategies. This suggests that further developments in speech production take place during later adolescence. [Work supported by ESRC.]

4aSCb9. Acoustic characteristics of infant-directed speech to normal-hearing and hearing-impaired twins with hearing aids and cochlear implants: A case study. Maria V. Kondaurova, Tonya R. Bergeson-Dana (Otolaryngol. – Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr. – RR044, Indianapolis, IN 46202, mkondaur@iupui.edu), and Neil A. Wright (The Richard and Roxelyn Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

The study examined acoustic characteristics of maternal speech to normal-hearing (NH) and hearing-impaired (HI) twins who received hearing aids (HAs) or a unilateral cochlear implant (CI). A mother of female-male NH twins (NH-NH; age 15.8 months), a mother of two male twins, one NH and another HI with HAs (NH-HA; age 11.8 months) and a mother of a NH female twin and a HI male twin with a CI (NH-CI; age 14.8 months) were recorded playing with their infants during three sessions across a 12-month period. We measured pitch characteristics (normalized F0 mean, F0 range, and F0 SD), utterance and pause duration, syllable number, and speaking rate. ANOVAs demonstrated that speech to NH-NH twins was characterized by lower, more variable pitch with greater pitch range as compared to speech to NH-HA and NH-CI pairs. Mothers produced more syllables, had faster speaking rate and longer utterance duration in speech to NH-NH than the other pairs. The results suggest that the pediatric hearing loss in one sibling affects maternal speech properties to both NH and HI infants in the same pair. Future research will investigate vowel space and lexical properties of IDS to three twin pairs as well as their language outcome measures.

4aSCb10. Effects of vowel position and place of articulation on voice onset time in children: Longitudinal data. Elaine R. Hitchcock (Dept. of Commun. Sci. and Disord., Montclair State Univ., 1515 BRd. St., Bloomfield, NJ 07444, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Dept. of Commun. Sci. and Disord., Long Island Univ., Queens, NY)

Voice onset time (VOT) has been found to vary according to phonetic context, but past studies report varying magnitudes of effect, and no past work has evaluated the degree to which such effects are consistent over time for a single speaker. This study explores the relationships between vowel position, consonant place of articulation [POA], and voice onset time (VOT) in children, comparing the results to past adult work. VOT in CV/CVC words was measured in nine children ages 5;3–7;6 every two-four

weeks for 10 months, for a total of 18 sessions yielding approximately 18,000 tokens for analysis. Bilabial and velar cognate pairs targeted a front-back vowel difference (/i/-/u/, /e/-/o/), while alveolar cognate pairs targeted a mid high-low vowel difference (/o/-/ɑ/). VOT variability over time was also evaluated. Preliminary results suggest that POA yields a robust pattern of bilabial < alveolar < velar, but vowel effects are less clear. Vowel height shows the most obvious effect with consistently longer VOT values observed for mid high vowels. Front-back vowel comparisons yielded no obvious differences. On the whole, contextual variations based on POA and vowel context do not show clear correlations with overall VOT variation.

4aSCb11. Longitudinal data on the production of content versus function words in children's spontaneous speech. Jeffrey Kallay and Melissa A. Redford (Linguist, Univ. of Oregon, 1455 Moss St., Apt. 215, Eugene, Ohio 97403, jkallay@uoregon.edu)

Allen and Hawkins (1978; 1980) were among the first to note rhythmic differences in the speech of children and adults. Sirsa and Redford (2011) found that rhythmic differences between younger and older children's speech was best accounted for by age-related differences in function word production. In other on-going work (Redford, Kallay & Dilley) we found an effect of age on the perceived prominence of function words in children's speech, but no effect on content words. The current longitudinal study investigated the effect of word class (content versus function words) on the development of reduction in terms of syllable duration and pitch range (a correlate of accenting). Spontaneous speech was elicited for 3 years from 36 children aged 5; 2–6; 11 at time of first recording. There were effects of word class (content > function) and of time on median duration, but no interaction between these factors. The median duration decreased 13% in function words from the 1st to 3rd year; a similar decrease (15%) was found for content words. Pitch range only varied systematically with word class. Other spectral measures are being collected to further investigate the development of reduction in children's speech. [Work supported by NICHD.]

4aSCb12. Audiovisual speech integration development at varying levels of perceptual processing. Kaylah Lalonde (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, klalonde@indiana.edu) and Rachael Frush Holt (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

There are multiple mechanisms of audiovisual (AV) speech integration with independent maturational time courses. This study investigated development of both basic perceptual and speech-specific mechanisms of AV speech integration by examining AV speech integration development across three levels of perceptual processing. Twenty-two adults and 24 6- to 8-year-old children completed three auditory-only and AV yes/no tasks varying only in the level of perceptual processing required to complete them: detection, discrimination, and recognition. Both groups demonstrated benefits from matched AV speech and interference from mismatched AV speech relative to auditory-only conditions. Adults, but not children, demonstrated greater integration effects at higher levels of perceptual processing (i.e., recognition). Adults seem to rely on both general perceptual mechanisms of speech integration that apply to all levels of perceptual processing and speech-specific mechanisms of integration that apply when making phonetic decisions and/or accessing the lexicon; 6- to 8-year-old children seem to rely only on general perceptual mechanisms of AV speech integration. The general perceptual mechanism allows children to attain the same degree of AV benefit to detection and discrimination as adults, but the lack of a speech-specific mechanism in children might explain why they attain less AV recognition benefit than adults.

4aSCb13. Developmental and linguistic factors of audiovisual speech perception across different masker types. Rachel Reetzke, Boji Lam, Zilong Xie, Li Sheng, and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas at Austin, The University of Texas at Austin, 2504A Whitis Ave., Austin, TX 78751, rreetzke@gmail.com)

Developmental and linguistic factors have been found to influence listeners' ability to recognize speech-in-noise. However, there is paucity of evidence exploring how these factors modulate speech perception in everyday listening situations, such as multisensory environments and backgrounds with informational maskers. This study assessed sentence recognition for 30 children (14 monolingual, 16 simultaneous bilingual; ages 6–10) and 31 adults (21 monolingual, ten simultaneous bilingual; ages 18–22). Our experimental design included three within-subject variables: (a) masker type: pink noise or two-talker babble, (b) modality: audio-only and audiovisual, and (c) signal-to-noise ratio (SNR): 0 to -16 dB. Results revealed that across both modalities and noise types, adults performed better than children, and simultaneous bilinguals performed similarly to monolinguals. The age effect was largest at the lowest SNRs of -12 and -16 dB in the audiovisual two-talker babble condition. These findings suggest that children experience greater difficulty in segregation of target speech in informational maskers relative to adults, even with audiovisual cues. This may provide evidence for children's less developed higher-level cognitive strategies in dealing with speech-in-noise (e.g., selective attention). Findings from the second analysis suggest that despite two competing lexicons, simultaneous bilinguals do not experience a speech perception-in-noise deficit relative to monolinguals.

4aSCb14. Experience-independent effects of matching and non-matching visual information on speech perception. D. Kyle Danielson, Alison J. Greuel, Padmapriya Kandhadai, and Janet F. Werker (Psych., Univ. of Br. Columbia, 2136 West Mall, Vancouver, BC V6T 1Z4, Canada, kdanielson@psych.ubc.ca)

Infants are sensitive to the correspondence between visual and auditory speech. Infants exhibit the McGurk effect, and matching audiovisual information may facilitate discrimination of similar consonant sounds in an infant's native language (e.g., Teinonen *et al.*, 2008). However, because most existing research in audiovisual speech perception has been conducted using native speech sounds with infants in their first year of life, little work has explored whether this link between the auditory and visual modalities of speech perception arises due to experience with the native language. In the present set of studies, English-learning six- and ten-month-old infants are tested for discrimination of a non-English speech contrast following familiarization with matching and mismatching audiovisual speech. Furthermore, the looking fixation behaviors of the two age groups are compared between the two conditions. Although it has been demonstrated that infants in the younger age range attend preferentially to the eye region when viewing matched audiovisual speech and that infants in the older age range temporarily attend to the mouth region (Lewkowicz & Hansen-Tift, 2012), here deviations in this behavior for matching and mismatching non-native speech are examined (a link that has only been previously explored in the native language (Tomalski *et al.*, 2013)).

4aSCb15. Switched-dominance bilingual speech production: Continuous usage versus early exposure. Michael Blasingame and Ann R. Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, mblasingame@u.northwestern.edu)

Switched dominance bilinguals (i.e., "heritage speakers," HS, with L2 rather than L1 dominance) have exhibited native-like heritage language (L1) sound perception (e.g., Korean three-way VOT contrast discrimination by Korean HS; Oh, Jun, Knightly, & Au, 2003) and sound production (e.g., Spanish VOT productions by Spanish HS; Au, Knightly, Jun, & Oh, 2002), but far from native-like proficiency in other aspects of L1 function, including morphosyntax (Montrul, 2010). We investigated whether native-like L1 sound production proficiency extended to heritage language sentence-in-noise intelligibility. We recorded English and Spanish sentences by Spanish HS (SHS) and monolingual English controls (English only). Native listeners of each language transcribed these recordings under easy (-4 dB SNR) and hard (-8 dB SNR) conditions. In easy conditions, SHS English and Spanish intelligibility were not significantly different, yet in hard conditions, SHS English intelligibility was significantly higher than SHS Spanish intelligibility. Furthermore, we observed no differences between SHS English and English-control intelligibility in both conditions. These results suggest for SHS, while early Spanish exposure provided some resistance to heritage language/L1 intelligibility degradation, the absence of continuous Spanish usage impacted intelligibility in severely degraded conditions. In contrast, the absence of early English exposure was entirely overcome by later English dominance.

4aSCb16. Genetic variation in catechol-O-methyl transferase activity impacts speech category learning. Han-Gyol Yi (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504 Whitis Ave., A1100, Austin, TX 78712, gyol@utexas.edu), W. T. Maddox (Psych., The Univ. of Texas at Austin, Austin, TX), Valerie S. Knopik (Behavioral Genetics, Rhode Island Hospital, Providence, RI), John E. McGeary (Providence Veterans Affairs Medical Ctr. , Providence, RI), and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

Learning non-native speech categories is a challenging task. Little is known about the neurobiology underlying speech category learning. In vision, two dopaminergic neurobiological learning systems have been identified: a rule-based reflective learning system mediated by the prefrontal cortex, wherein processing is under deliberative control, and an implicit reflexive learning system mediated by the striatum. During speech learning, successful learners initially use simple reflective rules but eventually transition to a multidimensional reflexive strategy during later learning. We use a neurocognitive-genetic approach to identify intermediate phenotypes that modulate reflective brain function and examine their effects on speech learning. We focus on the COMT Val158Met polymorphism, which is linked to altered prefrontal function. The COMT-Val variant catabolizes dopamine more rapidly and is linked to poorer performance on prefrontally-mediated tasks. Adults (Met-Met: = 40; Met-Val= 75; Val-Val = 54) learned to categorize non-native Mandarin tones over five blocks of feedback-based training. Learning rates were the highest for the Met-Met genotype; the Val-Val genotype was associated with poorer overall learning. Poorer learning indicates increased perseveration of reflective unidimensional rule use, thereby preventing the transition to the reflexive system. We conclude that genetic variation is an important source of individual differences in complex phenotypes such as speech learning.

THURSDAY MORNING, 30 OCTOBER 2014 — INDIANA G, 9:00 A.M. TO 10:00 A.M.

Session 4aSPa

Signal Processing in Acoustics: Imaging and Classification

Grace A. Clark, Chair

Grace Clark Signal Sciences, 532 Alden Lane, Livermore, CA 94550

Contributed Papers

9:00

4aSPa1. Optimal smoothing splines improve efficiency of entropy imaging for detection of therapeutic benefit in muscular dystrophy. Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, 1632 Ridge Bend Dr., St Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Dept. of Mathematics, Washington Univ., St. Louis, MO), Jon Marsh (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO), and Samuel Wickline (Dept. of Mathematics, Washington Univ., Saint Louis, MO)

We have reported previously on sensitivity comparisons of signal energy and several entropies to changes in skeletal muscle architecture in experimental muscular dystrophy before and after pharmacological therapeutic intervention [M. S. Hughes, IEEE Trans. UFFC. 54, 2291–2299 (2007)]. The study was based on a moving window analysis of simple cubic splines that were fit to the backscattered ultrasound and required that the radio frequency ultrasound (RF) be highly oversampled. The current study employs optimal smoothing splines instead to determine the effect of analyzing the same data with increasing levels of decimation. The RF data were obtained from selected skeletal muscles of muscular dystrophy mice (mdx: dystrophin -/-) that were randomly blocked into two groups: 4 receiving steroid treatment over 2 weeks, and 4 untreated positive controls. Ultrasonic imaging was performed on day 15. All mice were anesthetized then each forelimb was imaged in transverse cross sections using a Vevo-660 with a single-element 40 MHz wobbler-transducer (model RMV-704, Visualsonics). The result of each scan was a three dimensional data set $384 \times 8192 \times$ # frames in size. We find the equivalent sensitivity of this new approach for detecting treatment benefits as before ($p<0.03$), but now at a decimated sampling rate slightly below the Nyquist frequency. This implies that optimal smoothing splines are useful for analysis of data acquired from point of care imaging devices where hardware cost and power consumption must be minimized.

9:15

4aSPa2. Waveform processing using entropy instead of energy: A quantitative comparison based on the heat equation. Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, 1632 Ridge Bend Dr., St Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Mathematics, Washington Univ., St Louis, MO), Jon Marsh (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO), and Samuel Wickline (Mathematics, Washington Univ., Saint Louis, MO)

Virtually all modern imaging devices function by collecting electromagnetic or acoustic backscattered waves and using the energy carried by these waves to determine pixel values that build up what is basically an "energy" picture. However, waves also carry "information" that also may be used to compute the pixel values in an image. We have employed several measures of information, most sensitive being the "joint entropy" of the backscattered wave and a reference signal. Numerous published studies have demonstrated the advantages of "information imaging," over conventional methods for materials characterization and medical imaging. A typical study is comprised of repeated acquisition of backscattered waves from a specimen that is changing slowing with acquisition time or location. The sensitivity of repeated experimental observations of such a slowly changing quantity may be defined as the mean variation (i.e., observed change) divided by mean variance (i.e., observed noise). Assuming the noise is Gaussian and using Wiener integration to compute the required mean values and variances, solutions to the Heat equation may be used to express the sensitivity for joint entropy and signal energy measurements. There always exists a reference such that joint entropy has larger variation and smaller variance than the corresponding quantities for signal energy, matching observations of several studies. A general prescription for finding an "optimal" reference for the joint entropy emerges, which has been validated in several studies.

9:30

4aSPa3. The classification of underwater acoustic target signals based on wave structure and support vector machine. Qingxin Meng, Shie Yang, and Shengchun Piao (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., No.145,Nantong St.,Nangang District, Harbin City, Heilongjiang Province 150001, China, mengqingxin005@hrbeu.edu.cn)

The sound of propeller is a remarkable feature of ship-radiated noise, the loudness and timbre of which are usually applied to identify types of ships. Since the information of loudness and timbre is indicated in the wave structure of time series, the feature of wave structure can be extracted to classify types of various underwater acoustic targets. In this paper, the method of feature vector extraction of underwater acoustic signals based on wave structure is studied. The nine-dimension features are constructed via signal statistical characteristics of zero-crossing wavelength, peek-to-peek amplitude, zero-crossing wavelength difference, and wave train areas. And then, the support vector machine (SVM) is applied as a classifier for two kinds of underwater acoustic target signals. The kernel function is set radial basis function (RBF). By properly setting the penalty factor and parameter of RBF, the recognition rate reaches over 89.5%, respectively. The sea-test data shows the validity of target recognition ability of the method above.

9:45

4aSPa4. Determination of Room Impulse Response for synthetic data acquisition and ASR testing. Philippe Moquin (Microsoft, One Microsoft Way, Redmond, WA 98052, pmoquin@microsoft.com), Kevin Venalainen (Univ. of Br. Columbia, Vancouver, BC, Canada), and Dinei A. Florêncio (Microsoft, Redmond, WA)

Automatic Speech Recognition (ASR) works best when the speech signal best matches the ones used for training. Training, however, may require thousands of hours of speech, and it is impractical to directly acquire them in a realistic scenario. Some improvement can be obtained by incorporating typical building acoustics measurement parameters such as RT, Cx, LF, etc., with limited gain. Instead, we estimate Room Impulse Responses (RIRs), and convolve speech and noise signals with the estimated RIRs. This produces realistic signals, which can then be processed by the audio pipeline, and used for ASR training. In our research, we use rooms with variable acoustics and repeatable source-receiver positions. The receivers are microphone arrays making the relative phase and magnitude critical. A standard mouth simulator for voice signals at various positions in the room is under robot control. A limited corpus of speech data as well as noise sources is recorded and the RIR at these 27 positions is determined using a variety of methods (chirp, MLS, impulse, and noise). The convolved RIR with the "clean speech" is compared to the actual measurements. Test methods used, differences from the measurements, and the difficulty of determining the unique RIR will be presented.

THURSDAY MORNING, 30 OCTOBER 2014 INDIANA G, 10:15 A.M. TO 12:00 NOON

Session 4aSPb

Signal Processing in Acoustics: Beamforming, Spectral Estimation, and Sonar Design

Brian E. Anderson, Cochair
Geophysics Group, Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545

R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, PA 16804

4a THU. AM

Contributed Papers

10:15

4aSPb1. Quantifying the depth profile of time reversal focusing in elastic media. Brian E. Anderson, Marcel C. Remillieux, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, bea@lanl.gov)

A focus of elastic energy on the surface of a solid sample can be useful to nondestructively evaluate whether the surface or the near-surficial region is damaged. Time reversal techniques allow one to focus energy in this manner. In order to quantify the degree to which a time reversal focus can probe near-surficial features, the depth profile of a time reversal focus must be quantified. This presentation will discuss numerical modeling and experimental results used to quantify the depth profile. [This work was supported by the U.S. Dept. of Energy, Fuel Cycle R&D, Used Fuel Disposition (Storage) Campaign.]

10:30

4aSPb2. Competitive algorithm blending for enhanced source separation of convolutive speech mixtures. Keith Gilbert (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 36 Walnut St., Berlin, MA 01503, kgilbert@umassd.edu), Karen Payton (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, N. Dartmouth, MA), Richard Goldhor, and Joel MacAuslan (Speech Technol. & Appl. Res., Corp., Bedford, MA)

This work investigates an adaptive filter network in which multiple blind source separation methods are run in parallel, and the individual outputs are combined to produce estimates of acoustic sources. Each individual algorithm makes assumptions about the environment (dimensions of enclosure, reflections, reverberation, etc.) and the sources (speech, interfering noise, position, etc.), which constitutes an individual hypothesis about the observed microphone outputs. The goal of this competitive algorithm blending (CAB) approach is to achieve the performance of the "true" method, i.e., the method that has full knowledge of the environment's and the sources' characteristics *a priori*, without any prior information. Results are given for time-invariant, critically- and over- determined, convolutive mixtures of speech and interfering noise sources, and the performance of the CAB method is compared with the "true" method in both the transient adaptation phase and in steady state.

10:45

4aSPb3. Structural infrasound signals in an urban environment. Sarah McComas, Henry Diaz-Alvarez, Mike Pace, and Mihan McKenna (US Army Engineer Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, sarah.mccomas@usace.army.mil)

Historically, infrasound arrays have been deployed in rural environments where anthropological noise sources are limited. As interest in monitoring sources at local distances grows in the infrasound community, it will be vital to understand how to monitor infrasound sources in an urban environment. Arrays deployed in urban centers have to overcome the decreased signal to noise ratio and reduced amount of real estate available to deploy an array. To advance the understanding of monitoring infrasound sources in urban environments, we deployed local and regional infrasound arrays on building rooftops of the campus of Southern Methodist University (SMU) and collected data for one seasonal cycle. The data was evaluated for structural source signals (continuous-wave packets) and when a signal was identified the back azimuth to the source was determined through frequency wavenumber analysis. This information was used to identify hypothesized structural sources; these sources were verified through direct measurement, structural numerical modeling and/or full waveform propagation modeling. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.

11:00

4aSPb4. Design of a speaker array system based on adaptive time reversal method. Gee-Pinn J. Too, Yi-Tong Chen, and Shen-Jer Lin (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd., Tainan 701, Taiwan, z8008070@email.ncku.edu.tw)

A system for focusing sound around desired locations by using a speaker array of controlled sources is proposed. To increase acoustic signal in certain locations where the user is within but to reduce it in the other certain locations by controlling source signals is the main objective of this study. Based on adaptive time reversal theory, input weighting coefficients for speakers are evaluated for the speaker sources. Experiments and simulations with a speaker array of controlled sources are established in order to observe the distribution of sound field under different boundary and control conditions. The results indicate that based on the current algorithm, the difference of sound pressure level between bright point and dark point can be as high as 12 dB with an eight speakers array system.

11:15

4aSPb5. Focusing the acoustic signal of a maneuvering rotorcraft. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

An algorithm was developed and tested to blindly focus the acoustic spectra of a rotorcraft that was blurred by time-varying Doppler shifts and other effects such atmospheric distortion. First, the fundamental frequency generated by the main rotor blades of a rotorcraft was tracked using a fixlag smoother. Then, the frequency estimates were used to resample the data in time using interpolation. Next, the motion compensated data were further focused using a technique based upon the phase gradient autofocus algorithm. The performance of the focusing algorithm was evaluated by analyzing the increase in the amplitude of the harmonics. For most of the data, the algorithm focused the harmonics between approximately 10–90 Hz to within 1–2 dB of an estimated upper bound obtained from conservation of energy and estimates of the Doppler shift. In addition, the algorithm was able to separate two closely spaced frequencies in the spectra of the rotorcraft. The algorithm developed can be used to preprocess data for classification, nulling, and tracking algorithms.

11:30

4aSPb6. Representing the structure of underwater acoustic communication data using probabilistic graphical models. Atulya Yellepeddi (Elec. Engineering/Appl. Ocean Phys. and Eng., Massachusetts Inst. of Technology/Woods Hole Oceanographic Inst., 77 Massachusetts Ave., Bldg. 36-683, Cambridge, MA 02139, atulya@mit.edu) and James C. Preisig (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Exploiting the structure in the output of the underwater acoustic communication channel in order to improve the performance of the communication system is a problem that has received much recent interest. Methods such as physical constraints and sparsity have been used to represent such structure in the past. In this work, we consider representing the structure of the received signal using probabilistic graphical models (more specifically Markov random fields), which capture the conditional dependencies amongst a collection of random variables. In the frequency domain, the inverse covariance matrix of the received signal is shown to have a sparse structure. Under the assumption that the signal may be modeled as a multivariate Gaussian random variable, this corresponds to a Markov random field. It is argued that the underlying cause of the structure is the cyclostationary nature of the signal. In practice, the received signal is not exactly cyclostationary, but data from the SPACE08 acoustic communication experiment is used to demonstrate that field data exhibits exploitable structure. Finally, techniques to exploit graphical model structure to improve the performance of wireless underwater acoustic communication are briefly considered.

11:45

4aSPb7. Choice of acoustics signals family in multi-users environment. Benjamin Ollivier, Frédéric Maussang, and René Garello (ITI, Institut Mines-Telecom / Telecom Bretagne - Lab-STICC, 655 Ave. du Technopole, Plouzané 29200, France, benjamin.ollivier@telecom-bretagne.eu)

Our application concerns a system immersed in an underwater acoustical context, with Nt transmitters and Nr slowly moving receivers. The objective is that all receivers detect the transmitted signals, in order to estimate the time of arrival (TOA) and then to facilitate the localization when several TOA (more than 3) are present. We have to choose a method to generate a number Ns of broad-band signals to use the Code Division Multiple Access (CDMA) modulation, specially adapted to our problem. This work is devoted to selecting Nt signals among the Ns available. The aim is to choose the most distinctly detectable ones. First, in a no Doppler context, the criterion of signals selection is based on a ratio between maximum of auto-correlation and cross-correlation. Second, in the presence of Doppler, we rely on Ambiguity Function which allows representing the correlation function to several frequency Doppler shifts. The choice of Nt signals is then based on ratio between maximum of auto-ambiguity and cross-ambiguity. In this paper we will highlight the relevance of the criteria (correlation, ambiguity function) in the choice of the most appropriate signals in function of the multi-users context.

Session 4aUW

Underwater Acoustics: Shallow Water Reverberation II

Brian T. Hefner, Chair
Applied Physics Laboratory, University of Washington, Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:00

Contributed Paper

8:05

4aUW1. SONAR Equation perspective on TREX13 measurements. Dajun Tang and Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Modeling shallow water reverberation is a problem that can be approximated as two-way propagation (including multiple forward scatter) and a single backward scatter. This can be effectively expressed in terms of the SONAR equation: RL = SL-2 x TL + SS, where RL is reverberation level, SL is the source level, TL is the one way transmission loss, and SS is the integrated scattering strength. In order to understand the reverberation problem at the basic research level, both propagation and scattering physics need to be properly addressed. A major goal of TREX13 (Target and Reverberation EXperiment 2013) is to quantitatively investigate reverberation with sufficient environmental measurement to support full modeling of reverberation data. Along a particular reverberation track at the TREX13 site, TL and direct-path backscatter were separately measured. Environmental data were extensively collected along this track. This talk will bring together all the components of the SONAR equation measured separately at the TREX13 site to provide an assessment of the reverberation process along with environmental factors impacting each of the components.

Invited Papers

8:20

4aUW2. Environmental measurements collected during TREX13 to support acoustic modeling. Brian T. Hefner and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Appl. Phys. Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

The major goal of TREX13 (Target and Reverberation EXperiment 2013) was to quantitatively investigate reverberation with sufficient environmental measurements to support full modeling of reverberation data. The collection of environmental data to support reverberation modeling is usually limited by the large ranges (10s of km) involved, the temporal and spatial variability of the environment and the time variation of towed source/receiver locations within this environment. In order to overcome these difficulties, TREX13 was carried out in a 20 m deep shelf environment using horizontal line arrays mounted on the seafloor. The water depth and well controlled array geometry allowed environmental characterization to be focused on the main beam of the array, i.e., along a track roughly 5 km long and 500 m wide. This talk presents an overview of the efforts made to characterize the sea surface, water column, seafloor, and sub-bottom along this track to support the modeling of acoustic data collected over the course of the experiment. [Work supported by ONR Ocean Acoustics.]

8:40

4aUW3. Persistence of sharp acoustic backscatter transitions observed in repeat 400 kHz multibeam echosounder surveys offshore Panama City, Florida, over 1 and 24 months. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org) and Barbara J. Kraft (10dBx LLC, Barrington, New Hampshire)

The Target and Reverberation Experiment 2013 (TREX13), conducted offshore Panama City, FL, from April to June 2013, sought to determine which environmental parameters contribute the most to acoustic reverberation and control sonar performance prediction modeling for acoustic frequencies between 1 kHz and 10 kHz. In that context, a multibeam echosounder operated at 400 kHz was used to map the seafloor relief and its high-frequency acoustic backscatter characteristics along the acoustic propagation path of the reverberation experiment. Repeat surveys were conducted a month apart, before and after the main reverberation experiment. In addition, repeat surveys were conducted at 200 kHz in April 2014. Similar mapping work was also conducted in April 2011 during a pilot experiment (GulfEx11) near the site chosen for TREX13. Both experiments revealed a persistent occurrence of sharp transitions from high to low acoustic backscatter at the bottom of swales. Hypotheses are presented for observable differences in bathymetry and acoustic backscatter in the overlap region between the GulfEx11 survey and the TREX13 surveys conducted 2 y apart. [Work supported by ONR 322 OA.]

Contributed Papers

9:00

4aUW4. Roughness measurement by laser profiler and acoustic scattering strength of a sandy bottom. Nicholas P. Chotiros, Marcia J. Isakson, Oscar E. Siliceo, and Paul M. Abkowitz (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

The roughness of a sandy seabed off Panama City, FL, was measured with a laser profiler. This was the site of the target and reverberation experiment of 2013 (TREX13) in which propagation loss and reverberation strength were measured. The area may be characterized as having small scale roughness due to bioturbation overlaying larger sand ripples due to current activity. The area was largely composed of sand with shell hash crossed by ribbons of softer sediment at regular intervals. The roughness measurements were concentrated in the areas where the ribbons intersected the designated sound propagation track. Laser lines projected on the sand were imaged by a high-definition video recorder. The video images were processed to yield bottom profiles in three dimensions. Finally, the roughness data are used to estimate acoustic bottom scattering strength. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

9:15

4aUW5. Seafloor sub-bottom Imaging along the TREX reverberation track. Joseph L. Lopes, Rodolf Arrieta, Iris Paustian, Nick Pineda (NSWC PCD, 110 Vernon Ave, Panama City, FL 32407-7001, joseph.l.lopes@navy.mil), and Kevin Williams (Appl. Phys. Lab. / Univ. of Washington, Seattle, WA)

The Buried Object Scanning Sonar (BOSS) integrated into a Bluefin12 autonomous underwater vehicle was used to collect seafloor sub-bottom data along the TREX reverberation track. BOSS is a downward looking sonar and employs an omni-directional source to transmit a 3 to 20 kHz linear frequency modulated (LFM) pulse. Backscattered signals are received by two 20-channel linear hydrophone arrays. The BOSS survey was carried out to support long-range reverberation measurements at 3 kHz. The data were beamformed in three dimensions and processed into 10cm x 10cm x 10cm voxel maps of backscattering to a depth of 1 m. Comparison of the BOSS imagery with 400 kHz multibeam sonar imagery of the seafloor allows muddy regions to be identified and shows differences rationalized by the differences in sediment penetration of the two frequency ranges utilized. Processed BOSS data are consistent with observations from diver cores and the reverberation data collected by stationary arrays deployed on the seafloor. Specifically, stronger and deeper backscattering from muddy regions is observed (relative to near-by sandy regions). This correlates well with the large amounts of detritus (e.g., shell fragments) and complicated vertical layering within cores, and the enhanced reverberation, from those regions. [Work supported by ONR.]

9:30

4aUW6. Seabed characterisation using a low cost digital thin line array: Results from the Target and Reverberation Experiments 2013. Unnikrishnan K. Chandrika, Venugopalan Pallayil (Acoust. Res. Lab, TMSI, National Univ. of Singapore, Acoust. Res. Lab, 18 Kent Ridge Rd., Singapore 119227, Singapore, venu@arl.nus.edu.sg), Nicholas Chotiros (Appl. Res. Lab, Univ. of Texas, Austin, TX), and Marcia Isakson (Appl. Res. Lab, Univ. of Texas, Ausitn, TX)

During TREX-13 experiments in the Gulf of Mexico in May 2013, the use of a low cost digital thin line array (DTLA) developed at the Acoustic Research Lab, National University of Singapore was explored towards sea-bottom characterisation. The array, developed for use from AUV platforms, was hosted on a Sea-eye ROV from UT Austin and towed using R/V Smith, as no AUV platform was available during the course of the above experiment. The ROV was also hosting a wide-band acoustic source sending out chirp waveforms in the frequency range of 3 to 15 kHz. It has been observed that despite the complexity of set-up used, the array dynamics could be maintained well during the tow test and also the data collected were useful in estimating the bottom type from reflection coefficient measurements and comparing with the models available. Our analysis by matched filtering the received data and estimating the bottom reflection coefficient showed that the bottom type at the experimental site was sandy-silt, which fairly compared with observations on the same by other means. Details of experiments performed and the results from the data analyzed would be presented during the meeting. Some suggestions for improvement for future experiments will be discussed.

9:45

4aUW7. Wide-angle reflection measurements (TREX13): Evidence of strong seabed lateral heterogeneity at two scales. Charles W. Holland, Chad Smith (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Paul Hines (Elec. and Comput. Eng., Dalhousie Univ., Dalhousie, NS, Canada), Jan Dettmer, Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Samuel Pinson (Appl. Res. Lab., The Penn State Univ., State College, PA)

Broadband wide-angle reflection data possess high information content, yielding both depth and frequency dependence of sediment wave velocities, attenuations, and density. Measurements at two locations off Panama City, FL (TREX13), however, presented a surprise: over the measurement aperture (a few tens of meters) the sediment was strongly laterally variable. This prevented the usual analysis in terms of depth dependent geoacoustic properties. Only rough estimates could be made. On the other hand, the data provide clear evidence of lateral heterogeneity at O(10^0-10^1) m scale. The two sites were separated by ~6 km, one on a ridge (lateral dimension 10^2 m) and one in a swale of comparable dimension; the respective sound speeds are roughly 1680 m/s and 1585 m/s. The lateral variability, especially at the 1–10 m scale is expected to impact both propagation and reverberation. Characteristics of the reflection data and its attendant "surprise" suggest the possibility of objectively separating the intermingled angle and range dependence; this would open the door to detailed geoacoustic estimation in areas of strong lateral variability. [Research supported by ONR Ocean Acoustics.]

Invited Papers

10:00

4aUW8. Modeling reverberation in a complex environment with the finite element method. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Acoustic finite element models solve the Helmholtz equation exactly and are customizable to the scale of the discretization of the environment. This makes them an ideal candidate for reverberation studies in complex environments. In this study, reverberation is calculated for a realistic shallow water waveguide. The environmental parameters are taken from the extensive characterization completed for the Target and Reverberation Experiment (TREX) conducted off the coast of the Florida panhandle in 2013. Measured sound speed profiles, sea surface roughness, bathymetry, and measured ocean bottom roughness are included in the model. Measurements of the normal incidence bottom loss are used as a proxy for range dependent sediment density. Results are compared with a closed form solution for reverberation. [Work sponsored by ONR, Ocean Acoustics.]

10:20–10:35 Break

Contributed Papers

10:35

4aUW9. Normal incidence reflection measurements (TREX13): Inferences for lateral heterogeneity over a range of scales. Charles W. Holland, Chad Smith (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), and Paul Hines (Elec. and Comput. Eng., Dalhousie Univ., Dalhousie, NS, Canada)

Normal incidence seabed reflection data suffer from a variety of ambiguities that make quantitative interpretation difficult. The reflection coefficient has an inseparable ambiguity between bulk density and compressional sound speed. Even more serious, reflection data are a function of other sediment characteristics including interface roughness, volume heterogeneities, and local bathymetry. Seafloor interface curvature is especially important and can lead to focusing/defocusing of the reflected field. An attempt is made with ancillary data including bathymetry, 400 kHz backscatter, and wide angle seabed reflection data to separate some of the mechanisms. Resulting analysis of 1–12 kHz reflection data suggest: (1) strong lateral sediment heterogeneity exists on scales of 10–100 m; (2) there are distinct geoacoustic regimes on the lee and stoss side of the ridge crest, and also between crest and the swale, and (3) the ridge crest geoacoustic properties are similar across distances of 6 km along two perpendicular transects (1 correlation). [Research supported by ONR Ocean Acoustics.]

10:50

4aUW10. Acoustic measurements on mid-shelf sediments with cobble: Implications for reverberation. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Gavin Steininger, Jan Dettmer, Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Allen Lowrie (Picayune, MS)

The vast majority of sediment acoustics research has focused on rather homogeneous sandy sediments. Measurements for sediments containing cobbles (grain size greater than 6 cm) are rare. Here, measurements are presented for mid-shelf sediments containing pebbles/cobbles mixed with other grain sizes spanning 7 orders of magnitude, including silty clay, sand, and shell hash. The 2 kHz sediment sound speed in two distinct layers with cobble is 1531 ± 5 m/s and 1800 ± 20 m/s at the 95% credibility interval. The dispersion over the 400–2000 Hz band was relatively weak, 2 and 7 m/s respectively. The objective is to (1) present results for a sediment type for which little is known, (2) motivate development of theoretical wave propagation models for wide grain size distributions, and (3) speculate on the possibility of cobble as a scattering mechanism for mid shelf reverberation. The presence of cobbles from 1 to 3 m (possibly extending to 6 m) sub-bottom suggest they are the dominant scattering mechanism at this site. Though sediments with cobbles might be considered unusual, especially on the mid-shelf, they may be more common than the paucity of measurements would suggest since typical direct sampling techniques (e.g., cores and grab samples) have fundamental sampling limitations. [Research supported by ONR Ocean Acoustics.]

Session 4pAAa

Architectural Acoustics and Speech Communication: Acoustic Trick-or-Treat: Eerie Noises, Spooky Speech, and Creative Masking

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Eric J. Hunter, Cochair
Department of Communicative Sci., Michigan State University, 1026 Red Cedar Road, East Lansing, MI 48824

Chair's Introduction—1:10

Invited Papers

1:15

4pAAa1. Auditory illusions of supernatural spirits: Archaeological evidence and experimental results. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com) and Miriam A. Kolar (Amherst College, Amherst, MA 01002)

Sound reflection, reverberation, ricochets, and interference patterns were perceived in the past as eerie sounds attributable to invisible echo spirits, thunder gods, ghosts, and sound-absorbing bodies. These beliefs in the supernatural were recorded in ancient myths, and expressed in tangible archaeological evidence including canyon petroglyphs, cave paintings, and megalithic stone circles including Stonehenge. Disembodied voices echoing throughout canyons gave the impression of echo spirits calling out from the rocks. Thunderous reverberation filling deep caves gave the impression of the same thundering stampedes of invisible hoofed animals that were believed to accompany thunder gods in stormy skies. If you did not know about sound wave reflection, would the inexplicable noise of a ricochet in a large room have given you the impression of a ghost moaning "BOOoo" over your shoulder? Mysterious silent zones in an open field gave the impression of a ring of large phantom objects blocking pipers' music. Complex behaviors of sound waves such as reflection and interference (which scientists today dismiss as acoustical artifacts) can experimentally give rise to psychoacoustic misperceptions in which such unseen sonic phenomena are attributed to the supernatural. See https://sites.google.com/site/rockartacoustics/ for further details.

1:35

4pAAa2. Pututus, resonance and beats: Acoustic wave interference effects at Ancient Chavín de Huántar, Perú. Miriam A. Kolar (Program in Architectural Studies and Dept. of Music, Amherst College, Barrett Hall, 21 Barrett Hill Dr., AC# 2255, PO Box 5000, Amherst, MA 01002, mkolar@amherst.edu)

Acoustic wave interference produces audible effects observed and measured in archaeoacoustic research at the 3,000-year-old Andean Formative site at Chavín de Huántar, Perú. The ceremonial center's highly-coupled network of labyrinthine interior spaces is riddled with resonances excited by the lower-frequency range of site-excavated conch shell horns. These *pututus*, when played together in near-unison tones, produce a distinct "beat" effect heard as the result of the amplitude variation that characterizes this linear interaction. Despite the straightforward acoustic explanation for this architecturally enhanced instrumental sound effect, the performative act reveals an intriguing perceptual complication. While playing pututus inside Chavín's substantially intact stone-and-earthen-mortar buildings, pututu performers have reported an experience of having their instruments' tones "guided" or "pulled" into tune with the dominant spatial resonances of particular locations. In an ancient ritual context, the recognition and understanding of such a sensory component would relate to a particular worldview beyond the reach of present-day investigators. Despite our temporal distance, an examination of the intertwined acoustic phenomena operative to this architectural–instrumental–experiential puzzle enriches the interdisciplinary research perspective, and substantiates perceptual claims.

1:55

4pAAa3. Tapping into the theatre of the mind; creating the eerie scene through sound. Jonathon Whiting (Media and Information, Michigan State Univ., College of Commun. Arts and Sci., 404 Wilson Rd., Rm. 409, East Lansing, MI 48824, whitin26@msu.edu)

Jaws. Psycho. Halloween. Halo. Movies and video games depend on music and acoustics to evoke certain emotional states in the audience or game player. But what is the recipe for creating a haunting scene? A creaky door, a scream, a minor chord on a piano. How and why are certain emotions pulled out of a listener in response to sound? From sound environments to mental expectations, the media industry uses a variety of techniques to elicit responses from an audience. This presentation will discuss and present examples of the principles behind the sound of fright.

2:15

4pAAa4. Disquiet: Epistemological bogeymen and other exploits in audition. Ean White (unaffiliated, 1 Westinghouse Plaza C-216, Boston, MA 02136-2079, ean@eanwhite.org)

Beginning with an interest in "physiological musics," Ean White's sound art exploits interstices in our sensory apparatus with unnerving results. He will recount a series of audio experiments with effects ranging from involuntary muscle contractions to the creation of sounds eerily unique to each listener. The presentation will include discussion of his techniques and how they inform his artistic practice.

2:35

4pAAa5. Removing the mask in multitrack music mixing. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The sound recording heard via stereo loudspeakers and headphones is made up of many dozens—sometimes more than 100—discrete tracks of musical elements. Multiple individual performances across a variety of instruments are fused into the final, two-channel recording—left and right—that is released to consumers. Achieving sonic success in this many-into-two challenge requires strategic, creative release from masking. Part of the artistry of multitrack mixing includes finding innovative signal processing approaches that enable the full arrangement and the associated interaction among the multitrack components of the music to be heard and enjoyed. Masking among tracks clutters and obscures the music. But audio engineers are not afraid. They want you hear what's behind the mask. Hear how. Happy Halloween.

2:55

4pAAa6. Documenting and identifying things that go bump in the night. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

Acoustical consultants are occasionally asked to help diagnose mysterious noises in buildings, and it can be difficult to be present and ready to make measurements when such noises occur. This paper will presents some of the tools and methods the author uses for recording and analyzing these events. These include the use of tablet-based measurement devices and high-speed playback of long-term recordings.

3:15–3:30 Break

3:30

4pAAa7. Inaudible information, disappearing declamations, misattributed locations, and other spooky ways your brain fools you—every day. Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

We bumble through life convinced that our senses provide reliable, faithful information about the world. Yet on closer inspection, our brains constantly misinform us, creepily convincing us of "truths" that are just plain false. We hear information that is not really there. We are oblivious to sounds that are perfectly audible. For sounds that we do hear, we cannot tell when they actually occurred. We completely overlook changes that even a simple acoustic analysis would detect with 100% accuracy. In short, we misinterpret the sounds reaching our ears all the time, and do not even realize it. This talk will review the evidence for how unreliable and biased we are in interpreting the world—and why the chilling failures of our perceptual machinery may be excusable, or even useful, as we navigate the complex world in which we live.

3:50

4pAAa8. The mysterious case of the singing toilets and other nerve wracking tales of unwanted sound. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Lightweight construction nightmares, devilish designs that never see acoustic review, improper purposing of spaces, and other stories involving the relentless torture of building occupants. Will they survive?

4:10

4pAAa9. Sound effects with AUditory syntaX—A high-level scripting language for sound processing. Bomjun J. Kwon (Hearing, Speech and Lang., Gallaudet University, 800 Florida Ave NE, Washington, DC 20002, bomjun.kwon@gallaudet.edu)

AUditory syntaX (AUX) is a high-level scripting programming language specifically crafted for the generation and processing of auditory signals (Kwon, 2012; Behav Rev 44, 361–373). AUX does not require knowledge or prior experience in computer programming. Rather, AUX provides an intuitive and descriptive environment where users focus on perceptual components of the sound, without tedious tasks unrelated to the perception such as memory management or array handling often required in other computer languages such as C++ or MATLAB that are popularly used in auditory science. This presentation provides a demonstration of AUX for the generation and processing of various sound effects, particularly "fun" or "spooky" sounds. Processing methods for sound effects widely used in arts, films and other media, such as reverberation, echoes, modulation, pitch shift, and flanger/phaser, will be reviewed and coding in AUX to generate those effects and the generated sounds will be demonstrated.

4:30

4pAAa10. Eerie voices: Odd combinations, extremes, and irregularities. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

The human voice can project an eerie quality when certain characteristics are present in a particular context. Some types of eerie voices may be derived from physiological scaling of the speech production system that is either humanly impossible or nearly so. By combining previous work on adult speech, and current research on speech development, the purpose of this study was to simulate vocalizations and speech based on unusual configurations of the vocal tract and vocal folds, and by imposing irregularities on movement and vibration. The resulting sound contains qualities that are human-like, but not typical, and hence may give the perceptual impression of eeriness. [Supported in part by NIH R01-DC011275.]

4:50

4pAAa11. Segregation of ambiguous pulse-echo streams and suppression of clutter masking in FM bat sonar by anticorrelation signal processing. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

Big brown bats often fly in conditions where the density and spatial extent of clutter requires a high rate of pulse emissions. Echoes from one broadcast still are arriving when the next broadcast is sent out, creating ambiguity about matching echoes to corresponding broadcasts. Biosonar sounds are widely beamed and impinge on the entire surrounding scene. Numerous clutter echoes typically are received from different directions at similar times. The multitude of overlapping echoes and the occurrence of pulse-to-echo ambiguity compromises the bat's ability to peer into the upcoming path and determine whether it is free of collision hazards. Bats have to associate echoes with their corresponding broadcasts to prevent ambiguity, and off-side clutter echoes have to be segregated from on-axis echoes that inform the bat about its immediate forward path. In general, auditory streaming to resolve elements of an auditory scene depends on differences in pitch and temporal pattern. Bats use a combination of temporal and spectral pitch to assign echoes to "target" and "clutter" categories within the scene, which prevents clutter masking, and they associate incoming echoes with the corresponding broadcast by treating the mismatch of echoes with the wrong broadcast as a type of clutter. [Supported by ONR.]

5:10

4pAAa12. Are you hearing voices in the high frequencies of human speech and voice? Brian B. Monson (Pediatric Newborn Medicine, Brigham and Women's Hospital, Harvard Med. School, 75 Francis St., Boston, MA 02115, bmonson@research.bwh.harvard.edu)

The human voice produces acoustic energy at frequencies above 6 kHz. Energy in this high-frequency region has long been known to affect perception of speech and voice quality, but also provides non-qualitative information about a speech signal. This presentation will demonstrate how much useful information can be gleaned from the high frequencies with a report on studies where listeners were presented with only high-frequency energy extracted from speech and singing. Come to test your own abilities and decide if you can hear strange voices or just chirps and whistles in the high frequencies of human speech and voice.

Contributed Paper

5:30

4pAAa13. Measuring the impact of room acoustics on emotional responses to music using functional neuroimaging: A pilot study. Martin S. Lawless and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu)

Past cognitive neuroscience studies have established links between music and an individual's emotional response. Specifically, music can induce activations in brain regions most commonly associated with reward and pleasure (Blood/Zatorre PNAS 2001). To further develop concert hall design criteria, functional magnetic resonance imaging (fMRI) techniques can be used to investigate the emotional preferences of room acoustics stimuli. Auralizations were created under various settings ranging from anechoic to extremely reverberant. These stimuli were presented to five participants in an MRI machine, and the subjects were prompted to rate the stimuli in terms of preference. Noise stimuli that matched the acoustic stimuli temporally and spectrally were also presented to the participants for the analysis of main contrasts of interest. In addition, the participants were first tested in a mock scanner to acclimatize the subjects to the environment and later validate the results of the study. Voxel-wise region of interest analysis was used to locate the emotion and reward epicenters of the brain that were activated when the subjects enjoyed a hall's acoustics. The activation levels of these regions, which are associated with positive-valence emotions, were examined to determine if the activations correlate with preference ratings.

Session 4pAAb

Architectural Acoustics, Speech Communication, and Noise: Room Acoustics Effects on Speech Comprehension and Recall II

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

David H. Griesinger, Cochair
Research, David Griesinger Acoustics, 221 Mt Auburn St #504, Cambridge, MA 02138

Invited Papers

1:15

4pAAb1. Challenges for second-language learners in difficult acoustic environments. Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

Most anyone who has lived in a foreign country for any length of time knows that even everyday tasks can become tiring and frustrating when one must accomplish them while navigating a seemingly endless maze of unfamiliar social customs, vocabulary and speech that seem far removed from one's language laboratory experience. Add to these challenges noise, reverberation, and/or cognitive demand (e.g., learning caculus, responding to multiple customer, and co-worker demands) and even experienced learners may begin to question their proficiency. This presentation will provide an overview of the speech perception and production challenges faced by second-language learners in difficult acoustic environments that we may encounter every day, such as in large lecture halls, retail or customer service, to name a few. Past and current research investigating the effects of various environmental challenges on both relatively early and later learners of a second language will be considered, as well as strategies that may mitigate challenges for both speakers and listeners in some of these conditions.

1:35

4pAAb2. Development of speech perception under adverse listening conditions. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

Speech communication success is dependent on interactions among the talker, listener, and listening environment. One such important interaction is between the listener's age and the noise and reverberation in the environment. Previous work has demonstrated that children have greater difficulty than adults in noisy and highly reverberant environments, such as those frequently found in classrooms. I will review research that considers how a talker's production patterns also contribute to speech comprehension, focusing on nonnative talkers. Studies from my lab have demonstrated that children have more difficulty than adults perceiving speech that deviates from native language norms, even in quiet listening conditions in which adults are highly accurate. When a nonnative talker's voice was combined with noise, children's word recognition was particularly poor. Therefore, similar to the developmental trajectory for speech perception in noise or reverberation, the ability to accurately perceive speech produced by nonnative talkers continues to develop well into childhood. Metrics to quantify speech intelligibility in specific rooms must consider both listener characteristics, talker characteristics, and their interaction. Future research should investigate how children's speech comprehension is influenced by the interaction between specific types of background noise and reverberation and talker production characteristics. [Work supported by NIH-R21DC010027.]

1:55

4pAAb3. Measurement and prediction of speech intelligibility in noise and reverberation for different sentence materials, speakers, and languages. Anna Warzybok, Sabine Hochmuth (Cluster of Excellence Hearing4All, Medical Phys. Group, Universität Oldenburg, Oldenburg D-26111, Germany, a.warzybok@uni-oldenburg.de), Jan Rennies (Cluster of Excellence Hearing4All, Project Group Hearing, Speech and Audio Technol., Fraunhofer Inst. for Digital Media Technol. IDMT, Oldenburg, Germany), Thomas Brand, and Birger Kollmeier (Cluster of Excellence Hearing4All, Medical Phys. Group, Universität Oldenburg, Oldenburg, Germany)

The present study investigates the role of the speech material type, speaker, and language for speech intelligibility in noise and reverberation. The experimental data are compared to predictions of the speech transmission index. First, the effect of noise only, reverberation only, and the combination of noise and reverberation was systematically investigated for two types of sentence tests. The hypothesis to be tested was that speech intelligibility is more affected by reverberation when using an open-set speech material consisting of everyday sentences than when using a closed-set test with syntactically fixed and semantically unpredictable sentences. In order to distinguish between the effect of speaker and language on speech intelligibility in noise and reverberation, the closed-set speech material was recorded using bilingual speakers of German-Spanish and German-Russian. The experimental data confirmed that the effect of

4p THU. PM

reverberation was stronger for an open-set test than for a closed-set test. However, this cannot be predicted by the speech transmission index. Furthermore, the inter-language differences in speech reception thresholds were on average up to 5 dB, whereas inter-talker differences were of about 3 dB. The Spanish language suffered more under reverberation than German and Russian, what again challenged the predictions of the speech transmission index.

2:15

4pAAb4. Speech comprehension in realistic classrooms: Effects of room acoustics and foreign accent. Zhao Peng, Brenna N. Boyd, Kristin E. Hanna, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182, zpeng@huskers.unl.edu)

The current classroom acoustics standard (ANSI S12.60) recommends that core learning spaces shall not exceed reverberation time (RT) of 0.6 second and background noise level (BNL) of 35 dBA, based on speech intelligibility performance mainly by the native English-speaking population. This paper presents two studies on the effects of RT and BNL on more realistic classroom learning experiences. How do native and non-native English-speaking listeners perform on speech comprehension tasks under adverse acoustic conditions, if the English speech is produced by talkers whose native language is English (Study 1) versus Mandarin Chinese (Study 2)? Speech comprehension materials were played back in a listening chamber to individual listeners: native and non-native English-speaking in Study 1; native English, native Mandarin Chinese, and other non-native English-speaking in Study 2. Each listener was screened for baseline English proficiency for use as a covariate in the statistical analysis. Participants completed dual tasks simultaneously (speech comprehension and adaptive dot-tracing) under 15 different acoustic conditions, comprised of three BNL conditions (RC-30, 40, and 50) and five RT scenarios (0.4–1.2 s). Results do show distinct differences between the listening groups. [Work supported by a UNL Durham School Seed Grant and the Paul S. Veneklasen Research Foundation.]

Contributed Papers

2:35

4pAAb5. Speech clarity in lively theatres. Gregory A. Miller and Carl Giegold (Threshold Acoust., LLC, 53 W. Jackson Boulevard, Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com)

By their very nature, theatres must be "lively" acoustic spaces. The audience must hear one another, so laughter and applause can ripple around the room, and they must have the aural sensation of being in a large space heightens the excitement of being at a live performance. Similarly, the theatre must reflect sound back to the actors in a way that helps them to gauge how well their voices are filling the room, and to gauge audience response throughout the performance. And yet this liveliness runs counter to much of conventional wisdom regarding the acoustic conditions to support speech clarity. This paper will describe ways in which the acoustic response of a room can be built up to support both speech clarity and liveliness, with a particular emphasis on theatre spaces in which the actors are placed in the same volume as the audience (thrust and surround stages).

2:50

4pAAb6. Speech communication in noise to valid the virtual sound capturing system. Hyung Suk Jang, Seongmin Oh, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, janghyungs@gmail.com)

The microphone systems were designed to capture the real sound field for the creation of the remote virtual coexistence space: omnidirectional microphone, binaural dummy head, linear array microphones, and spherical microphone. The captured signals were applied to synthesize into the binaural signal. These binaural cues were generated using head-related transfer function (HRTF) through headphone. For the validation, the sentence recognition tests were carried out to quantify the ability of speech perception with the sentence lists for normal listeners. In addition, the readability and the naturalness were used to assess the quality of the synthesized sounds. The different noise environments were applied with different signal to noise ratio and an efficient sound capturing system was suggested by the comparing the results of the sentence recognition tests.

3:05

4pAAb7. Quantifying a measure and exploring the effect of varying reflection densities from realistic room impulse responses. Hyun Hong and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, hhong@huskers.unl.edu)

Perceptual studies using objective acoustic metrics calculated from room impulse responses, such as reverberation time and clarity index, are common. Less work has been conducted looking explicitly at the reflection density, or the number of reflections per second. The reflection density, though, may well have its own perceptual influence when reverberation time and source-receiver distances are controlled, particularly in relation to room size perception. This paper presents first an investigation into quantifying the reflection density from realistic room impulse responses that may be measured or simulated. The resolution of the sampling frequency, time window applied, and cut-off level for including a reflection in the count are considered. The quantification method is subsequently applied to select a range of realistic RIRs for use in a perceptual study on determining the maximum audible reflection density by humans, using both speech and clapping signals. Results from this study are compared to those from similar previous work by the authors which used artificially simulated impulse responses with constant reflection densities over time.

Session 4pAB

Animal Bioacoustics and Acoustical Oceanography: Use of Passive Acoustics for Estimation of Animal Population Density II

Tina M. Yack, Cochair
Bio-Waves, Inc., 364 2nd Street, Suite #3, Encinitas, CA 92024

Danielle Harris, Cochair
Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, The Observatory, Buchanan Gardens, St. Andrews KY16 9LZ, United Kingdom

Chair's Introduction—1:15

Invited Papers

1:20

4pAB1. Estimating singing fin whale population density using frequency band energy. David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Elizabeth T. Küsel (NW Electromagnetics and Acoust. Res. Lab., Portland State Univ., Portland, OR), Danielle Harris, Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), and Luis Matias (Instituto Dom Luiz, Faculdade de Ciências, Universidade de Lisboa, Lisbon, Portugal)

Fin whale (*Balaenoptera physalus*) song occurs in a narrow frequency band between approximately 15 and 25 Hz. During the breeding season, the sound from many distant fin whales in tropical and subtropical parts of the world may be seen as a "hump" in this band of the ocean acoustic spectrum. Since a higher density of singing whales leads to more energy in the band, the size of this hump—the total received acoustic energy in this frequency band—may be used to estimate the population density of singing fin whales in the vicinity of a sensor. To estimate density, a fixed density of singing whales is simulated; using acoustic propagation modeling, the energy they emit is propagated to the sensor, and the received level calculated. Since received energy in the fin whale band increases proportionally with the density of whales, the density of whales may then be estimated from the measured received energy. This method is applied to a case study of sound recorded on ocean-bottom recorders southwest of Portugal; issues covered include variance due to acoustic propagation modeling, reception area, variation in whale song acoustic level and frequency, and elimination of interfering sounds. [Funding from ONR.]

1:40

4pAB2. Large-scale passive-acoustics-based population estimation of African forest elephants. Yu Shiu, Sara Keen, Peter H. Wrege, and Elizabeth Rowland (BioAcoust. Res. Program, Cornell Univ., 159 Sapsucker Woods Rd, Ithaca, NY 14850, atoultaro@gmail.com)

African forest elephants (*Loxodonta cyclotis*) live in tropical rainforests in Central Africa and often use low-frequency vocalizations for long-distance communication and coordination of group activities. There is great interest in monitoring population size in this species; however, the dense rainforest canopy severely limits visibility, making it difficult to estimate abundance using traditional methods such as aerial surveys. Passive acoustic monitoring offers an alternative approach of estimating its abundance in a low visibility environment. The work we present here can be divided into three steps. First, we apply an automatic elephant call detector, which enables the processing of large-scale acoustic signals in a reasonable amount of time. Second, we apply a density estimation method we designed for a single microphone. Because microphones are often positioned far apart in order to cover a large area in the rainforest, meaning that the same call will not produce multiple arrivals on different recording units. Lastly, we examine results from our historic data across five years in six locations in central Africa, which includes over 1000 days of sound stream. We will address the feasibility of long-term population monitoring and also the potential impact of human activity on elephant calling behavior.

2:00

4pAB3. A generalized random encounter model for estimating animal density with remote sensor data. Elizabeth Moorcroft, Tim C. D. Lucas (Ctr. for Mathematics, Phys. and Eng. in the Life Sci. and Experimental Biology, UCL, CoMPLEX, University College London, Gower St., London WC1E 6BT, United Kingdom, e.moorcroft@ucl.ac.uk), Robin Freeman, Marcus J. Rowcliffe (Inst. of Zoology, Zoological Society of London, London, United Kingdom), and Kate E. Jones (Ctr. for Biodiversity and Environment Res., UCL, London, United Kingdom)

Acoustic detectors are commonly being used to monitor wildlife. Current estimators of abundance or density require recognition of individuals or the distance of the animal from the sensor, which is often difficult. The random encounter model (REM) has been successfully applied to count data without these requirements. However, count data from acoustic detectors do not fit the assumptions of the REM due to the directionality of animal signals. We developed a generalized REM (gREM), to estimate animal density from count data, derived for different combinations of sensor detection widths and animal signal widths. We tested the accuracy and precision of this model using simulations for different combinations of sensor detection and animal signal widths, number of captures, and animal movement models. The gREM produces accurate estimates of absolute animal density. However, larger sensor detection and animal signal widths, and larger number of captures give more precise estimates. Different animal movement models had no effect on the gREM. We conclude that the gREM provides an effective method to estimate animal densities in both marine and terrestrial environments. As acoustic detectors become more ubiquitous, the gREM will be increasingly useful for monitoring animal populations across broad spatial, temporal, and taxonomic scales.

2:20

4pAB4. Using sound propagation modeling to estimate the number of calling fish in an aggregation from single-hydrophone sound recordings. Mark W. Sprague (Phys., East Carolina Univ., M.S. 563, Greenville, NC 27858, spraguem@ecu.edu) and Joseph J. Luczkovich (Biology, East Carolina Univ., Greenville, NC)

Many fishes make sounds during spawning events that can be used to estimate abundance. Spawning stock size is a measure of fish population size that is used by fishery biologists to manage harvests levels. It is desirable that such an estimate be assessed easily and remotely using passive acoustics. Passive acoustics techniques (hydrophones) can be used to identify sound-producing species, but it is difficult to count individual sound sources in the sea, where it is dark, background noise levels can be high, but species can be identified by their sounds. We have developed a method that can estimate the density of calling fish in an aggregation from single-hydrophone recordings. Our method requires a sound propagation model for the area in which the aggregation is located. We generate a library of modeled sounds of virtual Monte-Carlo generated distributions of fish to determine the range of fish population densities that match the characteristics of single-hydrophone sound recording. Such a model could be used from a fixed station (e.g., an observatory) to estimate the population size of the sound producers. In this presentation, we will present some calculations made using this method and will examine the benefits and limitations of the technique.

Contributed Papers

2:40

4pAB5. An experimental evaluation of the performance of acoustic recording systems for estimating avian species richness and abundance. Antonio Celis Murillo (Natural Resources and Environmental Sci., Univ. of Illinois at Urbana-Champaign, 1704 Harrington Dr., Champaign, IL 61821, celismu1@illinois.edu), Jill Deppe (Biological Sci., Eastern Illinois Univ., Champaign, IL), Jason Riddle (Natural Resources, Univ. of Wisconsin at Stevens Point, Stevens Point, WI), Michael P. Ward (Natural Resources and Environmental Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Theodore Simons (USGS cooperative fish and Wildlife Res. unit, North Carolina State Univ., Raleigh, NC)

Comparisons between field observers and acoustic recording systems have shown great promise for sampling birds using acoustics methods. Comparisons provide information about the performance of recording systems and field observers but do not provide a robust validation of their true sampling performance—i.e., precision and accuracy relative to known population size and richness. We used a 35-speaker bird song simulation system to experimentally test the accuracy and precision of two stereo (Telinga and SS1) and one quadraphonic recording system (SRS) for estimating species richness, abundance, and total abundance (across all species) of vocalizing birds. We simulated 25 bird communities under natural field conditions by placing speakers in a wooded area at 4–119 m from the center of the survey at differing heights and orientations. We assigned recordings randomly to one of eight skilled observers. We found a significant difference among microphones in their ability to accurately estimate richness ($p = 0.0019$) and total bird abundance ($p = < 0.0001$). Our study demonstrates that acoustic recording systems can potentially estimate bird abundance and species richness accurately; however, their performance is likely to vary by its technical characteristics (recording pattern, microphone arrangement, etc.).

2:55–3:15 Break

3:15

4pAB6. Spatial variation of the underwater soundscape over coral reefs in the Northwestern Hawaiian Islands. Simon E. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 7038 Old Brentford Rd., Alexandria, VA 22310, simon.freeman@gmail.com), Lauren A. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), Marc O. Lammers (Oceanwide Sci. Inst., Honolulu, HI), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Coral reefs create a complex acoustic environment, dominated by sounds produced by benthic creatures such as crustaceans and echinoderms. While there is growing interest in the use of ambient underwater biological sound as a gauge of ecological state, extracting meaningful information from recordings is a challenging task. Single hydrophone (omnidirectional) recorders can provide summary time and frequency information, but as the spatial distribution of reef creatures is heterogeneous, the properties of reef sound arriving at the receiver vary with position and arrival angle. Consequently, the locations and acoustic characteristics of individual sound producers remain unknown. An L-shaped hydrophone array, providing direction-and-range sensing capability, can be used to reveal the spatial variability of reef sounds. Comparisons can then be made between sound sources and other spatially referenced information such as photographic data. During the summer of 2012, such an array was deployed near four different benthic ecosystems in the Northwestern Hawaiian Islands, ranging from high-latitude coral reefs to communities dominated by algal turf. Using conventional and adaptive acoustic focusing (equivalent to curved-wavefront beamforming), time-varying maps of sound production from benthic organisms were created. Comparisons with the distribution of nearby sea floor features, and the makeup of benthic communities, will be discussed.

3:30

4pAB7. Density estimates of odontocetes in an active military base using passive acoustic monitoring. Bethany L. Roberts (School of Biology, Univ. of St. Andrews, Sea Mammal Res. Unit, St. Andrews, Fife KY16 8LB, United Kingdom, blr2@st-andrews.ac.uk), Zach Swaim, and Andrew J. Read (Duke Marine Lab, Duke Univ., Beaufort, NC)

We deployed passive acoustic monitoring devices in Camp Lejeune, North Carolina, USA, to estimate density of odontocete populations. Four C-PODs (echolocation click detectors) were deployed in water depths ranging from 13 to 21 meters from 30 November 2012 to 13 November 2013. Two species of odontocetes are known to inhabit the survey area: bottlenose dolphins and Atlantic spotted dolphins. These methods incorporate (i) the rate at which the animals produce echolocation cues, (ii) the probability of detecting cues, and (iii) the false positive rate of detections. To determine the cue rate of bottlenose dolphins, we attached DTAGs to 14 bottlenose dolphins during 2012 and 2013 in Sarasota, Florida. To determine cue rate of spotted dolphins, we used six recordings of focal follows from 2001-2003 in an area adjacent to C-POD deployment locations. Echolocation playbacks to C-PODs were used to obtain false positive rate and detection radius of each C-POD. Furthermore, we obtained proportions of bottlenose and spotted dolphins in the survey area from concurrent line transect surveys. Preliminary results indicate that dolphins were detected on all four C-PODs during every month of the survey period. Future studies in areas where multiple species are present could potentially use methods described here.

3:45

4pAB8. Preliminary calculation of individual echolocation signal emission rate of Franciscana dolphins (*Pontoporia blainvillei*). Artur Andriolo (Zoology Dept., Federal Univ. of Juiz de Fora, Universidade Federal de Juiz de Fora, Rua José Lourenço Kelmer, s/n - Campus Universitário Bairro São Pedro, Juiz de Fora, Minas Gerais 36036-900, Brazil, artur.andriolo@ufjf.edu.br), Federico Sucunza (Ecology Graduate Program, Federal Univ. of Juiz de Fora, Juiz de Fora, Brazil), Alexandre N. Zerbini (Ecology, Instituto Aqualie, Juiz de Fora, Brazil), Daniel Danilewicz (Zoology Graduate Program, State Univ. of Santa Cruz, Ilhéus, Brazil), Marta J. Cremer (Biological Sci., Univ. of Joinville Region, Joinville, Brazil), and Annelise C. Holz (Graduate Program in Health and Environment, Univ. of Joinville Region, Joinville, Brazil)

Calculation of echolocation signals emission rate is necessary to estimate how many individuals are vocalizing, especially if passive acoustic density estimation methods are to be implemented. We calculated the individual emission rate of echolocation signals of franciscana dolphin. Fieldwork was between 22 and 31 January of 2014 at Babitonga Bay, Brazil. Acoustic data and group size were registered when animals were within visual range at maximum distance of 50 meters. We used a Cetacean Research™ hydrophone. The sound was digitized by Analogic/Digital IOtech, stored as wav-files and analyzed with Raven software. A band limited energy detector was set to automatically extract echolocation signals. The emission rate was calculated dividing the clicks registered for each file by the file duration and by the number of individuals in the group. We analyzed 240 min of sound of 36 groups. A total of 29,164 clicks were detected. The median individual click rate was 0.290 clicks/s (10th = 0.036 and 90th = 1.166 percentiles). The result is a general approximation of the individual echolocation signal emission rate. Sound production rates are potentially dependent on a number of factors, like season, group size, sex, or even density itself. [This study was supported by IWC/Australia, Petrobras, Fundo de Apoio à Pesquisa/UNIVILLE.]

4:00

4pAB9. Investigating the potential of a wave glider for cetacean density estimation—A Scottish study. Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, The Observatory, Buchanan Gardens, St. Andrews KY16 9LZ, United Kingdom, dh17@st-andrews.ac.uk) and Douglas Gillespie (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom)

A major advantage of autonomous vehicles is their ability to provide both spatial and temporal coverage of an area during a survey. However, there is a need to assess whether these technologies are suitable for monitoring cetacean population densities. Data are presented from a Wave Glider deployed off the east coast of Scotland between March and April 2014. Key areas of survey design, data collection, and analysis were investigated. First, the ability of the glider to complete a designed line transect survey was assessed. Second, the encounter rates of all detected species were estimated. Harbour porpoise (*Phocoena phocoena*) was the most commonly encountered species and became the focal species in this study. Using the harbor porpoise encounter rate, the amount of survey effort required to estimate density with a suitable level of uncertainty was estimated. A separate experiment was designed to estimate the average probability of harbor porpoise detection by the glider. The glider was deployed near an array of nine C-PODs (odontocete detection instruments) and the same harbor porpoise click events were matched across instruments. Such matches can be analyzed using spatially explicit capture recapture methods, which allow the detection efficiency of the glider to be estimated.

4:15

4pAB10. Toward acoustically derived population estimates in marine conservation: An application of the spatially-explicit capture-recapture methodology for North Atlantic right whales. Danielle Cholewiak, Steven Brady, Peter Corkeron, Genevieve Davis, and Sofie Van Parijs (Protected Species Branch, NOAA Northeast Fisheries Sci. Ctr., 166 Water St., Woods Hole, MA 02543, danielle.cholewiak@noaa.gov)

Passive acoustics provide a flexible tool for developing understanding of the ecology and behavior of vocalizing marine animals. Yet despite a robust capacity for detecting species presence, our ability to estimate population abundance from acoustics still remains poor. Critically, abundance estimates are precisely what conservation practitioners and policymakers often require. In the current study, we explored the application of acoustic data in the spatially-explicit capture-recapture (SECR) methodology, to evaluate whether acoustics can be used to infer abundance in the endangered North Atlantic right whale. We sub-sampled a year-long acoustic dataset from archival recorders deployed in Massachusetts Bay. Multichannel data were reviewed for the presence of up-calls. A total of 1659 unique up-calls were detected. Estimates of up-call density ranged from zero to 608 ($\pm$ 70 SE) up-calls/hour. Estimates of daily abundance, when corrected for average calling rate, ranged from 0—69 ($\pm$ 21 SE) individuals per day. These results qualitatively compare well with patterns in right whale occurrence reported from aerial-based visual surveys. Since acoustic abundance calculations are affected by variation in calling behavior, estimates should be interpreted cautiously; however, these results indicate that passive acoustics has the potential to directly inform conservation and management strategies.

4:30

4pAB11. Statistical mechanics techniques applied to the analysis of humpback whale inter-call intervals. Gerald L. D'Spain (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Tyler A. Helble (SPAWAR SSC Pacific, San Diego, CA), Heidi A. Batchelor, and Dennis Rimington (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

Techniques developed in statistical mechanics recently have been applied to the analysis of the topology of complex human communication networks. These methods examine the network's macroscopic statistical properties rather than the details of individual interactions. Here, these methods are applied to the analysis of the time intervals between humpback whale calls detected in passive acoustic monitoring data collected by the bottom-mounted hydrophones on the Pacific Missile Range Facility (PMRF) west of Kauai, Hawaii. Recently developed localization and tracking algorithms for use with PMRF data have been applied to separate the calls of an individual animal from those of a collection of animals. As with the distributions of time intervals between human communications, the distributions of time intervals between humpback whale call detections are distinctly different than those expected for a purely independent, random (Poisson) process. This conclusion holds both for time intervals between calls from individual animals and from the collection of animals vocalizing simultaneously. although significant differences in these probability distributions occur. A model based on the migration of clusters of animals is developed to fit the distributions. Possible mechanisms giving rise to aspects of the distributions are discussed. [Work supported by the Office of Naval Research, Code 322-MMB.]

4:45–5:15 Panel Discussion

Session 4pBA

Biomedical Acoustics: Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications II

Vera A. Khokhlova, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Jeffrey B. Fowlkes, Cochair
Univ. of Michigan Health System, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Invited Papers

1:30

4pBA1. High intensity focused ultrasound-induced bubbles stimulate the release of nucleic acid cancer biomarkers. Tatiana Khokhlova (Medicine, Univ. of Washington, Harborview Medical Ctr., 325 9th Ave. Box 359634, Seattle, WA 98104, tdk7@uw.edu), John R. Chevillet (Inst. for Systems Biology, Seattle, WA), George R. Schade (Urology, Univ. of Washington, Seattle, WA), Maria D. Giraldez (Medicine, Univ. of Michigan, Ann Arbor, MI), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joo Ha Hwang (Medicine, Univ. of Washington, Seattle, WA), and Muneesh Tewari (Medicine, Univ. of Michigan, Ann Arbor, MI)

Recently, several nucleic acid cancer biomarkers, e.g., microRNA and mutant DNA, have been identified and shown promise for improving cancer diagnostics. However, the abundance of these biomarker classes in the circulation is low, impeding reliable detection and adoption into clinical practice. Here, the ability of HIFU-induced bubbles to stimulate release of cancer-associated microRNAs by tissue fractionation or permeabilization was investigated in a heterotopic syngeneic rat prostate cancer model. A 1.5 MHz HIFU transducer was used to either mechanically fractionate subcutaneous tumor with boiling histotripsy (BH) (~20 kW/cm^2, 10 ms pulses, and duty factor 0.01) or to permeabilize tumor tissue with inertial cavitation activity (p- = 16 MPa, 1 ms pulses, duty factor 0.001). Blood was collected immediately prior to and serially up to 24-hours after treatments. Plasma concentrations of microRNAs were measured by quantitative RT-PCR. Both exposures resulted in a rapid (within 15 min), short ($\leq$3 h) and dramatic (over ten-fold) increase in relative plasma concentrations of tumor-associated microRNAs, Histologic examination of excised tumor confirmed complete fractionation of targeted tumor by BH and localized areas of intraparenchymal hemorrhage and tissue disruption by cavitation-based treatment. These data suggest a clinically useful application of HIFU-induced bubbles for non-invasive molecular biopsy. [Grant support: NIH 1K01EB015745, R01CA154451, R01DK085714.]

1:50

4pBA2. Tissue decellularization with boiling histotripsy and the potential in regenerative medicine. Yak-Nam Wang (APL, CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynwang@u.washington.edu), Tatiana Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Adam Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Wayne Kreider (APL, CIMU, Univ. of Washington, Seattle, WA), Ari Partanen (Clinical Sci. MR Therapy, Philips Healthcare, Andover, Maryland), Navid Farr (Dept. of BioEng., Univ. of Washington, Seattle, WA), George Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), Michael Bailey (APL, CIMU, Univ. of Washington, Seattle, WA), and Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

There have been major advances in the development of replacement organs by tissue engineering (TE); however, one of the holy grails is still in the development of biomimetic structures that replicate the complex 3-D vasculature. Creation of bioartificial organs by decellularization shows greater promise in reaching the clinic compared to TE. However, current decellularization techniques require the use of chemical and biological agents, often in combination with physical force, which could result in damage to the matrix. Here we evaluate the use of boiling histotripsy (BH) to selectively decellularize large volumes of tissue. BH lesions (10–20 mm diameter) were produced in bovine liver with a clinical 1.2 MHz MR-HIFU system (Sonalleve, Philips, Finland), using thirty 10 ms pulses, and pulse repetition frequencies of 1–10 Hz. Peak acoustic powers corresponding to an estimated *in situ* shock front amplitude of 65 MPa were used. Macroscopic and histological evaluation revealed treatment conditions that produced decellularized lesions in which major fibrous structures such as stroma and vasculature remained intact while parenchymal cells were mostly lyzed. With further tailoring of the pulsing scheme parameters, this treatment modality could potentially be optimized for organ decellularization. [Work supported by NIH EB007643, K01-EB-015745-01, T32-DK007779, and NSBRI NASA-NCC 9-58.]

2:10

4pBA3. Destruction of microorganisms by high-energy pulsed focused ultrasound. Timothy A. Bigelow (Elec. and Comput. Eng., Mech. Eng., Iowa State Univ., 2113 Coover Hall, Ames, IA 50011, bigelow@iastate.edu)

The use of high-energy ultrasound pulses to generate and excite clouds of microbubbles has shown great potential to mechanically destroy soft tissue in a wide range of clinical applications. In our work, we have focused on extending the application of cavitation based histotripsy to the destruction of microorganisms such as bacteria biofilms and microalgae. Bacteria biofilms pose a significant problem when treating infections on medical implants while the fractionation of microalgae in an efficient manner could lower the production cost of biofuels. In the past, we have shown a 4.4-log10 reduction of viable *Escherichia coli* bacteria capable of forming a colony in a biofilm following a high-energy pulsed focused ultrasound exposure. We have also shown complete removal of *Pseudomonas aeruginosa* biofilms from a Pyrolytic graphite substrate based on fluorescence imaging following live/dead staining. We also showed minimal temperature increase when the appropriate ultrasound pulse parameters were utilized. Recently, we have shown that high-energy pulsed ultrasound at 1.1 MHz can fractionate the microalgae model system *Chlamydomonas reinhardtii* for lipid extraction/biofuel production in both flow and stationary exposure systems with improved efficiency over traditional sonicators. In these studies, the fractionation of the cells was quantified by protein and chlorophyll release following exposure.

Contributed Papers

2:30

4pBA4. Dependence of ablative ability of high-intensity focused ultrasound cavitation-based histotripsy on mechanical properties of agar. Jin Xu (Eng., John Brown Univ., Siloam Springs, AR), Timothy Bigelow (Elec. and Comput. Eng., Iowa State Univ., Iowa State University, 2113 Coover Hall, Ames, IA 50011, bigelow@iastate.edu), Gabriel Davis, Alex Avendano, Pranav Shrotriya, Kevin Bergler (Mech. Eng., Iowa State Univ., Ames, IA), and Zhong Hu (Elec. and Comput. Eng., Iowa State Univ., Ames, IA)

Cavitation-based histotripsy uses high-intensity focused ultrasound (HIFU) at low duty factor to create bubble clouds inside tissue to liquefy a region and provides better fidelity to planned lesion coordinates and the ability to perform real-time monitoring. The goal of this study was to identify the most important mechanical properties for predicting lesion dimensions, among these three: Young's modulus, bending strength, and fracture toughness. Lesions were generated inside tissue-mimicking agar, and correlations were examined between the mechanical properties and the lesion dimensions, quantified by lesion volume and by the width and length of the equivalent bubble cluster. Histotripsy was applied to agar samples with varied properties. A cuboid of 4.5 mm width (lateral to focal plane) and 6 mm depth (along beam axis) was scanned in a raster pattern with respective step sizes of 0.75 mm and 3 mm. The exposure at each treatment location was 15 s, 30 s, or 60 s long. Results showed that only Young's modulus influenced histotripsy's ablative ability and was significantly correlated with lesion volume and bubble cluster dimensions. The other two properties had negligible effects on lesion formation. Also, exposure time differentially affected the width and depth of the bubble cluster volume.

2:45

4pBA5. Shear waves induced by Lorentz force in soft tissues. Stefan Catheline, Graland-Mongrain Pol, Ali Zorgani, Remi Souchon, Cyril Lafon, and Jean-yves Chapelon (LabTAU, INSERM, Univ. of Lyon, 151 cours albert thomas, Lyon 69003, France, stefan.catheline@inserm.fr)

This study presents the observation of elastic shear waves generated in soft solids using a dynamic electromagnetic field. The first and second experiments of this study show that Lorentz force can induce a displacement in a soft phantom and that this displacement is detectable by an ultrasound scanner using speckle-tracking algorithms. For a 100 mT magnetic field and a 10 ms, 100 mA peak-to-peak electrical burst, the displacement reached a magnitude of 1 m. In the third experiment, we show that Lorentz force can induce shear waves in a phantom. A physical model using electromagnetic and elasticity equations is proposed and computer simulations are in good agreement with experimental results. The shear waves induced by Lorentz force are used in the last experiment to estimate the elasticity of a swine liver sample.

3:00–3:15 Break

3:15

4pBA6. Acoustic field characterization of the Waterlase2: Acoustic characterization and high speed photomicrography of a clinical laser generated shock wave therapy device for the treatment of periodontal biofilms in orthodontics and periodontics. Camilo Perez, Yak-Nam Wang (BioEng. and Ctr. for Industrial and Medical Ultrasound, CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, camipiri@uw.edu), Alina Sivriver, Dmitri Boutoussov, Vladimir Netchitailo (Biolase Inc., Irvine, CA), and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Recent applications in endodontics and periodontics use erbium solid state lasers with fiber delivery in order to effectively kill bacteria and biofilms. In this paper, the acoustic field together with the bubble dynamics of a clinical portable Er,Cr:YSGG laser-generating device (Waterlase 2) was characterized. Field mapping with a calibrated PVDF hydrophone together with high speed imaging were performed in water for two different tip geometries (flat or tapered), three different tip diameters (200, 300, or 400μm), and two different laser pulse durations (60 or 700μs) at several laser pulse energy settings (5 mJ–400 mJ) for individual pulses and at different pulse repetition frequencies (5, 20, and 100 Hz). Peak positive pressures 5–50 mm away from the tip ranged from 0.1 to 2MPa, while peak negative pressures ranged from 0.1 to 1.2 MPa. There was a strong correlation between the acoustic emissions generated by the bubble and the high speed imaging dynamics of the bubble. An initial thermoelastic response, initial bubble collapse and further rebounds where analyzed individually and compared across different test parameters. For the initial thermoelastic pulse (laser generated), pulse rise times ranged from 40 to 200 ns. Differences between flat and tapered tips will be discussed.

3:30

4pBA7. Simulations of focused shear shock waves in soft solids and the brain. Bruno Giammarinaro, François Coulouvrat, and Gianmarco Pinton (Institut Jean le Rond d'Alembert UMR 7190, CNRS, Université Pierre et Marie Curie, d'alembert, case 162, 4, Pl. Jussieu, Paris cedex 05 75252, France, bruno.giam@hotmail.fr)

Because of a very small speed, shear waves in soft solids are extremely nonlinear, with nonlinearities four orders of magnitude larger than in classical solids. Consequently, these nonlinear shear waves can transition from a smooth to a shock profile in less than one wavelength. We hypothesize that traumatic brain injuries (TBI) could be caused by the sharp gradients resulting from shear shock waves. However, shear shock waves are not currently modeled by simulations of TBI. The objective of this paper is to describe shear shock wave propagation in soft solids within the brain, with source geometry determined by the skull. A 2D nonlinear paraxial equation with cubic nonlinearities is used as a starting point. We present a numerical scheme based on a second order operator splitting which allows the application of optimized numerical methods for each terms. We then validate the scheme with Guiraud's nonlinear self-similarity law applied to cusped caustics. Once validated, the numerical scheme is then applied to a blast wave

problem. A CT measurement of the human skull is used to determine the initial conditions and shear shock wave simulations are presented to demonstrate the focusing effects of the skull geometry.

3:45

4pBA8. Tissue damage produced by cavitation: The role of viscoelasticity. Eric Johnsen (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48104, ejohnsen@umich.edu) and Matthew Warnez (Eng. Phys., Univ. of Michigan, Ann Arbor, MI)

Cavitation may cause damage at the cellular level in a variety of medical applications, e.g., therapeutic and diagnostic ultrasound. While cavitation damage to bodies in water has been studied for over a century, the dynamics of bubbles in soft tissue remain vastly unexplored. One difficulty lies in the viscoelasticity of tissue, which introduces additional physics and time scales. We developed a numerical model to investigate acoustic cavitation in soft tissue, which accounts for liquid compressibility, full thermal effects, and viscoelasticity (including nonlinear relaxation and elasticity). The bubble dynamics are represented by a Keller-Miksis formulation and a spectral collocation method is used to solve for the stresses in the surrounding medium. Our numerical studies of a gas bubble exposed to a relevant waveform indicate that under inertial conditions high pressures and velocities are generated at collapse, though they are lower than those observed in water due to the elasticity and viscosity of the medium. We further find that significant deviatoric stresses and increased heating in tissue are attributable to viscoelasticity, due to material properties and different bubble responses compared to water.

4:00

4pBA9. Comparison of Gilmore-Akulichev's, Keller-Miksis's and Rayleigh-Plesset's equations on therapeutic ultrasound bubble cavitation. Zhong Hu (Elec. and Comput. Eng., Mech. Eng., Iowa State Univ., 2201 Coover Hall, Ames, IA 50011, zhonghu@iastate.edu), Jin Xu (Eng., John Brown Univ., Siloam Springs, AR), and Timothy A. Bigelow (Elec. and Comput. Eng., Mech. Eng., Iowa State Univ., Ames, IA)

Many models have been utilized to simulate inertial cavitation for ultrasound therapies such as histotripsy. The models range from the very simple Rayleigh-Plesset model to the complex Gilmore-Akulichev model. The computational time increases with the complexity of the model, so it is important to know when the results from the simpler models are sufficient. In this paper the simulation performance of the widely used Rayleigh-Plesset model, Keller-Miksis model, and Gilmore-Akulichev model both with and without gas diffusion are compared by calculating the bubble radius response and bubble wall velocity as a function the ultrasonic pressure and frequency. The bubble oscillates similarly with the three models within the first collapse for small pressures (<3MPa), but the Keller-Miksis model diverges at higher pressures. In contrast, the maximum expansion radius of the bubble is similar at all pressures with Rayleigh-Plesset and Gilmore-AKulichev although the collapse velocity is unrealistically high with Rayleigh-Plesset model. After multiple cycles, the Rayleigh-Plesset model starts to behave disparately both in the expansion and collapse stages. The inclusion of rectified gas diffusion lengthens the collapse time and increases the expansion radius. However, for frequency smaller than 1 MHz, the impact of gas diffusion is not significant.

4:15

4pBA10. Removal of residual bubble nuclei to enhance histotripsy soft tissue fractionation at high rate. Alexander P. Duryea, Charles A. Cain (Biomedical Eng., Univ. of Michigan, 2131 Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, duryalex@umich.edu), William W. Roberts (Urology, Univ. of Michigan, Ann Arbor, MI), and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Previous work has shown that the efficacy of histotripsy soft tissue fractionation is dependent on pulse repetition frequency, with histotripsy delivered at low rates producing more efficient homogenization of the target volume in comparison to histotripsy delivered at high rates. This is attributed to the cavitation memory effect: microscopic residual cavitation nuclei that persist for hundreds of milliseconds following bubble cloud collapse can seed the repetitive nucleation of cavitation at a discrete set of sites within the target volume, producing heterogeneous lesion development. To mitigate this effect, we have developed low amplitude (MI<1) acoustic pulses to actively remove residual nuclei from the field. These bubble removal pulses utilize the Bjerknes forces to stimulate the aggregation and subsequent coalescence of remnant nuclei, consolidating the population from a very large number to a countably small number of remnant bubbles within several milliseconds. The effect is attainable in soft tissue mimicking phantoms following a very minimal degree of fractionation (within the first ten histotripsy pulses). Incorporation of this bubble removal scheme in histotripsy tissue phantom treatments at high rate (100 pulses/second) resulted in highly homogeneous lesions that closely approximated those achieved using an equal number of pulses applied at low rate (1 pulse/second); lesions generated at high rate without bubble removal had heterogeneous structure with increased collateral damage.

4:30

4pBA11. Two-dimensional speckle tracking using zero phase crossing with Riesz transform. Mohamed Khaled Almekkawy (Elec. Eng., Western New England, 2056 Knapp St., Saint Paul, MN 55108, alme0078@umn.edu), Yasaman Adibi, Fei Zheng (Elec. Eng., Univ. of Minnesota, Minneapolis, MN), Mohan Chirala (Samsung Res. America, Richardson, TX), and Emad S. Ebbini (Elec. Eng., Univ. of Minnesota, Minneapolis, MN)

Ultrasound speckle tracking provides robust estimates of fine tissue displacements along the beam direction due to the analytic nature of echo data. We introduce a new multi-dimensional ST method (MDST) with subsample accuracy in all dimensions. The algorithm based on the gradient of the magnitude and the zero-phase crossing of 2D complex correlation of the generalized analytic signal. The generalization method utilizes the Riesz transform which is the vector extension of the Hilbert transform. Robustness of the tracking algorithm is investigated using a realistic synthetic data sequences created with (Field II) for which the bench mark displacement was known. In addition, the new MDST method is used in the estimation of the flow and surrounding tissue motion on human carotid artery *in vivo*. The data was collected using a linear array probe of a Sonix RP ultrasound scanner at 325 fps. The vessel diameter has been calculated from the upper and lower vessel walls displacement, and clearly shows a blood pressure wave like pattern. The results obtained show that using Riesz transform produces more robust estimation of the true displacement of the simulated model compared to previously published results. This could have significant impact on strain calculations near vessel walls.

4:45

4pBA12. 1-MHz ultrasound stimulates *in vitro* production of cardiac and cerebrovascular endothelial cell vasodilators. Azzdine Y. Ammi (Knight Carviovascular Inst., OHSU, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, ammia@ohsu.edu), Catherine M. Davis (Dept. of Anesthesiology and Perioperative Medicine, OHSU, Portland, OR), Brian Mott (Knight Carviovascular Inst., OHSU, Portland, OR), Nabil J. Alkayed (Dept. of Anesthesiology and Perioperative Medicine, OHSU, Portland, OR), and Sanjiv Kaul (Knight Carviovascular Inst., OHSU, Portland, OR)

Ultrasound exposure of the heart and brain during vessel occlusion reduces infarct size. Our aim was to study the production of vasodilatory compounds by endothelial cells after ultrasound stimulation. A 1.05-MHz single element transducer was used to insonify primary mouse endothelial cells (ECs) from heart and brain with a 50-cycle tone burst at a pulse repetition frequency of 50 Hz. Two time points were studied after ultrasound exposure: 15 and 45 minutes. In heart ECs, EETs levels increased significantly with 0.5 MPa (139 $\pm$ 16%, $p<0.05$) and 0.3 MPa (137 $\pm$ 15%, $p<0.05$) at 15 and 45 min post stimulation, respectively. HETEs and DHETs did not change significantly. There was a trend toward increased adenosine, with maximum release at 0.5 MPa (332 $\pm$ 73% vs. 100% control, $p<0.05$). The trend toward increased eNOS phosphorylation was greater at 15 than 45 min. In brain ECs adenosine release was increased, however increased eNOS phosphorylation was not significant. 11, 12- and 14, 15- EETs were increased while 5- and 15-HETEs were decreased. Pulsed ultrasound at 1.05 MHz has the ability to increase adenosine, p-eNOS, and EET production by cardiac and cerebrovascular ECs. Interestingly, in brain ECs, the vasoconstricting HETEs were decreased.

5:00

4pBA13. Ultrasound-induced fractionation of the intervertebral disk.
Delphine Elbes, Olga Boubriak, Shan Qiao, Michael Molinari (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Jocelyn Urban (Dept. of Physiol., Anatomy and Genetics, Univ. of Oxford, Oxford, United Kingdom), Robin Cleveland, and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford, Oxfordshire, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Current surgical treatments for lower back pain, which is strongly associated with degeneration of the intervertebral disk, are highly invasive and have low long-term success rates. The present work thus aims to develop a novel, minimally invasive therapy for disk replacement without the need for surgical incision. Using *ex vivo* bovine coccygeal spinal segments as an experimental model, two confocally aligned 0.5 MHz HIFU transducers were positioned with their focus inside the disc and used to generate peak rarefactional pressures in the range of 1–12 MPa. Cavitation activity was monitored, characterized, and localized in real time using both a single-element passive cavitation detector and a 2D Passive Acoustic Mapping array. The inertial cavitation threshold in the central portion of the disk, the nucleus pulposus (NP), was first determined both in the absence and in the presence of externally injected cavitation nuclei. HIFU exposure parameters were subsequently optimized to maximize sustained inertial cavitation over 10 min and achieve fractionation of the NP. Following sectioning of treated disks, staining of live and dead cells as well as microscopy under polarized light were used to assess the impact of the treatment on cell viability and collagen structure within the NP, inner annulus and outer annulus.

THURSDAY AFTERNOON, 30 OCTOBER 2014 — MARRIOTT 9/10, 1:30 P.M. TO 4:00 P.M.

Session 4pEA

Engineering Acoustics: Acoustic Transduction: Theory and Practice II

Roger T. Richards, Chair
US Navy, 169 Payer Ln, Mystic, CT 06355

Contributed Papers

1:30

4pEA1. Vibration sensitivity measurements of silicon and acoustic-gradient microphones. Marc C. Reese (Harman Embedded Audio, Harman Int.., 6602 E 75th St. Ste. 520, Indianapolis, IN 46250, marc.reese@harman.com)

Microphones are often required to record audio while in a vibration environment. Therefore, it is important to maximize the acoustic-to-vibration sensitivity of such microphones. It has previously been shown that the vibration sensitivity of a microphone is, to first order, proportional to the mass per unit area of the diaphragm including the air loading effect. Although the air loading is generally minimal for omnidirectional condenser microphones with thick diaphragms, these measurements show that it cannot be ignored for newer silicon-based micro-electro-mechanical-system (MEMS) and acoustic-gradient microphones. Additionally, since microphone vibration sensitivities are typically not reported by microphone manufacturers, nor measured using standardized equipment, the setup of an inexpensive vibration measurement apparatus and associated challenges are discussed.

1:45

4pEA2. Non-reciprocal acoustic devices based on spatio-temporal angular-momentum modulation. Romain Fleury, Dimitrios Sounas, and Andrea Alu (ECE Dept., The Univ. of Texas at Austin, 1 University Station C0803, Austin, TX 78712, romain.fleury@utexas.edu)

Acoustic devices that break reciprocity, for instance acoustic isolators or circulators, may find exciting applications in a variety of fields, including imaging, acoustic communication systems, and noise control. Non-reciprocal acoustic propagation has typically been achieved using non-linear phenomena, which require high input power levels and introduce distorsions. In contrast, we have recently demonstrated compact linear isolation for audible airborne sound by means of angular momentum bias [Fleury *et al*., Science 343, 516 (2014)], exploiting modal splitting in a ring cavity polarized by an internal, constantly circulating fluid, whose motion is imparted using low-noise CPU fans. We present here an improved design with no moving parts, which is directly scalable to ultrasonic frequencies and fully integrable. Instead of imparting angular momentum in the form of a moving medium as in our previous approach, we make use of spatio-temporal acoustic modulation of three coupled acoustic cavities, a strategy that can be readily implemented in integrated ultrasonic devices, for instance, using piezoelectric effects. In this new paradigm, the required modulation frequency is orders of magnitude lower than the signal frequency, and the modulation efficiency is maximized. This constitutes a pivotal step towards practically realizing compact, linear, noise-free, tunable non-reciprocal acoustic components for full-duplex acoustic communications and isolation.

2:00

4pEA3. An analysis of multi-year acoustic and energy performance data for bathroom and utility residential ventilation fans. Wongyu Choi, Antonio Gomez, Michael B. Pate, and James F. Sweeney (Mech. Eng., Texas A&M Univ., 2401 Welsh Ave. Apt. 615, 615, College Station, TX 77845, wongyuchoi@tamu.edu)

Loudness levels have been established as a new requirement in residential ventilation standards and codes including ASHRAE and IECC. Despite the extensive application of various standards and codes, the control of loudness has not been a common target in past whole-house ventilation standards and codes. In order to evaluate the appropriate loudness of ventilation fans, especially in terms of leading standards and codes, a statistical analysis is necessary. Therefore, this paper provides statistical data for bathroom and utility ventilation fans over a nine year period from 2005 to 2013. Specifically, this paper presents an evaluation of changes in fan loudness over the 9 year test period and the relevance of loudness to leading standards including HVI and ASHRAE. The loudness levels of brushless DC-motor fans are also evaluated in comparison to the loudness of AC-motor fans. For AC and DC motor fans, relationships between loudness and efficacy was determined and then explained with regression models. Based on observations, this paper introduces a new "loudness-to-energy ratio" coefficient, L/E, which is a measure of the acoustic and energy performance of a fan. Relationships between acoustic and energy performances are established by using L/E coefficients with supporting statistics for bathroom and utility fans.

2:15

4pEA4. Non contact ultrasound stethoscope. Nathan Jeger, Mathias Fink, and Ros Kiri Ing (Institut Langevin, ESPCI ParisTech, 1 rue Jussieu, Paris 75005, France, nathan.jeger@espci.fr)

Heartbeat and respiration are very important vital signs that indicate health and psychological states of a person. Recent technologies allow to detect both physical parameters on a human subject by using different techniques with and without contact. Noncontact systems often use electromagnetic waves for contactless measurement but approaches based on ultrasound waves, laser or video processes are also proposed. In this abstract an alternative ultrasound system for non-contact and local measurement is presented. The system works in echographic mode and ultrasound signals are processed using two methods. The experimental setup uses an elliptic mirror to focus ultrasonic waves onto the skin surface. Backscattered waves are recorded by a microphone located close to the emitting transducer. Heartbeat and respiration signals are determined from the skin displacement caused by the chest-wall motion. For comparison purpose, the cross-correlation method, which uses broadband signal, and the Doppler method, which uses narrowband signal, are applied to measure the skin displacement. Sensitivity and accuracy parameters of the two methods are compared. At least, as the measurement is local, the system can act as a noncontact stethoscope to listen the internal sounds of the human body even through the light clothes of the patient.

2:30

4pEA5. High sensitivity imaging of resin-rich regions in graphite/epoxy laminates using joint entropy. Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, School of Medicine Campus Box 8215, St. Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Mathematics, Washington Univ., St. Louis, MO), Jon Marsh, and Samuel Wickline (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO)

The continuing difficulty of detecting critical flaws in advanced materials requires novel approaches that enhance sensitivity to defects that might impact performance. This study compares different approaches for imaging a near-surface resin-rich defect in a thin graphite/epoxy plate using backscattered ultrasound. The specimen, having a resin-rich void immediately below the top surface ply, was scanned with a 1 in. dia., 5 MHz center frequency, and 4 in. focal length transducer. A computer controlled apparatus comprised of an x-y-z motion controller, a digitizer (LeCroy 9400A), and an ultrasonic pulser/receiver (Panametrics 5800) was used to acquire data on a 100×100 grid of points covering a 3×3 in. square. At each grid point 256 512-word, 8-bit backscattered waveforms, were digitized, signal averaged, and then stored on computer for off-line analysis. The same backscattered waveforms were used to produce peak-to-peak, signal energy, as well as entropy images. All of the entropy images exhibit better border delineation and defect contrast than the either peak-to-peak or signal energy. The best results are obtained using the joint entropy of the backscattered waveforms with a reference function. Two different references are examined: a reflection from a stainless steel reflector, and an approximate optimum obtained from an iterative parametric search. The joint entropy images produced using the optimum reference exhibit ~3 times the contrast obtained in previous studies.

2:45

4pEA6. New compensation factors for the apparent propagation speed In transmission line matrix uniform grid meshes. Alexandre Brandao (Graduate Program in Elec. and Telecommunications Eng., Universidade Federal Fluminense, Rua Passo da Patria, 156, Sao Domingos, Niteroi, RJ 24210-240, Brazil, abrand@operamail.com), Edson Cataldo (Appl. Mathematics Dept., Universidade Federal Fluminense, Niteroi, RJ, Brazil), and Fabiana R. Leta (Mech. Eng. Dept., Universidade Federal Fluminense, Niteroi, RJ, Brazil)

Numerical models consisting of two-dimensional (2D) and three-dimensional (3D) uniform grid meshes for the Transmission Line Matrix Method (TLM), use sqrt(2) and sqrt(3), respectively, to compensate for the apparent sound speed. In this work, new compensation factors are determined from a priori simulations, performed without compensation, in 2D and 3D TLM one-section cylindrical waveguide acoustic models. The mistuned resonance peaks obtained from these simulations are substituted in the corresponding equations for the resonance frequencies in one-section cylindrical acoustical waveguides to find the mesh apparent sound speed and, thus, the necessary compensation. The TLM meshes are constructed over the voxels (Volumetric Picture Elements) of segmented MRI volumes, so that the extracted mesh fits the segmented object. The TLM method provides a direct simulation approach instead of solving a PDE by variational methods that must consider the plane wave assumption to run properly. Results confirm the improvement over the conventional compensation factors, particularly for frequencies above 4 kHz, providing a concrete reduction of the topology-dependent numerical dispersion for both 2D and 3D TLM lattices. Since this dispersion problem is common to all TLM applications using uniform grids, investigators in other areas of wave propagation can also benefit from these findings.

3:00–3:15 Break

3:15

4pEA7. A low-cost alternative power supply for integrated electronic piezoelectric transducers. Ricardo Brum, Sergio L. Aguirre, Stephan Paul, and Fernando Corrêa (Centro de Tecnologia, Universidade Federal de Santa Maria, Rua Erly de Almeida Lima, 650, Santa Maria, RS 97105-120, Brazil, ricardozbrum@yahoo.com.br)

Comercial hardware compatible with IEPE precision sensors normally are expensive and often coupled to proprietary and expensive software packages. commercially available sound cards are a low cost option for AD, but are incompatible with IEPE sensors. To create 4 mA constant current for IEPE transducers commercial solutions are available and labs also have created such solutions, e.g., ITA at RWTH Aachen University. Unfortunately, commercially available circuits are still to expensive for large scale classroom use in Brazil and circuits created elsewhere contain parts subject to US export restrictions or require machines for creation of circuits. Thus, based on a previous project, a new low-cost prototype was mounted on phenolic board. The circuit was tested with an IEPE microphone connected to a commercial soundcard and ITA-Toolbox software and compared to a commercial hardware/software package. The results were very similar in the frequency range between 20 Hz and 10 kHz. The difference below 20 Hz probably occurs due the different high pass filters in the AD-cards. The differences in the high frequency range are very likely due to differences in the electrical background noise. The results suggest the device works well and is a good alternative to make measurements with IEPE sensors.

3:30

4pEA8. Determination of the characteristic impedance and the complex wavenumber of an absorptive material used in dissipative silencer. Key F. Lima, Nilson Barbieri (Mech. Eng., PUCPR, Imaculada Conceição, 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com), and Renato Barbieri (Mech. Eng., UDESC, Joinville, Brazil)

The silencers are acoustic filters that have the purpose of reducing unwanted noise emitted by engines or equipment to acceptable levels. The vehicular silencers have large volume and dissipative properties. Dissipative silencers have absorptive material inside. These materials are typically fibrous and have good acoustic dissipation. However, few works depict the acoustic behavior of silencers with absorptive materials. The difficulty in evaluating this type of silencer is determining the acoustic properties of the absorptive material: the characteristic impedance and the complex wavenumber. This work shows an inverse methodology for determining the acoustic properties of the absorptive material used in silencers. First, it is found the silencer's acoustic efficiency in terms of the experimental sound transmission loss. Second, the absorptive material properties are determined with a parameters adjustment through a direct search optimization algorithm. In this step, the adjustment is done by applying The Finite Element Method in the search for the silencer's computational efficiency. The final step is to verify the difference between the experimental and computational results. For this work is used the acoustic efficiency of a silencer that has already been published in the literature. The results show good agreement.

3:45

4pEA9. Flat, lightweight, transparent acoustic transducers based on dielectric elastomer and gel. Kun Jia (The State Key Lab. for Strength and Vib. of Mech. Structures, Xian Jiaotong Univ., South 307 Rm., 1st Teaching Bldg.,West of the Xianning Rd. No.28, Xian, Shannxi 710049, China, kunjia@mail.xjtu.edu.cn)

The advances in flat-panel displays and Super Hi-Vision with a 22.2 multichannel sound system exhibit an entirely new viewing and listening environment for the audience; however, flat and lightweight acoustic transducers are required to fulfill this prospect. In this paper, a flat lightweight acoustic transducer with a rather simple structure is proposed. Polyacrylic elastomer membrane (VHB4905, 3M corporation) with 4 mm diameter, 0.5 mm thickness is biaxially prestretched and fixed on a polyurethane ring as the vibrator, then ionic gel is painted on the center region of the membrane as electrodes, finally, conducting wires which are also made by ionic gel is attached to the edge of the electrodes for applying the AC voltage with a DC bias. The ultrahigh transmittance of the VHB4905, gel, and polyurethane makes the transducer totally transparent, which is of great interest in advanced media technology. The dynamic properties of the membrane are studied experimentally along with its acoustic performance. It has been found that the behavior of the dielectric elastomer membrane is quite complicated, both of the in plane and out of plane vibration mode exist. The transducer shows better performance below 10 kHz for the low elastic modular of the membrane.

THURSDAY AFTERNOON, 30 OCTOBER 2014 SANTA FE, 1:00 P.M. TO 4:10 P.M.

Session 4pMU

Musical Acoustics: Assessing the Quality of Musical Instruments

Andrew C. H. Morrison, Chair
Joliet Junior College, 1215 Houbolt Rd., Natural Science Department, Joliet, IL 60431

Invited Papers

1:00

4pMU1. Bamboo musical instruments: Some physical and mechanical properties related to quality. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

Bamboo is one of the most widely used materials in musical instruments, including string instruments and percussion as well as wind instruments. Bamboo pipe walls are complex, composed of a layered structure of fibers. The pipe walls exhibit non-uniformity in radial structure and density, and there is a significant difference between the elastic moduli parallel to and perpendicular to the bamboo fibers. This paper presents a summary of results from the literature on bamboo as a material for musical instruments. In addition, results are presented from recent measurements of the physical and mechanical properties of materials used in some typical instruments. In particular, a case study will be presented comparing measurements made on reeds and pipes from two Southeast Asian khaen. Of the two khaen discussed, one is a high quality khaen made by craftsmen in northeastern Thailand, while the other is an inexpensive instrument purchased at an import shop. For this pair of instruments, analysis and comparison have been made of the material properties of the bamboo pipes and the composition and mechanical properties of the metal alloy reeds.

1:20

4pMU2. Descriptive maps to illustrate the quality of a clarinet. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu), Philippe Guillemain, Jean-Baptiste Doc, Alexis Guilloteau, and Christophe Vergez (Laboratroire de mécanique et d'acoustique, Marseille, France)

Generally, subjective opinions and decisions are made when judging the quality of musical instruments. In an attempt to become more objective, this research presents methods to numerically and experimentally create maps, over a range of control parameters, that describe instrument behavior for a variety of different sounds features or "quality markers" (playing regime, intonation, loudness, etc.). The behavior of instruments is highly dependent on the control parameters that are adjusted by the musician. Observing this behavior as a function of one control parameter (e.g., blowing pressure) can hide diversity of the overall behavior. An isovalue quality marker can be obtained for a multitude of control parameter combinations. Using multidimensional maps, where quality markers are a function of two or more control parameters, can solve this problem. Numerically: in two dimensions, a regular discretization on a subspace of control parameters can be implemented while conserving a reasonable calculation time. However, in higher dimensions (if, for example, aside from the blowing pressure and the lip force, we vary the reed parameters), it is necessary to use auto-adaptive sampling methods. Experimentally: the use of an artificial mouth allows us to maintain control conditions while creating these maps. We can also use an instrumented mouthpiece: this allows us to measure simultaneously and instantly these control parameters and create the maps "on the fly."

1:40

4pMU3. Recent works on the (psycho-)acoustics of wind instruments. Adrien Mamou-Mani (IRCAM, 1 Pl. Stravinsky, Paris 75004, France, adrien.mamou-mani@ircam.fr)

Two experiments aiming at linking acoustical properties and perception of wind instruments will be presented. The first one is a comparison between five oboes of the same model type. An original methodology is proposed, based on discrimination tests in playing conditions and default detection using acoustical measurements. The second experiment has been done on a simplified bass clarinet with an embedded active control system. A comparison of perceptual attributes, like sound color and playability, for different acoustical configurations (frequency and damping of resonances) is possible to test using a single system. A specific methodology and first results will be presented.

2:00

4pMU4. The importance of structural vibrations in brass instruments. Thomas R. Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu) and Wilfried Kausel (Inst. of Musical Acoust., Univ. of Music and Performing Arts, Vienna, Austria)

It is often thought that the input impedance uniquely determines the quality of a brass wind instrument. However, it is known that structural vibrations can also affect the playability and perceived sound produced by these instruments. The processes by which the structural vibrations affect the quality of brass instruments are not completely understood, but it is likely that vibrations of the metal couple to the lips as well as introducing small changes in the input impedance. We discuss the mechanisms by which structural vibrations can affect the quality of a brass instrument and suggest methods of incorporating these effects into an objective assessment of instrument quality.

2:20–2:40 Break

Contributed Papers

2:40

4pMU5. Investigating the colloquial description of sound by musicians and non-musicians. Jack Dostal (Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

What is meant by the words used in a subjective judgment of sound? Interpreting these words accurately allows these musical descriptions of sound to be related to scientific descriptions of sound. But do musicians, scientists, instrument makers, and others mean the same things by the same words? When these groups converse about qualities of sound, they often use an expansive lexicon of terms (bright, brassy, dark, pointed, muddy, etc.). It may be inaccurate to assume that the same terms and phrases have the same meaning to these different groups of people or even remain self-consistent for a single individual. To investigate the use of words and phrases in this lexicon, subjects with varying musical and scientific backgrounds were surveyed. The subjects were asked to listen to different pieces of recorded music and asked to use their own colloquial language to describe the musical qualities and differences they perceived in these pieces. In this talk, I describe some qualitative results of this survey and identify some of the more problematic terms used by these various groups to describe sound quality.

2:55

4pMU6. Chaotic behavior of the piccolo? Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu)

A direct numerical solution of the Navier-Stokes equations has been used to calculate the sound produced by a model of the piccolo. At low to moderate blowing speeds and at appropriate blowing angles, the sound pressure is approximately periodic with the expected frequency. As the blowing speed is increased or as the blowing angle is varied, the time dependence of the sound pressure becomes more complicated, and examination of the spectrum and the sensitivity of the sound pressure to initial conditions suggest that the behavior becomes chaotic. Similarities with the behavior found in Taylor-Couette and Rayleigh-Bènard instabilities of fluids are noted and possible implications for the nature of the piccolo tone are discussed.

3:10

4pMU7. Modeling the low-frequency response of an acoustic guitar. Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., PO Box 30, mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

The low-frequency response of an acoustic guitar is strongly influenced by the combined behavior of the air cavity and the top plate. The sound hole–air cavity resonance (often referred to as the Helmholtz resonance) interacts with the first elastic mode of the top plate creating a coupled oscillator with two resonance frequencies that are shifted away from the frequencies of the two original, uncoupled oscillators. This effect was modeled using finite elements for the top plate and boundary elements for the air cavity with rigid sides and back and no strings. The natural frequencies of the individual and combined oscillators were then predicted and compared to measurements. The model predicts the mode shapes, natural frequencies, and damping well thus validating the modeling procedure. The effect of changing the cavity volume was then simulated to predict the behavior for a deeper air cavity.

3:25

4pMU8. Experiment to evaluate musical qualities of violin strings. Maxime Baelde (Acoust. & Environ. HydroAcoust. Lab, Université Libre de Bruxelles, 109 rue Barthélemy Delespaul, Lille 59000, France, maxime.baelde@centraliens-lille.org), Jessica De Saedeleer (Jessica De Saedeleer Luthier, Brussels, Belgium), and Jean-Pierre Hermand (Acoust. & Environ. hydroAcoust. lab, Université Libre de Bruxelles, Brussels, Belgium)

Most of violin strings on the market are made of different materials and size. They have different musical qualities: full, mellow, warm, and round, for example. Nevertheless, this description is subjective and related to string manufacturers. The aim of this study is to provide an experiment which gives an evaluation of the musical qualities of strings. This study is based on "musical descriptors," which gives information about a musical sound and psychoacoustics in order to match the musician point of view. "Musical descriptors" are also used for music classification. We use two sets of top-end strings model from two different brands. These strings are mounted on two similar violins and the strings are excited on their normal modes with harpsichord damper mechanism like and other means. The sound radiated is

recorded with a microphone and the vibration of the string with a coil-magnet device so as to have intrinsic and extrinsic string properties. Some musicians tried these strings and expressed what they thought about it. These acoustical and psychoacoustical analyzes will give information to the luthiers to know what string property allow one adjustment, in order to provide better advice aside from string manufacturers descriptions.

3:40

4pMU9. Vibration study of Indian folk instrument sambal. Ratnaprabha F. Surve (Phys., Nowrosjee Wadia College, 15 Tulip, Neco Gardens, Viman Nagar, Pune 21 411001, India, rfsurve@hotmail.com), Keith Desa (Phys., Nowrosjee Wadia College, 27, Maharashtra, India), and Dilip S. Joaj (Phys., Univ. of Pune, Pune, Maharashtra, India)

The percussion instruments family, in its folk category has many instruments like Dholki, Dimdi, Duff, Halagi, and Sambal. The Sambal is a folk membranophone made up of wood, played mainly in western India. Sambal a traditional drum, which is used in some religious functions. It is played by the people who are believed to be servants of goddess Mahalaxmi Devi. This instrument is made up of two approximately cylindrical wooden drums united along a common edge, having skin membranes stretched over their mouths. This instrument is played using two wooden sticks, of which one has a curved end. The right hand side drum's pitch is higher than the left. Its membrane is excited by striking repeatedly to generate sound of a constant pitch. This paper relates to vibrational analysis of the Sambal. A study has been carried out to check it's vibrational properties like modes of the vibration. The study is done by spectrum analysis (Fast Fourier Transform) using a simple Digital Storage Oscilloscope. The tonal quality of wood used for the cylinders and membrane is compared.

3:55

4pMU10. Experimental investigation of crash cymbal acoustic quality. Devyn P. Curley (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155), Zachary A. Hanan (Elec. Eng., Univ. of Colorado, Boulder, CO), Dan Luo (Mech. Eng., Tufts Univ., Medford, MA), Christopher W. Penny (Phys., Tufts Univ., Medford, MA), Christopher F. Rodriguez (Elec. and Comput. Eng., Tufts Univ., Medford, MA), Paul D. Lehrman (Music, Tufts Univ., Medford, MA), Chris B. Rogers, and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA, r.white@tufts.edu)

A methodology to quantitatively evaluate the quality of the transmitted acoustic signature of cymbals is under development. High speed video recordings of a percussionist striking both a Zildjian 14 in. A-custom crash cymbal and a Zildjian Gen 16 low volume 16 in. crash cymbal were recorded and used to determine biometrically accurate crash and ride striking motions. A two degree of freedom robotic arm has been developed to mimic human striking motion. The robotic arm includes a high torque elbow joint driven in closed loop trajectory tracking and an impedance controlled wrist joint to approximate the variable stiffness of the stick grip. A quantitative comparison of robotic and human strikes will be made using high speed video. Repeatable strikes will be carried out using the robotic system in an anechoic chamber for different grades of Zildjian cymbals, including low volume Gen 16 cymbals. Acoustic features of the measured sound output will be compared to seek quantitative metrics for evaluating cymbal sound quality that compare favorably with the results of qualitative human assessments that are currently in use by the industry. Preliminary results indicate noticeable differences in cymbal acoustic output including variations in modal density, decay time, and beating phenomena.

THURSDAY AFTERNOON, 30 OCTOBER 2014 — MARRIOTT 3/4, 1:15 P.M. TO 4:20 P.M.

Session 4pNS

Noise: Virtual Acoustic Simulation

Stephen A. Rizzi, Cochair
NASA Langley Research Center, 2 N Dryden St, MS 463, Hampton, VA 23681

Patricia Davies, Cochair
Ray W. Herrick Labs., School of Mechanical Engineering, Purdue University, 177 South Russell Street, West Lafayette, IN 47907-2099

Chair's Introduction—1:15

Invited Papers

1:20

4pNS1. Recent advances in aircraft source noise synthesis. Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., 2 N Dryden St., MS 463, Hampton, VA 23681, stephen.a.rizzi@nasa.gov), Daniel L. Palumbo (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Jonathan R. Hardwick (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), and Andrew Christian (National Inst. of Aerosp., Hampton, VA)

For several decades, research and development has been conducted at the NASA Langley Research Center directed at understanding human response to aircraft flyover noise. More recently, a technology development effort has focused on the simulation of aircraft flyover noise associated with future, large commercial transports. Because recordings of future aircraft are not available, the approach taken utilizes source noise predictions of engine and airframe components which serve as a basis for source noise syntheses. Human subject response studies have been conducted aimed at determining the fidelity of synthesized source noise, and the annoyance and detectability once the noise is propagated (via simulation) to the ground. Driven by various factors, human response to less common noise sources are gaining interest. Some have been around for a long time (rotorcraft), some have come and gone, and are back again (open rotors), and some are entirely new (distributed electric driven propeller systems). Each has unique challenges associated with source noise synthesis. Discussed in this work are some of those challenges including source noise characterization from wind tunnel data, flight data, or prediction; factors affecting perceptual fidelity including tonal/broadband separation, and amplitude and frequency modulation; and a potentially expansive range of operating conditions.

1:40

4pNS2. An open architecture for auralization of dynamic soundscapes. Aric R. Aumann (Analytical Services & Mater., Inc., 107 Res. Dr., Hampton, VA 23666-1340, aric.r.aumann@nasa.gov), William L. Chapin (AuSIM, Inc., Mountain View, CA), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

An open architecture for auralization has been developed by NASA to support research aimed at understanding human response to sound within a complex and dynamic soundscape. The NASA Auralization Framework (NAF) supersedes an earlier auralization tool set developed for aircraft flyover noise auralization and serves as a basis for a future auralization plug-in for the NASA Aircraft Noise Prediction Program (ANOPP2). It is structured as a set of building blocks in the form of dynamic link libraries, so that other soundscapes, e.g., those involving ground transportation, wind turbines, etc., and other use cases, e.g., inverse problems, may easily be accommodated The NAF allows users to access auralization capabilities in several ways. The NAF's built-in functionality may be exercised utilizing either basic (e.g., console executable) or advanced (e.g., MATLAB, LabView, etc.) host environments. The NAF's capabilities can also be extended by augmenting or replacing major activities through programming its open architecture. In this regard, it is envisioned that third parties will develop plug-in capabilities to augment those included in the NAF.

2:00

4pNS3. Simulated sound in advanced acoustic model videos. Kenneth Plotkin (Wyle, 200 12th St. South, Ste. 900, Arlington, VA 22202, kenneth.plotkin@wyle.com)

The Advanced Acoustic Model (AAM) and other time-step aircraft noise simulation models developed by Wyle can generate video animations of the noise environment. The animations are valuable for understanding details of noise footprints and for community outreach. Using algorithms developed by NASA, audio simulation for jet aircraft noise has recently been added to the video capability. Input data for the simulations consist of AAM's one-third octave band sound level time history output, plus flight path geometry and ground properties. Working at an audio sample rate of 44.1 kHz and a sample "hop" period of 0.0116 s, a random phase narrow band sample is shaped to match spectral amplitudes. Ground reflection and low frequency oscillation are added to the hops, which are merged into a WAV file. The WAV file is then mixed with an existing animation generated from the same AAM run. The process takes place in near-real time, based on a location that a user selects from a site map. The presentation includes demonstrations of the results for a simple level flyover and for the departure of a high performance jet aircraft from an airbase.

2:20

4pNS4. Combining local source propagation modeling results with a global acoustic ray tracer. Michael Williams, Darrel Younk, and Steve Mattson (Great Lakes Sound and Vib., 47140 N. Main St., Houghton, MI 49931, mikew@glsv.com)

A common method of sound auralization in large virtual environments is through acoustic ray tracing. The purpose of an acoustic ray tracer is to supply accurate source to listener impulse response functions for a virtual scene. Currently, sources are modeled as an omnidirectional point source in the ray tracer. This limits the fidelity of the results and is not accurate for complicated noise sources involving multiple audible parts. The proposed method is to simulate local source propagation to a sphere using various energy modeling techniques. These results may be used to increase the fidelity of a ray trace by giving directionality to the source and allowing for source audio to be mixed from recordings of components of the source. This is especially relevant when a full source has not yet been constructed. Because of this, there are many real world applications in engineering, architecture, and other fields that need high fidelity auralization of future products.

2:40

4pNS5. Modelling sound propagation in the presence of atmospheric turbulence for the auralization of aircraft noise. Frederik Rietdijk, Kurt Heutschi (Acoust. / Noise Control, Empa, Überlandstrasse 129, Dübendorf, Zurich 8600, Switzerland, frederik.rietdijk@empa.ch), and Jens Forssén (Appl. Acoust., Chalmers Univ. of Technol., Gothenburg, Sweden)

A new tool for the auralization of aircraft noise in an urban environment is in development. When listening to aircraft noise sound level fluctuations caused by atmospheric turbulence are clearly audible. Therefore, to create a realistic auralization of aircraft noise, atmospheric turbulence needs to be included. Due to spatial inhomogeneities of the wind velocity and temperature in the atmosphere acoustic scattering occurs, affecting the transfer function between source and receiver. Both these inhomogeneities and the aircraft position are time-dependent, and therefore the transfer function varies with time resulting in the audible fluctuations. Assuming a stationary (frozen) atmosphere, the movement of the aircraft alone gives rise to fluctuations. A simplified model describing the influence of turbulence on a moving elevated source is developed, which can then be used to simulate the influence of atmospheric turbulence in the auralization of aircraft noise.

3:00–3:20 Break

3:20

4pNS6. Simulation of excess ground attenuation for aircraft flyover noise synthesis. Brian C. Tuttle (Analytical Mech. Assoc., Inc., 1318 Wyndham Dr., Hampton, VA 23666, btuttle1@gmail.com) and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

Subjective evaluations of noise from proposed aircraft and flight operations can be performed using simulated flyover noise. Such simulations typically involve three components: generation of source noise, propagation of that noise to a receiver on or near the ground, and reproduction of that sound in a subjective test environment. Previous work by the authors focused mainly on development of high-fidelity source noise synthesis techniques and sound reproduction methods while assuming a straight-line propagation path with standard atmospheric absorption and simple (plane-wave) ground reflection models. For aircraft community noise applications, this is usually sufficient because the aircraft are nearly overhead. However, when simulating noise sources at low elevation angles, the plane-wave assumption is no longer valid and must be replaced by a model that takes into account the reflection of spherical waves from a ground surface of finite impedance. Recent additions to the NASA Community Noise Test Environment (CNoTE) software suite have improved real-time simulation capabilities of ground-plane reflections for low incidence angles. The models are presented along with the resulting frequency response of the filters representing excess ground attenuation. Discussion includes an assessment of the performance and limitations of the filters in a real-time simulation.

3:40

4pNS7. Evaluation of the perceptual fidelity of a novel rotorcraft noise synthesis technique. Jonathan R. Hardwick (Dept. of Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), Andrew Christian (National Inst. of Aerosp., 100 Exploration Way, Hampton, VA 23666, andrew.christian@nasa.gov), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

A human subject experiment was recently conducted at the NASA Langley Research Center to evaluate the perceptual fidelity of synthesized rotorcraft source noise. The synthesis method combines the time record of a single blade passage (i.e., of a main or tail rotor) with amplitude and frequency modulations observed in recorded rotorcraft noise. Here, the single blade passage record can be determined from a time-averaged recording or from a modern aeroacoustic analysis. Since there is no predictive model available, the amplitude and frequency modulations were derived empirically from measured flyover noise. Thus, one research question was directed at determining the fidelity of four synthesis implementations (unmodulated and modulated main rotor only, and unmodulated and modulated main and tail rotor) under thickness and loading noise dominated conditions, using modulation data specific to those conditions. A second research question was aimed at understanding the sensitivity of fidelity to the choice of modulation method. In particular, can generic modulation data be used in lieu of data specific to the condition of interest, and how do modifications of generic and specific modulation data affect fidelity? The latter is of importance for applying the source noise synthesis to the simulation of complete flyover events.

4:00

4pNS8. A comparison of subjects' annoyance ratings of rotorcraft noise in two different testing environments. Andrew McMullen and Patricia Davies (Purdue Univ., 177 S Russel Dr, West Lafayette, IN 47906, almvz5@mail.missouri.edu)

Two subjective tests were conducted to investigate people's responses to rotorcraft noise. In one test subjects heard the sounds in a room designed to simulate aircraft flyovers. The frequency range of the Exterior Effects Room (EER) at NASA Langley is 17 Hz to 18,750 Hz. In the other test, subjects heard the sounds over earphones and the frequency range of the playback was 25 Hz–16 kHz. Some of the sounds in this earphone test, high-pass filtered at 25 Hz, were also played in the EER. Forty subjects participated in each of the tests. Subjects' annoyance responses in each test were highly correlated with EPNL, ASEL, and Loudness exceeded 20% of the time (correlation coefficient close to 0.9). However, at some metric values there was a large variation in response levels, which could be linked to characteristics of harmonic families present in the sound. While the results for both tests are similar, subjects in the EER generally found the sounds less annoying than the subjects who heard the sounds over earphones. Certain groups of signals were rated similarly in one test environment, but differently in the other. This may be due to playback method, subject population, or other factors.

Session 4pPA

Physical Acoustics: Topics in Physical Acoustics II

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

William Slaton, Cochair
Physics & Astronomy, The University of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034

Contributed Papers

1:30

4pPA1. Aeroacoustic response of coaxial Helmholtz resonators in a low-speed wind tunnel. William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, wvslaton@uca.edu)

The aeroacoustic response of coaxial Helmholtz resonators with different neck geometries in a low-speed wind tunnel has been investigated. Experimental test results of this system reveal strong aeroacoustic response over a Strouhal number range of 0.25–0.1 for both increasing and decreasing the flow rate in the wind tunnel. Ninety-degree bends in the resonator necks does not significantly change the aeroacoustic response of the system. Aeroacoustic response in the low-amplitude range has been successfully modeled by describing-function analysis. This analysis, coupled with a turbulent flow velocity distribution model, gives reasonable values for the location in the flow of the undulating stream velocity that drives vortex shedding at the resonator mouth. Having an estimate for the stream velocity that drives the flow-excited resonance is crucial when employing the describing-function analysis to predict aeroacoustic response of resonators.

1:45

4pPA2. Separation of acoustic waves in isentropic flow perturbations. Christian Henke (ATLAS ELEKTRONIK, Sebaldsbruecker Heerstrasse 235, Bremen 28309, Germany, christian.henke@atlas-elektronik.com)

The present contribution investigates the mechanisms of sound generation and propagation in the case of highly-unsteady flows. It is based on the linearisation of the isentropic Navier-Stokes equation around a new path-line-averaged base flow. As a consequence of this unsteady and non-radiating base flow, the perturbation equations satisfy a conservation law. It is demonstrated that this flow perturbations can be split into acoustic and vorticity modes, with the acoustic modes being independent of the vorticity modes. Moreover, we conclude that the present acoustic perturbation is propagated by the convective wave equation and fulfills Lighthill's acoustic analogy. Therefore, we can define the deviations from the convective wave equation as the "true" sound sources. In contrast to other authors, no assumptions on a slowly varying or irrotational flow are necessary.

2:00

4pPA3. The sliding mode controller on the rijke-type combustion systems with mean temperature gradients. Dan Zhao and Xinyan Li (Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave. Singapore, Singapore 639798, Singapore, xli037@e.ntu.edu.sg)

Thermoacoustic instabilities are typically generated due to the dynamic coupling between unsteady heat release and acoustic pressure waves. To eliminate thermoacoustic instability, the coupling must be somehow interrupted. In this work, we designed and implemented a sliding mode controller to mitigate self-sustained thermoacoustic oscillations in a Rijke-type combustion system. An acoustically-compact heat source is confined and modeled by using a modified King's Law. The mean temperature gradient is considered by expanding the acoustic waves via Galerkin series. Coupling the unsteady heat release with the acoustic model enables the flow disturbances to be calculated, thus providing a platform on which to evaluate the performance of the controller. As the controller is actuated, the limit cycle oscillations are quickly dampened and the thermoacoustic system with multiple eigenmodes is stabilized. The successful demonstration indicates that the sliding mode controller can be applied to stabilize unstable thermoacoustic systems.

2:15

4pPA4. Feedback control of thermoacoustic oscillation transient growth of a premixed Laminar flame. Dan Zhao and Xy Li (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore, XLI037@e.ntu.edu.sg)

Transient growth of combustion-excited oscillations could trigger thermoacoustic instability in a combustion system with nonorthogonal eigenmodes. In this work, feedback control of transient growth of combustion-excited oscillation in a simplified thermoacoustic system with Dirichlet boundary conditions is considered. For this a thermoacoustic model of a premixed laminar flame with an actuator is developed. It is formulated in state-space by expanding acoustic disturbances via Galerkin series and linearizing flame model and recasting it into the classical time-lag N-τ for controllers implementation. As a linear-quadratic-regulator (LQR) controller is implemented, the system becomes asymptotically stable. However, it is associated with transient growth of thermoacoustic oscillations, which may potentially trigger combustion instability. To eliminate the oscillations transient growth, a strict dissipativity controller is then implemented. Comparison is then made between the performances of these controllers. It is found that the strict dissipativity controller achieves both exponential decay of the oscillations and unity maximum transient growth.

2:30

4pPA5. Nonlinear self-sustained thermoacoustic instability in a combustor with three bifurcating branches. Dan Zhao and Shihuai Li (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore, LISH0025@e.ntu.edu.sg)

In this work, experimental investigations of a bifurcating thermoacoustic system are conducted first. It has a mother tube splitting into three bifurcating branches. It is surprisingly found that the flow oscillations in the bifurcating branches resulting from unsteady combustion in the bottom stem are at different temperatures. Flow visualization reveals that one branch is associated with "cold" pulsating flow, while the other two branches are "hot." Such unique flow characteristics cannot be predicted by simply assuming the bifurcating combustor consisting of three curved Rijke tube. 3D Numerical investigations are then conducted. Three parameters are identified and studied one by one: (1) the heat source location, (2) the heat flux, and (3) the flow direction in the bifurcating branches. As each of the parameters is varied, the heat-driven acoustics signature is found to change. The main nonlinearity is identified in the heat fluxes. Comparing the numerical and experimental results reveals that good agreement is obtained in terms of mode frequencies, mode shapes, sound pressure level and supercritical Hopf bifurcating behavior.

2:45

4pPA6. Application of Mach-Zehnder interferometer to measure irregular reflection of a spherically divergent N-wave from a plane surface in air. Maria M. Karzova (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Université Lyon I, Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Petr V. Yuldashev (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Sebastien Ollivier (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Université Lyon I, Lyon, France), Vera A. Khokhlova (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Université Lyon I, Lyon, France)

Mach stem is a well-known structure typically observed in the process of strong (acoustical Mach numbers greater than 0.4) step-shock waves reflection from a rigid boundary. However, this phenomenon has been much less studied for weak shocks in nonlinear acoustic fields where Mach numbers are in the range from 0.001 to 0.01 and pressure waveforms have more complicated temporal structure than step shocks. In this work, the results are reported for Mach stem formation observed in the experiment of N-wave reflections from a plane surface. Spherically divergent N-waves were generated by a spark source in air and were measured using a Mach-Zehnder interferometer. Pressure waveforms were reconstructed using the inverse Abel transform applied to the phase of the interferometer measurement signal. Temporal resolution of 0.4 μs was achieved. Regular and irregular types of reflection were observed. It was shown that the length of the Mach stem increased linearly while the N-wave propagated along the surface. In addition, preliminary results of the influence of surface roughness on the Mach stem formation will be presented. [Work supported by the President of Russia MK-5895.2013.2 grant, student stipend from the French Government, and by LabEx CeLyA ANR-10-LABX-60/ANR-11-IDEX-0007.]

3:00–3:15 Break

3:15

4pPA7. Statistical inversion approach to estimating water content in an aquifer from seismic data. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, timo.lahivaara@uef.fi), Nicholas F. Dudley Ward (Otago Computational Modelling Group Ltd., Dunedin, New Zealand), Tomi Huttunen (Kuava Ltd, Kuopio, Finland), and Jari P. Kaipio (Mathematics, Univ. of Auckland, Auckland, New Zealand)

This study focuses on developing computational tools to estimate water content in an aquifer from seismic measurements. The poroelastic signature from an aquifer is simulated and methods that use this signature to estimate the water table level and aquifer thickness are investigated. In this work, the spectral-element method is used to solve the forward model that characterizes the propagation of seismic waves. The inverse problem is formulated in the Bayesian framework, so that all uncertainties are explicitly modelled as probability distributions, and the solution is given as summary statistics over the posterior distribution of parameters relative to data. For the inverse problem, we use the Bayesian approximation error method which reduces the overall computational demand. In this study, results in the two-dimensional case with simulated data are presented.

3:30

4pPA8. Surfactant-free emulsification in microfluidics using strongly oscillating bubbles. Siew-Wan Ohl, Tandiono Tandiono, Evert Klaseboer (Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Dave Ow, Andre Choo (Bioprocessing Technol. Inst., Singapore, Singapore), Fenfang Li, and Claus-Dieter Ohl (Division of Phys. and Appl. Phys., School of Physical and Mathematical Sci., Nanyang Technol. Univ., Singapore, Singapore)

In this study, two immiscible liquids in a microfluidics channel has been successfully emulsified by acoustic cavitation bubbles. These bubbles are generated by the attached piezo transducers which are driven to oscillate at resonant frequency of the system (about 100 kHz) [1, 2]. The bubbles oscillate and induce strong mixing in the microchamber. They induce the rupture of the liquid thin layer along the bubble surface due to the high shear stress and fast liquid jetting at the interface. Also, they cause the big droplets to fragment into small droplets. Both water-in-oil and oil-in-water emulsions with viscosity ratio up to 1000 have been produced using this method without the application of surfactant. The system is highly efficient as submicron monodisperse emulsions (especially for water-in-oil emulsion) could be created within milliseconds. It is found that with a longer ultrasound exposure, the size of the droplets in the emulsions decreases, and the uniformity of the emulsion increases. Reference: [1] Tandiono, SW Ohl *et al.*, "Creation of cavitation activity in a microfluidics device through acoustically driven capillary waves," Lab Chip 10, 1848–1855 (2010). [2] Tandiono, SW Ohl *et al.*, "Sonochemistry and sonoluminescence in microfluidics," Proc. Natl. Acad. Sci. U.S.A. 108(15), 5996–5998 (2011).

3:45

4pPA9. Ultrasonic scattering from poroelastic materials using a mixed displacement-pressure formulation. Max denis (Mayo Clinic, 200 First St. SW, Rochester, MN 55905, denis.max@mayo.edu), Chrisna Nguon, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MD)

In this work, a numerical technique suitable for evaluating the ultrasonic scattering from a three-dimensional poroelastic material is presented. Following Biot's derivation of the macroscopic governing equations for a fluid saturated poroelastic material, the predicted two propagating wave equations are formulated in terms of displacement and pressure. Assuming that porosity variations on a microscopic scale have a cumulative effect in generating a scattered field, the scattering attenuation coefficient of a Biot medium can be determined. The scattered fields of the wave equations are numerically evaluated as Neumann series solutions of the Kirchhoff-Helmholtz integral equation. A Padé approximant technique is employed to extrapolate beyond the Neumann series' radius of convergence (weak scattering regime). In the case of bovine trabecular bone, the relationship between the scattering attenuation coefficient and the structural and mechanical properties of the trabecular bone is of particular interest. The results demonstrate the validity of the linear frequency-dependent assumption of attenuation coefficient n the low frequency range. Further comparisons, between measured observations and the numerical results will be discussed.

4:00

4pPA10. High temperature resonant ultrasound spectroscopy study on Lead Magnesium Niobate—Lead Titanate (PMN-PT) relaxor ferroelectric material. Sumudu P. Tennakoon and Joseph R. Gladden (Phys. and Astronomy, Univ. of MS, 1 Coliseum Dr., Phys.& NCPA, Univ. of MS, University, MS 38677, sptennak@go.olemiss.edu)

Lead magnesium niobate-lead titanate [(1-x)$PbMg_{1/3}Nb_{2/3}O_3$-x$PbTiO_3$] is a perovskite relaxor ferroelectric material exhibiting superior electromechanical coupling compared to the conventional piezoelectric materials. In this work, non-poled single crystal PMN-PT material with the composition near morphotropic phase boundary (MPB) was investigated in the temperature range of 400 K—800 K where the material is reported to be in the cubic phase. High temperature resonant ultrasound spectroscopy (HT-RUS) technique was used to probe temperature dependency of elastic constants derived from the measured resonant modes. Non-monotonic resonant frequency trends in the temperature regime indicate stiffening of the material, followed by gradual softening typically observed in heated materials. Elastic constants confirmed this stiffening in the temperature range of 400 K—673 K, where the stiffness constants C_{11} and C_{44} increased approximately by 40% and 33% respectively. Acoustic attenuation, derived from the quality factor (Q), exhibits a minimum around the temperature where the stiffness is maximum and, significantly higher attenuation observed at temperatures below 400 K. The temperature range 395 K—405 K was identified as a transition temperature range, where the material showed an abrupt change in the resonant spectrum and, the material emerges from the MPB characterized by this very high acoustic attenuation. This transition temperature compares favorably with dielectric constant measurements reported in the literature.

4:15

4pPA11. Structure of cavitation zones in a heavy magma under explosive character of its decompression. Valeriy Kedrinskiy (Physical HydroDynam.., Lavrentyev Inst. of HydroDynam.., Russian Acad. of Sci., Lavrentyev prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru)

The paper is devoted to the investigation of a dynamics of state and structure of compressed magma flow saturated by gas and microcrystallites which is characterized by phase transitions, diffusive processes, by increase of a magma viscosity magnitude by the orders and bubbly cavitation development behind the decompression wave front formed in the result of volcanic channel depressurization. The multi-phase mathematical model, which includes well-known conservation laws for mean pressure, mass velocity, and density as well as the system of the kinetic equations describing the physical processes that occur in a compressed magma during its explosive decompression, is considered. The results of numerical analysis show that the processes of a magma saturation by cavitation nuclei as their density magnitude increases by a few orders lead to the formation of separate zone with anomalously high values of the flow parameters. As it has turned out the abnormal zone is located in the vicinity of a free surface of a cavitating magma column. The mechanism of its formation is determined by diffusion flows redistribution as the nuclei density increases as well as by the change of the distribution character of main flow parameters in the abnormal zone from a gradual to an abrupt increase of their values on the lower zone bound. Note, the mass velocity jump by the order magnitude relatively main flow allows to conclude that the flow disintegration on the lower bound of the zone is quite probable. [Supp. RAS Presidium Program, Project 2.6].

4:30

4pPA12. Cavity collapse in a bubbly liquid. Ekaterina S. Bolshakova (Phys., Novosibirsk State Univ., Novosibirsk, Russian Federation) and Valeriy Kedrinskiy (Physical HydroDynam., Lavrentyev Inst. of HydroDynam., Russian Acad. of Sci., Lavrentyev prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru)

The effect of an ambient liquid state on a spherical cavity dynamics under atmospheric hydrostatic pressure p and extremely low initial gas pressure p(0) inside was investigated. The equilibrium bubbly (cavitating) medium with sound velocity C as the function of gas phase concentration k was considered as the ambient liquid model. The cavity dynamics is analyzed within the framework of Herring-equation for the following diapasons of main parameters : $k = 0$—5%, $p(0) = 0.02$—$10(-6)$ atm. Numerical analysis has shown that the deceleration C by two order does not have an influence neither on an asymptotic value of collapsed cavity radius nor on the acoustical losses under its collapse. It means than in the whole the integral acoustical losses remain invariable. However the collapse cavity dynamics and the radiation structure are essentially changed: from numerous pulsations with a decreasing amplitudes up to a single collapse and from a wave packet up to a single wave, correspondingly. It has turned out that the acoustic corrections in the Herring-equation don't influence practically on the cavity dynamics if the term of equation with dH/dt is absent. Naturally, the deceleration C exerts essential influence on an empty cavity dynamics. The graphs of dR/Cdt values as a function of R/Ro for different C values are located higher the data of classical models of Herring, Gilmore and Hunter. So the value $M = 1$ is reached at $R/Ro = 0.023$ for $k = 0$, and at the value 0.23 when $k = 5$ %. [Support RFBR, grant 12-01-00314.]

Session 4pPP

Psychological and Physiological Acoustics: Physiological and Psychological Aspects of Central Auditory Processing Dysfunction II

Frederick J. Gallun, Cochair
National Center for Rehabilitative Auditory Research, Portland VA Medical Center, 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Adrian KC Lee, Cochair
University of Washington, Box 357988, University of Washington, Seattle, WA 98195

Invited Papers

1:30

4pPP1. Aging as a window into central auditory dysfunction: Combining behavioral and electrophysiological approaches. David A. Eddins, Erol J. Ozmeral, and Ann C. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu)

Central auditory processing involves a complex set of feed-forward and feedback processes governed by a cascade of dynamic neuro-chemical mechanisms. Central auditory dysfunction can arise from disruption of one or more of these processes. With hearing loss, dysfunction may begin with reduced and/or altered input to the central auditory system followed by peripherally induced central plasticity. Similar central changes may occur with advancing age and neurological disorders even in the absence of hearing loss. Understanding the behavioral and physiological consequences of this plasticity on the processing of basic acoustic features is critical for effective clinical management. Major central auditory processing deficits include reduced temporal processing, impaired binaural hearing, and altered coding of spectro-temporal features. These basic deficits are thought to be primary contributing factors to the common complaint of difficulty understanding speech in noisy environments in persons with hearing loss, brain injury, and advanced age. The results of investigations of temporal, spectral, and spectro-temporal processing, binaural hearing, and loudness perception will be presented with a focus on central auditory deficits that occur with advancing age and hearing loss. Such deficits can be tied to reduced peripheral input, altered central coding, and complex changes in cortical representations.

2:00

4pPP2. Age-related declines in hemispheric asymmetry as revealed in the binaural interaction component. Ann C. Eddins, Erol J. Ozmeral, and David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, aeddins@usf.edu)

The binaural interaction component (BIC) is a physiological index of binaural processing. The BIC is defined as the brain activity resulting from binaural (diotic or dichotic) stimulus presentation minus the brain activity summed across successive monaural stimulus presentations. Smaller binaural-induced activity relative to summed monaural activity is thought to reflect neural inhibition in the central auditory pathway. Since aging is commonly associated with reduced inhibitory processes, we evaluate the hypothesis that the BIC is reduced with increasing age. Furthermore, older listeners typically have reduced hemispheric asymmetry relative to younger listeners, interpreted in terms of compensation or recruitment of neural resources and considered an indication of age-related neural plasticity. Binaural stimuli designed to elicit a lateralized percept generate maximum neural activity in the hemisphere opposite the lateralized position. In this investigation, we evaluated the hypothesis that the BIC resulting from stimuli lateralized to one side (due to interaural time differences) results in less hemispheric asymmetry in older than younger listeners with normal hearing. Behavioral data were obtained to assess the acuity of binaural processing. Data support the interpretation that aging is marked by reduced central auditory inhibition, reduced temporal processing, and broader distribution of activity across hemispheres compared to young adults.

2:30

4pPP3. Effects of blast exposure on central auditory processing. Melissa Papesh, Frederick Gallun, Robert Folmer, Michele Hutter, M. Samantha Lewis, Heather Belding (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 6303 SW 60th Ave., Portland, OR 97221, Melissa.Papesh@va.gov), and Marjorie Leek (Res., Loma Linda VA Medical Ctr., Loma Linda, CA)

Exposure to high-intensity blasts is the most common cause of injury in recent U.S. military conflicts. Prior work indicates that blast-exposed Veterans report significantly more hearing handicap than non-blast-exposed Veterans, often in spite of clinically normal hearing thresholds. Our research explores the auditory effects of blast exposure using a combination of self-report, behavioral, and electrophysiological measures of auditory processing. Results of these studies clearly indicate that blast-exposed individuals are significantly more likely to perform poorly on tests requiring the use of binaural information and tests of pitch sequencing and temporal acuity

compared to non-blast-exposed control subjects. Behavioral measures are corroborated by numerous objective electrophysiological measures, and are not necessarily attributable to peripheral hearing loss or impaired cognition. Thus, evidence indicates that blast exposure can lead to acquired deficits in central auditory processing (CAP) which may persist for at least 10 years following blast exposure. Future studies of these deficits in this and other adult populations are needed to address important issues such as individual susceptibility, anatomical, and physiological changes in auditory pathways which contribute to symptoms of these types of deficits, and development of effective evidence-based methods of rehabilitation in adult patients. [Work supported by the VA Rehabilitation Research & Development Service and the VA Office of Academic Affiliations.]

3:00–3:30 Break

3:30

4pPP4. Auditory processing demands and working memory span. Margaret K. Pichora-Fuller (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca) and Sherri L. Smith (Audiologic Rehabilitation Lab., Veterans Affairs, Mountain Home, TN)

The (in)dependence of auditory and cognitive processing abilities is a controversial topic for hearing researchers and clinicians. Some advocate for the need to isolate auditory and cognitive factors. In contrast, we argue for the need to understand how they interact. Working memory span (WMS) is a cognitive measure that has been related to language comprehension in general and also to speech understanding in noise. In healthy adults with normal hearing, there is typically a strong correlation between reading and listening measures of WMS. Some investigators have opted to use visually presented stimuli when testing people who do not have normal hearing in order to avoid the influence of modality-specific auditory processing deficits on WMS. However, tests conducted using auditory stimuli are necessary to evaluate how cognitive processing is affected by the auditory processing demands experienced by different individuals over a range of conditions in which the tasks to be performed, the availability of supportive context, and the acoustical and linguistic characteristics of targets and maskers are varied. Attempts to measure auditory processing independent of cognitive processing will fall short in assessing listening function in realistic conditions.

4:00

4pPP5. Auditory perceptual learning as a gateway to rehabilitation. Beverly A. Wright (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu)

A crucial aspect of the central nervous system is that it can be modified through experience. Such changes are thought to occur in two learning phases: acquisition—the actual period of training—and consolidation—a post-training period during which the acquired information is transferred to long-term memory. My coworkers and I have been addressing these principles in auditory perceptual learning by characterizing the factors that induce and those that prevent learning during the acquisition and consolidation phases. We also have been examining how these factors change during development and aging and are affected by hearing loss and other conditions that alter auditory perception. Application of these principles could improve clinical training strategies. Further, though learning is the foundation for behavioral rehabilitation, the capacity to learn can itself be impaired. Therefore, an individual's response to perceptual training could be used as an objective, clinical measure to guide diagnosis and treatment of a cognitive disorder. [Work supported by NIH.]

4:30–5:00 Panel Discussion

Session 4pSC

Speech Communication: Voice (Poster Session)

Richard J. Morris, Chair
Communication Science and Disorders, Florida State University, 201 West Bloxham Road, 612 Warren Building, Tallahassee, FL 32306-1200

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors an opportunity to see other posters, the contributors of odd-numbered papers will be at their posters from 1:00 p.m. and 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

4pSC1. Acoustical bases for the perception of simulated laryngeal vocal tremor. Rosemary A. Lester, Brad H. Story, and Andrew J. Lotto (Speech, Lang., and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721, rosemary.lester@gmail.com)

Vocal tremor involves atypical modulation of the fundamental frequency (F0) and intensity of the voice. Previous research on vocal tremor has focused on measuring the modulation rate and extent of the F0 and intensity without characterizing other modulations present in the acoustic signal (i.e., modulation of the harmonics). Characteristics of the voice source and vocal tract filter are known to affect the amplitude of the harmonics and could potentially be manipulated to reduce the perception of vocal tremor. The purpose of this study was to determine the adjustments that could be made to the voice source or vocal tract filter to alter the acoustic output and reduce the perception of modulation. This research was carried out using a computational model of speech production that allows for precise control and modulation of the glottal and vocal tract configurations. Results revealed that listeners perceived a higher magnitude of voice modulation when simulated samples had a higher mean F0, greater degree of vocal fold adduction, and vocal tract shape for /i/ vs. /ɑ/. Based on regression analyses, listeners' judgments were predicted by modulation information present in both low and high frequency bands. [Work supported by NIH F31-DC012697.]

4pSC2. Perception of breathiness in pediatric speakers. Lisa M. Kopf, Rahul Shrivastav (Communicative Sci. and Disord., Michigan State Univ., Rm. 109, Oyer Speech and Hearing Bldg., 1026 Red Cedar Rd., East Lansing, MI 48824, kopflisa@msu.edu), David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), and Mark D. Skowronski (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Extensive research has been done to determine acoustic metrics for voice quality. However, few studies have focused on voice quality in the pediatric population. Instead, metrics evaluated on adults have directly been applied to children's voices. Some variables, such as pitch, that differ between adult and pediatric voices, have been shown to be critical in the perception of breathiness. Furthermore, it is not known whether adults perceive voice quality similarly for pediatric and adult speakers. In this experiment, 10 listeners judged breathiness for 28 stimuli using a single-variable matching task. The stimuli were modeled after four pediatric speakers and synthesized using a Klatt-synthesizer to have a wide range of aspiration noise and open quotient. Both of these variables have been shown to influence the perception of breathiness. The resulting data were compared to that previously obtained for adult speakers using the same matching task. Comparison of adult and pediatric voices will help identify differences in the perception of breathiness for these groups of speakers and to develop more accurate metrics for voice quality in children. [Research supported by NIH (R01 DC009029).]

1:00

4pSC3. Combining differentiated electroglottograph and differentiated audio signals to reliably measure vocal fold closed quotient. Richard J. Morris (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu), Shonda Bernadin (Elec. and Comput. Eng., Florida A & M Univ., Tallahassee, FL), David Okerlund (College of Music, Florida State Univ., Tallahassee, FL), and Lindsay B. Wright (Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

Over the past few decades researchers have explored the use of the electroglottograph (EGG) as a non-invasive method for representing vocal fold contact during vowel production and to measure the closed quotient (CQ) and open quotient (OQ) of the glottal cycle. The first derivative of the EGG signal (dEGG) can be used to indicate these moments (Childers & Krishnamurthy, 1985). However, there can be double positive peaks in the dEGG as well as a variety of negative peak patterns (Herbst *et al*., 2010). Obviously these variations will alter any measurements made from the signal. Recently, the use of the dEGG with dAudio signal was reported as a means for more reliable measurement of the CQ from the EGG signal in combination with a time synchronized audio signal. The purpose of this study is to demonstrate the reliability of the dEGG and dAudio for determining CQ across a variety of vocal conditions. Files recorded from group of 15 trained females singing an octave that included their primo passaggio provided the data. Preliminary results indicate high reliability of the CQ measurements in both the chest and middle registers of all of the singers.

4pSC4. A reduced-order three-dimensional continuum model of voice production. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Although vocal fold vibration largely occurs in the transverse plane, control of voice is mainly achieved by adjusting vocal fold stiffness along the anterior–posterior direction through muscle activation. Thus, models of voice control need to be at least three-dimensional on the structural side. Modeling the detailed three-dimensional interaction between vocal fold dynamics, glottal aerodynamics, and the sub- and supra-glottal acoustics is computationally expensive, which prevents parametric studies of voice production using three-dimensional models. In this study, a Galerkin-based reduced-order three-dimensional continuum model of phonation was presented. Preliminary results showed that this model was able to qualitatively reproduce previous experimental observations. This model is computationally efficient and thus ideal for parametric studies in phonation research as well as practical applications such as speech synthesis. [Work supported by NIH.]

4pSC5. The influence of attentional focus on voice control. Eesha A. Zaher and Charles R. Larson (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, EeshaZaheer2014@u.northwestern.edu)

The present study tested the role of attentional focus on control of voice fundamental frequency (F0). Subjects vocalized an "ah" sound while hearing their voice auditory feedback randomly shifted upwards or downwards in pitch. In the "UP" condition, subjects vocalized, listened for and pressed a button for each upward pitch shift stimulus. In the "DOWN" condition, subjects listened for and pressed a button for each downward shift. In the CONTROL condition, subjects vocalized without paying attention to the stimulus direction or pressing a button. Data were analyzed by averaging voice F0 contours across several trials for each pitch shift stimulus in all conditions. Response magnitudes were larger for the CONTROL than for the UP or DOWN conditions. Responses for the UP and DOWN conditions did not differ. Results suggest that when subjects focus their attention to identify specific stimuli and produce a non-vocal motor response conditional upon the identification, the neural mechanisms involved in voice control are reduced, possibly because of a reduction in the error signal resulting from the comparison of the efference copy of voice output with auditory feedback. Thus, focusing attention away from vocal control reduces neural resources involved in control of voice F0.

4pSC6. Attention-related modulation of involuntary audio-vocal response to pitch feedback errors. Hanjun Liu, Huijing Hu, and Ying Liu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

It has been demonstrated that unexpected alterations in auditory feedback elicit fast compensatory adjustments in vocal production. Although generally thought to be involuntary in nature, whether these adjustments can be influenced by cognitive function such as attention remains unknown. The present event-related potential (ERP) study investigated whether neurobehavioral processing of auditory-vocal integration can be affected by attention. While sustaining a vowel phonation and hearing pitch-shifted feedback, participants were required to either ignore the auditory feedback perturbation, or attend to it with two levels of attention load. The results revealed enhancement of P2 response to the attended auditory perturbation with the low load level as compared to the unattended auditory perturbation. Moreover, increased auditory attention load led to a significant decrease of P2 response. By contrast, there was no attention-related change of vocal response. These findings provide the first neurophysiological evidence that involuntary auditory-vocal integration can be modulated as a function of auditory attention. Furthermore, it is suggested that auditory attention load can result in a decrease of the cortical processing of auditory-vocal integration in pitch regulation.

4pSC7. A study on the effect of intraglottal vortical structures on vocal fold vibration. Mehrdad H Farahani and Zhaoyan Zhang (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, mh.farahani@gmail.com)

Recent investigations suggested possible formation of the vortical structures in the intraglottal region during the closing phase of the phonation cycle. Vortical regions in the flow field are locations of negative pressure, and it has been hypothesized that this negative pressure might facilitate the glottal closure and thus affects the vibration pattern and voice production for high subglottal pressures. However, it is unclear whether the vortex-induced negative pressure is large enough, compared with vocal fold inertia and elastic recoil, to have a noticeable effect on glottal closure. In addition, the intraglottal vortical structures generally exist only for a small fraction of the closing phase when the glottis becomes divergent enough to induce flow separation. In the current work, oscillation of the vocal folds and the flow field are modeled using a non-linear finite element solver and a reduced order flow solver, respectively. The effect of vortical structures is modeled as a sinusoidal negative pressure wave applied to vocal fold surface between the flow separation point and the superior edge of the vocal folds. The effects of this vortex-induced negative pressure are quantified at different conditions of vocal fold stiffness and subglottal pressures. [Work supported by NIH.]

4pSC8. Effects of thyroarytenoid muscle activation on phonation in an *in vivo* canine larynx model. Georg Luegmair, Dinesh Chhetri, and Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California Los Angeles, 1000 Veteran Ave., Rehab 31-24, Los Angeles, CA 90095, gluegmair@ucla.edu)

Previous studies have shown that the thyroarytenoid (TA) muscle plays an important role in the control of vocal fold adduction and stiffness. The effects of TA stimulation on vocal fold vibration, however, are still unclear. In this study, the effects of TA muscle activation on phonation were investigated in an *in vivo* canine larynx model. Laryngeal muscle activation was achieved through parametric stimulation of the thyroarytenoid, the lateral cricoarytenoid (LCA), and the cricothyroid (CT) muscles. For each stimulation level, the subglottal pressure was gradually increased to produce phonation. The subglottal pressure, the volume flow, and the outside acoustic pressure were measured together with high-speed recording of vocal fold vibration from a superior view. The results show that, without TA activation, phonation was limited to conditions of medium to high levels of LCA and CT activations. TA activation allowed phonation to occur at a much lower activation level of the LCA and CT muscles. Compared to conditions of no TA activation, TA activation led to decreased open quotient. Increasing TA activation also allow phonation to occur at a much larger range of the subglottal pressure while still maintaining certain degree of glottal closure during vibration. [Work supported by NIH.]

4pSC9. Voice accumulation and voice disorders in primary school teachers. Pasquale Bottalico (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 10125, pasqualebottalico@yahoo.it), Lorenzo Pavese, Arianna Astolfi (Dipartimento di Energia, Politecnico di Torino, Torino, Italy), and Eric J. Hunter (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Statistics on professional voice users with vocal health issues demonstrate the significance of the problem. However, such disorders are not currently recognized as an occupational disease in Italy. Conducting studies examining the vocal health of occupational voice users is an important step in identifying this as an important public health issue. The current study was conducted in six primary schools in Italy with 25 teachers, one of the most affected occupational categories. A clinical examination was conducted (consisting of hearing and voice screening, a VHI, etc.). On this basis, teachers were divided into three groups: healthy subjects, subject with logopaedic disorders, and subjects with objectively measured pathological symptoms. The distributions of voicing and silence periods for the teachers at work were collected using the Ambulatory Phonation Monitor (APM3200), a device for long-term monitoring of vocal parameters. The APM senses the vocal fold vibrations at the base of the neck by means of a small accelerometer. Correlations were calculated between the voice accumulation slope (obtained by multiplying the number of occurrences for each period by the corresponding duration) and the clinical status of the teachers. The differences in voice accumulation distributions among the three groups were analyzed.

4pSC10. Room acoustics and vocal comfort in untrained vocalists. Eric J. Hunter, Pasquale Bottalico, Simone Graetzer, and Russell Banks (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu)

Talkers have long been studied in their speech accommodation strategies in noise. Vocal effort and comfort within noisy situations have also been studied. In this study, untrained vocalists were exposed to a range of room acoustic conditions. In each environment, the subject performed a vocal task, with a goal of being "heard" by a listener 5 m away. After each task, the subject completed a series of questions addressing vocal effort and comfort. Additionally, using a head and torso simulator (HATS), the environment was assessed using a sine sweep presented at the HATS mouth and recorded at the ears. It was found that vocal clarity (C50) and the initial reflection related to vocal comfort. The results are not only relevant to room design but also to understanding talkers' acuity to acoustic conditions and their adjustments to them.

4pSC11. Flow vibrato in singers. Srihimaja Nandamudi and Ronald C. Scherer (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health and Human Services Bldg., Bowling Green, OH 43403, nandas@bgsu.edu)

Frequency (F0) vibrato is commonly known, but not so for flow vibrato, the mean flow variation that accompanies frequency vibrato. Two classically trained singers, each with over 20 years professional experience, a soprano and a tenor, recorded /pa:pa:pa:/ sequences on three pitches (C4, A4, and G5 for the soprano, D3, D4, and G4 for the tenor) and three loudness levels (p, mf, and f) at each pitch. Each vowel had 3–6 frequency vibrato cycles. For both singers, flow vibrato (obtained using the Glottal Enterprises aerodynamic system) was present, and the lowest pitch had the most variability; otherwise, flow vibrato was fairly sinusoidal in shape. For the soprano, flow vibrato cycle extents were: 21–88 cc/s, lowest pitch; 60–147 cc/s, middle pitch; 115–214 cc/s, highest pitch, across loudness levels. For the soprano, the phase difference for the flow was 120–180 degrees ahead of the F0 vibrato. For the tenor, the flow vibrato cycle extents were: 32–85 cc/s, lowest pitch; 98–113 cc/s, middle pitch; 76–240 cc/s, highest pitch, across loudness levels. Flow vibrato for the tenor led the F0 vibrato typically by 40–120 degrees. For both subjects, some flow vibrato cycles had double peaks. Flow vibrato needs further study to determine its origin, shapes, and magnitudes.

4pSC12. Impact of vocal tract resonance on the perception of voice quality changes caused by vocal fold stiffness. Rosario Signorello, Zhaoyan Zhang, Bruce Gerratt, and Jody Kreiman (Head and Neck Surgery, Univ. of California Los Angeles David Geffen School of Medicine, 31-24 Rehab Ctr., UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, rsignorello@ucla.edu)

Experiments using animal and human larynx models are often conducted without a vocal tract. While it is reasonable to assume the absence of a vocal tract has only small effects on vocal fold vibration, it is unclear how sound production and its perception will be affected. In this study, the validity of using data obtained in the absence of a vocal tract for voice perception studies was investigated. Using a two-layer self-oscillating physical model, three series of voice stimuli were created: one produced with conditions of left-right symmetric vocal fold stiffness, and two with left-right asymmetries in vocal fold body stiffness. Each series included a set of stimuli created with a physical vocal tract, and a second set created without a physical vocal tract. Stimuli were re-synthesized to equalize the mean F0 for each series and normalized for amplitude. Listeners were asked to evaluate the three series in a sort-and-rate task. Multidimensional scaling analysis will be applied to examine the perceptual interaction between the voice source and the vocal tract resonances. [Work supported by NIH.]

4pSC13. Perceptual differences among models of the voice source: Further evidence. Marc Garellek (Linguist, UCSD, La Jolla, CA), Gang Chen (Elec. Eng., UCLA, Los Angeles, CA), Bruce R. Gerratt (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403), Abeer Alwan (Elec. Eng., UCLA, Los Angeles, CA), and Jody Kreiman (Head and Neck Surgery, UCLA, Los Angeles, CA, jkreiman@ucla.edu)

Models of the voice source differ in how they fit natural voices, but it is still unclear which differences in fit are perceptually salient. This study describes ongoing analyses of differences in the fit of six voice source models to 40 natural voices, and how these differences relate to perceptual similarities among stimuli. Listeners completed a visual sort-and-rate task to compare versions of each voice created with the different source models, and the results were analyzed using multi-dimensional scaling (MDS). Perceptual spaces were interpreted in terms of variations in model fit in both the time and spectral domains. The discussion will focus on the perceptual importance of matches to both time-domain and spectral features of the voice. [Work supported by NIH/NIDCD grant DC01797 and NSF grant IIS-1018863.]

4pSC14. The biological function of fundamental frequency in leaders' charismatic voices. Rosario Signorello (Head and Neck Surgery, Univ. of California Los Angeles David Geffen School of Medicine, 31-24 Rehab Ctr., UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, rsignorello@ucla.edu)

Charismatic leaders use voice based on two functions: a primary biological function and a secondary language and culture-based function (Signorello, 2014). In order to study the primary function in more depth, we conducted acoustic and perceptual studies on the use of F0 by French, Italian and Brazilian charismatic political leaders. Results show that leaders manipulate F0 in significantly different manners relative to: (1) the context of communication (persuasive goal, the place where communication occurs and the type of audience) in order to be recognized as the leader of the group; and (2) the elapse of time (from the beginning to the end of the speech) in order to create a climax with the audience. Results of a perceptual test show that the leader's use of low F0 voice results in the perception of the leader as a dominant or threatening leader and the use of higher F0 conveys sincere, calm, and reassuring leadership. These results show cross-language and cross-cultural similarities in leaders' vocal behavior and listeners' perception, and robustly demonstrate the two different functions of leaders' voices.

4pSC15. Voice quality variation and gender. Kara Becker, Sameer ud Dowla Khan, and Lal Zimman (Linguist, Reed College, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, kbecker@reed.edu)

Recent work on American English has established that speakers increasingly use creaky phonation to convey pragmatic information, with young urban women assumed to be the most active users of this phonetic feature. However, no large-scale acoustic or articulatory study has established the actual range and diversity of voice quality variation along gender identities, encompassing different sexual orientations, regional backgrounds, and socioeconomic statuses. The current study does exactly that, through four methods: (1) subjects identifying with a range of gender and other demographic identities were audio recorded while reading wordlists as well as a scripted narrative assuming characters' voices designed to elicit variation in vowel quality. Simultaneously, (2) electroglottographic readings were taken and analyzed to determine the glottal characteristics of this voice quality variation. (3) Subjects were then asked to rate recordings of other people's voices to identify the personal characteristics associated with the acoustic reflexes of phonation; in the final task, (4) subjects were explicitly asked about their language ideologies as they relate to gender. Thus, the current study explores the relation between gender identity and phonetic features, measured acoustically, articulatorily, and perceptually. This work is currently underway and preliminary results are being compiled at this time.

4pSC16. Towards standard scales for dysphonic voice quality: Magnitude estimation of reference stimuli. David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu) and Rahul Shrivastav (Communicative Sci. & Disord., Michigan State Univ., East Lansing, MI)

This work represents a critical step in developing standard measurement scales for the dysphonic voice qualities of breathiness and roughness. Methods such as Likert ratings, visual analog scales and magnitude estimation result in arbitrary units, limiting their clinical usefulness. A single-variable matching task can quantifying voice quality in terms of physical units but is too time consuming for clinical use. None of these methods result in information that has a direct or intuitive relationship with the underlying percept. A proven approach for the perception of loudness is the Sone scale which ties physical units to the perceptual estimates of loudness magnitude. As a first step in developing such a scale for breathiness and roughness, here we establish the relationship between the change in perceived VQ magnitude and the change in physical units along the continuum of each VQ dimension. A group of 25 listeners engaged in a magnitude estimation task to determine perceived magnitude associated with the comparison stimuli used in our single-variable matching tasks. This relationship is analogous to mapping intensity in dB to perceived loudness in Phons and is a critical step in developing a Sone-like scale for breathiness and roughness.

4pSC17. Divergent or convergent glottal angles: Which give greater flow? Ronald Scherer (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health Ctr., Bowling Green, OH 43403, ronalds@bgsu.edu)

During phonation, the glottis alters between convergent and divergent angles. For the same angle value, diameter, and transglottal pressure, which angle, divergent or convergent, results in greater flow? The symmetric glottal angles of the physical static model M5 were used. Characteristics (life-size) of the model were: axial glottal length 0.30 cm; angles of 5, 10, 20, and 40 degrees; diameters of 0.005, 0.01, 0.02, 0.04, 0.08, 0.16, and 0.32 cm; transglottal pressures from 1 to 25 cm H2O; resulting in flows from 2.7 to 1536 cc/s and Reynolds number from 29.4 to 13,058. Results: (1) For diameters of 0.04, 0.08 and 0.16 cm, the divergent angle always gave more flow than the convergent angle (about 5–25%); (2) for the smallest (0.005 cm) and largest diameter (0.32 cm), the divergent angles always gave less flow (10–30%); (3) for diameters of 0.01 and 0.02 cm, flow was greater for divergent 5 and 10 degrees, and less for divergent 20 and 40 degrees. These results suggest that the divergent glottal angle will increase the glottal flow for midrange glottal diameters (skewing the glottal flow further "to the right"?), and create less flow at very small diameters (increasing energy in the higher harmonics?).

4pSC18. Methodological issues when estimating subglottal pressure from oral pressure. Brittany Frazer (Commun. Sci. and Disord., Bowling Green State Univ., 200 Marie Pl., Perrysburg, OH 43551, bfrazer@bgsu.edu) and Ronald C. Scherer (Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

A noninvasive method to estimate subglottal pressure for vowel productions is to smoothly utter a CVCV string such as /p:i:p:i:p:i:…/ using a short tube in the mouth with the tube attached to a pressure transducer. The pressure during the lip occlusion estimates the subglottal pressure during the adjacent vowel. What should the oral pressure look like for it to provide accurate estimates? The study compared results using various conditions against a standard condition that required participants to produce /p:i:p:i:../ syllables smoothly and evenly at approximately 1.5 syllables per second. The non-standard tasks were: performing the task without training, increasing syllable rate, using a voiced /b/ instead of a voiceless /p/ initial syllable, adding a lip or velar leak, or using a two syllable production ("peeper") instead of a single syllable production. Lip leak, velar leak, and lack of time to equilibrate air pressure throughout the airway caused estimates of subglottal pressure to be inaccurate. Accuracy was better when estimates of subglottal pressure were obtained using the voiced initial consonant and the two-syllable word. Training improved the consistency of the oral pressure profiles and thus the assurance in estimating the subglottal pressure. Oral pressures with flat plateaus appear most accurate.

THURSDAY AFTERNOON, 30 OCTOBER 2014 — INDIANA F, 1:00 P.M. TO 4:45 P.M.

Session 4pUW

Underwater Acoustics: Shallow Water Reverberation III

Kevin L. Williams, Chair
Applied Physics Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105

Contributed Paper

1:00

4pUW1. Seafloor sound-speed profile and interface dip angle measurement by the image source method: Application to real data. Samuel Pinson (Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, LVA Dept de Engenharia Mecânica, UFSC, Bairro Trindade, Florianópolis, SC 88040-900, Brazil, samuelpinson@yahoo.fr) and Charles W. Holland (Penn State Univ., State College, PA)

The image source method characterizes the sediment sound-speed profile from seafloor reflection data with a low computational cost compared with inversion techniques. Recently, the method has been extended to treat non-parallel sediment layering. The method is applied to data from an autonomous underwater vehicle (AUV) towing a source (1600–3500 Hz) and an horizontal array of hydrophones. AUV reflection measurements were acquired every 3 m along 10 criss-cross lines over a 1km<+>2<+> area with evidently dipping layers. Mapping the along track sound-speed profiles in geographical coordinates results in a pseudo-3D (Nx2D) sediment structure characterization of the area down to several tens of meters in the sub-bottom. The sound speed profile agreement at crossing points is quite good.

Invited Papers

1:15

4pUW2. Requirements, technology, and science drivers of applied reverberation modeling. Anthony I. Eller and Kevin D. Heaney (OASIS, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, ellera@oasislex.com)

The historical development of reverberation modeling is a story driven by both supporting and sometimes conflicting features of application requirements, measurement and computing capability, and scientific understanding. This paper presents an overview of how underwater reverberation modeling technology has responded to application needs and how this process has helped the community to identify and resolve related science issues. Emphasis is on the areas of System Design and Acquisition Support, Deployment and Operational Support, and Training Support. Gaps in our scientific knowledge are identified and recent advances are described that help push forward our collective understanding of how to predict and mitigate reverberation.

1:35

4pUW3. Reverberation models as an aid to interpret data and extract environmental information. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com) and John R. Preston (Appl. Res. Lab., The Penn State Univ., State College, PA)

Reverberation measurements obtained with towed arrays are a valuable tool to extract information about the ocean environment. Preston pioneered the use of polar plots to display reverberation and superimpose the beam time series on bathymetry maps. As part of Rapid Environmental Assessment (REA) exercises Ellis and Preston [J. Marine Syst. 78, S359–S371, S372–S381] have used directional reverberation measurements to detect uncharted bottom features, and to extract environmental information using model-data comparisons. One enthusiast declared "This is like doing 100 simultaneous transmission loss runs and having the results available immediately." Though that was clearly an exaggeration and the results are not precise, the approach provides valuable information to direct more accurate and detailed surveys. The early work used range-independent (flat bottom) models for the model-data comparisons, while current work includes a range-dependent model based on adiabatic normal modes. A model has been developed which calculates reverberation from range-dependent bottom bathymetry, echoes from targets and discrete clutter objects, then outputs beam time series directly comparable with measured ones. Recent work has identified interesting effects in sea bottom sand dunes in the TREX experiments. This paper will provide an overview of the earlier work, and examples from the recent TREX experiment.

1:55

4pUW4. Reverberation data/model comparisons using transport theory. Eric I. Thorsos, Jie Yang, Frank S. Henyey, and W. T. Elam (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

Transport theory has been developed for modeling shallow water propagation and reverberation at mid frequencies (1–10 kHz) where forward scattering from a rough sea surface is taken into account in a computationally efficient manner. The method is based on a decomposition of the field in terms of unperturbed modes, and forward scattering at the sea surface leads to mode coupling that is treated with perturbation theory. Reverberation measurements made during TREX13 combined with extensive environmental measurements provide an important test of transport theory predictions. Modeling indicates that the measured reverberation was dominated by bottom reverberation, and the reverberation level in the 2−4 kHz band was observed to decrease as the sea surface conditions increased from a low sea state to a higher sea state. This suggests that surface forward scattering was responsible for the change in reverberation level. Results of data/model comparisons examining this effect will be shown. [Work supported by ONR Ocean Acoustics.]

Contributed Papers

2:15

4pUW5. Physics of backscattering in terms of mode coupling applied to measured seabed roughness spectra in shallow water. David P. Knobles (ARL, UT at Austin, PO BOX 8029, Austin, TX 78713-8029, dpknobles@yahoo.com)

Energy conserving coupled integral equations for the forward and backward propagating modal components have been previously developed [J. Acoust. Soc. Am. 130, 2673–2680 (2011)]. A rough seabed surface leads to a backscattered field and modifies the interference structure of the forward propagating field. Perturbation theory applied to the basic coupled integral equations allows for physical insight into the correlation of the roughness spectrum to the forward and backward modal intensities and cross mode coherence. This study applies the Nx2D integral equation coupled-mode approach to 3-D roughness measurements and examines the physics of the coupling of the forward and backward field components and computes the modal intensities as a function of azimuth. The roughness measurements were made in about 20 m of water off Panama City, Florida. [Work supported by ONR Code 322 OA.]

2:30

4pUW6. Energy conservation via coupled modes in waveguides with an impedance boundary condition. Steven A. Stotts and Robert A. Koch (Environ. Sci. Lab., Appl. Res. Labs/The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, stotts@arlut.utexas.edu)

A statement of energy conservation for a coupled mode formulation with real mode functions and eigenvalues has been demonstrated to be consistent with the statement of conservation derivable from the Helmholtz equation. The restriction to real mode functions and eigenvalues precludes coupled mode descriptions with waveguide absorption or untrapped modes. The demonstration, along with the derivation of the coupled mode range equation, relies on orthonormality in terms of a product of two modal depth functions integrated to infinite depth. This paper shows that energy conservation and the derivation of the coupled mode range equation can be extended to complex mode functions and eigenvalues, and that energy is conserved for ocean waveguides with a penetrable bottom boundary at a finite depth beneath any range dependence. For this, the penetrable bottom boundary is specified by an impedance condition for the mode functions. The new derivations rely on completeness and a modified orthonormality statement. Mode coupling is driven solely by waveguide range dependence. Thus, the form of the range equation and the values of the coupling coefficients are unaffected by a finite depth waveguide. Applications of energy conservation to examine the accuracy of a numerical coupled mode calculation are presented.

2:45

4pUW7. Effect of channel impulse response on matched filter performance in the 2013 Target and Reverberation Experiment. Mathieu E. Colin (Acoust. and Sonar, TNO, Postbus 96864, Den Haag 2509 JG, Netherlands, mathieu.colin@tno.nl), Michael A. Ainslie (Acoust. and Sonar, TNO, The Hague, Netherlands), Peter H. Dahl, David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Sean Pecknold (Underwater Surveillance and Communications, Defence Res. and Development Canada , Dartmouth, NS, Canada), and Robbert van Vossen (Acoust. and Sonar, TNO, The Hague, Netherlands)

Active sonar performance is determined by the characteristics of the target, the sonar system and the effect of the environment on the received waveform. The two main influences of the environment are propagation effects and the contamination of the target echo with a background. The ambient noise and reverberation are mitigated by means of signal processing, mostly through beamforming and matched-filtering. The improvement can be quantified by the signal to noise ratios before and after processing. Propagation effects can have a large influence on the gains obtained by the processing. To study the effect of the channel on the matched filter performance, broadband channel impulse responses were modeled and compared to measurements acquired during the Office of Naval Research-funded 2013 Target and Reverberation Experiment (TREX). In shallow water, a large time spread is often observed, reducing the effectiveness of the matched filter. TREX data show, however, a limited time spread. Model predictions indicate that this could be caused by a rough sea-surface, which while increasing propagation loss, at the same time increases matched filter gain.

3:00–3:15 Break

3:15

4pUW8. Using physical oceanography to improve transmission loss calculations in undersampled environments. Cristina Tollefsen and Sean Pecknold (Defence Res. and Development Canada, P. O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, cristina.tollefsen@gmail.com)

The vertical sound speed profile (SSP) is a critical input to any acoustic propagation model. However, even when measured SSPs are available they are frequently noisy "snapshots" of the SSP at a single moment in time and space and do not fully capture changes such as solar heating and wind-driven mixing that can significantly affect shallow water propagation on time scales of less than a day. Furthermore, SSPs measured in the field may not extend to the ocean bottom and are often based on measured profiles of temperature with an implicit assumption of constant salinity. In April–May 2013, the Target and Reverberation Experiment (TREX) was conducted in the Northeastern Gulf of Mexico near Panama City, Florida, a region strongly affected by local wind forcing, freshwater inputs, and the presence of a warm-core Gulf of Mexico Loop Current eddy ("Eddy Kraken") offshore of the experimental site. "Synthetic" SSPs were constructed for the trial area by combining knowledge of the physical oceanography and water masses in the area with the measured SSPs that were available. Transmission loss was modelled using both synthetic and measured SSPs and the results will be compared with measured transmission loss.

3:30

4pUW9. Analytic formulation for broadband rough surface and volumetric scattering including matched-filter range resolution. Wei Huang, Delin Wang, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., 006 Hayden Hall, 370 Huntington Ave., Boston, MA 02115, huang.wei1@husky.neu.edu)

An analytic formulation is derived for the broadband scattered field from a randomly rough surface based on Green's theorem employing perturbation theory. The matched filter is applied to resolve the scattered field within the range resolution footprint of a broadband imaging system. Statistical moments of the scattered field are then expressed in terms of the second moment characterization of the scattering surface. The broadband diffuse reverberation depends on the rough surface spectrum evaluated over a range of wavenumbers, centered at the Bragg wavenumber corresponding to the center frequency of the broadband pulse and extending to wavenumbers proportional to the signal bandwidth. A corresponding analytic broadband volume scattering model is derived from the Rayleigh-Born approximation to Green's theorem.

3:45

4pUW10. Objective identification of the dominant seabed scattering mechanism. Gavin Steininger (SEOS, U Vic, 201 1026 Johnson St., Victoria, BC v7v 3n7, Canada, gavin.amw.steininger@gmail.com), Charles W. Holland (SEOS, U Vic, State College, Pennsylvania), Stan E. Dosso, and Jan Dettmer (SEOS, U Vic, Victoria, BC, Canada)

This paper develops and applies a quantitative inversion procedure for scattering-strength data to determine the dominant scattering mechanism (surface and/or volume scattering) and to estimate the relevant scattering parameters and their uncertainties. The classification system is based on trans-dimensional Bayesian inversion with the deviance information criterion used to select the dominant scattering mechanism. Scattering is modeled using first-order perturbation theory as due to one of three mechanisms: interface scattering from a rough seafloor, volume scattering from a heterogeneous sediment layer, or mixed scattering combining both interface and volume scattering. The classification system is applied to six simulated test cases where it correctly identifies the true dominant scattering mechanism as having greater support from the data in five cases; the remaining case is indecisive. The approach is also applied to measured backscatter-strength data from the Malta Plateau where volume scattering is determined as the dominant scattering mechanism. This conclusion and the scattering/geoacoustic parameters estimated in the inversion are consistent with properties from previous inversions and/or with core measurements from the site. In particular, the scattering parameters are converted from the continuous scattering models used in the inversion to the equivalent discrete scattering parameters, which are found to be consistent with properties of the cores. [Work supported by ONR.]

4:00

4pUW11. Laboratory measurements of backscattering strengths from two-types of artificially roughened sandy bottoms. Su-Uk Son (Dept. of Marine Sci. and Convergent Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 426-791, South Korea, suuk2@hanyang.ac.kr), Sungho Cho (Maritime Security Res. Ctr., Korea Inst. of Ocean Sci. & Technol., Ansan, Gyeonggi-do, South Korea), and Jee Woong Choi (Dept. of Marine Sci. and Convergent Technol., Hanyang Univ., Ansan, Gyenggi-do, South Korea)

In the case of sandy bottom, the backscattering from the interface roughness is significantly dominant compared to that from the volume inhomogeneities, and the power spectrum of interface roughness thus becomes the most important factor to control the scattering mechanism. Backscattering strength measurements with a 50-kHz signal were made for two types of roughness (smooth and rough interfaces) which were artificially formed on a 0.5-m thick sandy bottom in a 5-m deep water tank. The roughness profiles were estimated by the arrival time analysis of 5-MHz backscattering signals emitted by the transducer moving parallel to the interface at a speed of 1 cm/s, which were then Fourier transformed to yield power spectra. In this talk, the measurements of backscattering strength as a function of grazing angle in a range of 35 to 90° are presented. Finally, the effect of different roughness types on the scattering strength will be discussed in comparison with the predictions obtained by theoretical scattering model including the perturbation and Kirchhoff approximations. [This research was supported by the Agency for Defense Development, Korea.]

4:15

4pUW12. Multistatic performance prediction for Doppler-sensitive waveforms in a shallow-water environment. Cristina Tollefsen (Defence Res. and Development Canada, P. O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, cristina.tollefsen@gmail.com)

Navies worldwide are now operationally capable of exploiting multistatic sonar technology. One of the purported advantages of multistatics when detecting directional targets should be the increased probability of receiving a strong reflection at one of the multistatic receivers. However, it is not yet clear (or intuitive) how best to deploy multistatic-capable assets to achieve particular mission objectives. The Performance Assessment for Tactical Systems (PATS) software was recently developed by Maritime Way Scientific under contract to Defence Research and Development Canada as a research tool to assist in exploring different approaches to multistatic performance modelling. Beginning with a user-defined environment and sensor layout, PATS uses transmission loss and reverberation model results to calculate signal excess at each grid point in the model domain. Monte Carlo simulations using many realizations of target tracks allow for the calculation of the cumulative probability of detection as a means to assess performance. Results will be presented comparing the shallow-water performance of monostatic and multistatic sensors using frequency-modulated and Doppler-sensitive waveforms as well as omnidirectional and directional targets in a variety of realistic military scenarios.

4:30

4pUW13. Twinkling exponents for backscattering by spheres in the vicinity of airy caustics associated with reflections by corrugated surfaces. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

High frequency sound reflected by corrugated surfaces produce caustic networks relevant to sea surface reflection [Williams *et al.*, J. Acoust. Soc. Am. 96, 1687–1702 (1994)]. When a sphere is positioned sufficiently far from the reflecting surface, it may be close to an Airy caustic which causes a significant increase in the backscattering [Dzikowicz and Marston, J. Acoust. Soc. Am. 116, 2751–2758 (2004)] for signals that bounce only once off of the focusing surface. For simplicity, here, it is assumed that those signals may be distinguished from the earlier direct echo from the sphere and the later (and sometimes stronger) doubly focused echo from the sphere [Dzikowicz and Marston, J. Acoust. Soc. Am. 118, 2811–2819 (2005)]. In 1977, M. V. Berry noticed that the third and higher intensity moments of wavefields containing caustics can increase in proportion to k^{ν}, where k is the wavenumber and ν is a "twinkling exponent" determined by the dependencies of the intensity and focal volume on k. Assuming that the sphere is impenetrable and sufficiently large that its direct scattering depends only weakly on k, for the single-bounce backscattering by a sphere considered here (the easiest situation for applying Berry's analysis) the predicted exponent for the third moment is $\nu = 1/3$.

THURSDAY EVENING, 30 OCTOBER 2014 — 7:30 P.M. TO 9:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. On Tuesday the meetings will begin at 8:00 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m. On Wednesday evening, the Technical Committee on Biomedical Acoustics will meet starting at 7:30 p.m. On Thursday evening, the meetings will begin at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

Animal Bioacoustics	Lincoln
Musical Acoustics	Santa Fe
Noise	Marriott 3/4
Psychological and Physiological Acoustics	Marriott 1/2
Signal Processing in Acoustics	Indiana G
Underwater Acoustics	Indiana F