Session 2aAA

Architectural Acoustics: Uncertainty in Describing Room Acoustics Properties I

Lily M. Wang, Cochair
Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

Ingo B. Witew, Cochair
Inst. of Techn. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany

Chair's Introduction—8:15

Invited Papers

8:20

2aAA1. Review of the role of uncertainties in room acoustics. Ralph T. Muehleisen (Decision and Information Sci., Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

While many aspects of room acoustics such as material characterization, acoustic propagation and room interaction, and perception have long been, and continue to be, active areas of room acoustics research, the study of uncertainty in room acoustics has been very limited. Uncertainty pervades the room acoustic problem: there is uncertainty in measurement and characterization of materials, uncertainty in the models used for propagation and room interaction, uncertainty in the measurement of sound within rooms, and uncertainty in perception. Only recently are the standard methods of uncertainty assessment being systematically employed within room acoustics. This paper explains the need for systematic study of uncertainty in room acoustic predictions and review some of the most recent research related to characterizing uncertainty in room acoustics.

8:40

2aAA2. Uncertainty and stochastic computations in outdoor sound propagation. D. Keith Wilson (CRREL, U.S. Army ERDC, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil) and Chirs L. Pettit (Aerosp. Eng., U.S. Naval Acad., Annapolis, MD)

Outdoor sound propagation provides an interesting and informative example of uncertainty in acoustics. Predictions are strongly impacted by imperfect knowledge of the atmospheric and ground properties, as well as by random turbulence and unresolved elements of the landscape. This presentation describes the impact of such uncertainties and how they can be quantified with stochastic sampling techniques that are applicable to a wide variety of acoustical problems. Efficient and accurate computational approaches result from simultaneously sampling over frequency, uncertain environmental properties, and random processes. Among the techniques considered are ordinary Monte Carlo and Latin hypercube sampling, importance sampling based on relatively simpler propagation models, and adaptive importance sampling. When uncertainties in the atmospheric and ground properties dominate, importance sampling is found to converge to an accurate estimate with the lowest calculation time. When random turbulent scattering dominates, the sampling method has little impact.

9:00

2aAA3. Bias and reproducibility of sound power test methods. Matthew G. Blevins, Lily M. Wang, and Siu-Kit Lau (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 909 S 70th plz #4, Omaha, NE 68106, mblevins@huskers.unl.edu)

Sound power is a useful quantity in describing the strength of an acoustic source because its value is independent of distance. However, many standardized methods exist for the measurement of sound power and comparison between methods can give rise to discrepancies. An interlaboratory study was designed according to the ISO 5725 series to quantify the bias and reproducibility of three common sound power measurement methods in the HVACR industry: free field method, diffuse field method, and sound intensity method. A loudspeaker sound source was used to generate two test signals: a broadband signal with decreasing 5 dB slope per octave band, and the same broadband signal with discrete frequency tones at 58, 120, 300, and 600 Hz. The objective of the study is to quantify repeatability, reproducibility, laboratory bias, and measurement method bias, as well as investigate the influence of tones. The design of the interlaboratory study and preliminary results will be presented. The ISO 5725 methods used to investigate the sound power measurement methods in this study may be applicable to other room acoustic measurements.

9:20

2aAA4. Uncertainty aspects regarding the input for reverberation time predictions. Margriet Lautenbach (Peutz bv, PO Box 696, Zoetermeer 2700 AR, Netherlands, m.lautenbach@peutz.nl) and Martijn Vercammen (Peutz bv, Mook, Netherlands)

The outcome of a reverberation time prediction cannot be more accurate than the combined accuracy of input parameters. In the current procedures of measuring absorption coefficients and using them in prediction models at least two aspects regarding the accuracy of absorption coefficients are quite underexposed:(1) The precision of the measurement: the reproducibility both within one laboratory as between different laboratories or laboratory conditions; (2) The accuracy of the measurement: the dependency of the absorption coefficient on the measurement procedure in general. The upcoming ISO 354 improves both aspects by means of a more elaborate qualification procedure, the use of a reference absorber combined with a correction procedure, as well as a guidance for the extrapolation of the results to other dimensions. Still, it is interesting to think about the impact of the remaining accuracy. To what extend is an acoustic consultant "at risk" using absorption coefficients, apart from using correct modeling algorithms. Two different, but common cases, can give an idea of the influence of the accuracy, two situations in which the reverberation time heavily depends on one absorption material: a class room with an absorptive ceiling, and a concert hall with absorptive chairs.

9:40

2aAA5. Investigations into the sound absorbing properties of gypsum wall board. Robert Healey (Architectural Eng., Univ. of Kansas, 1241 Tennessee Apt. 3, Lawrence, KS 66044, rwhealey@ku.edu), Kevin Butler (Henderson Engineers, Inc., Lawrence, KS), Ian Patrick, and Sean Quigley (Architectural Eng., Univ. of Kansas, Lawrence, KS)

Gypsum wall board is one of the most common materials encountered in buildings and thus encountered in architectural acoustics. Recent research into gypsum wall board sound absorption has indicated that the material may have significantly less absorption than is usually assumed when employed in certain construction assemblies. This paper examines both traditional mountings and a new method for determining sound absorption recently introduced involving mounting a gypsum board assembly in an opening between two reverberant rooms commonly used for transmission loss testing. Sound absorbing data were obtained with the two-room method along with the traditional measurement method, for several gypsum wall board wall assemblies, and with an attempt to compare the measured results to real world experience.

10:00–10:15 Break

10:15

2aAA6. The effects of uncertain scattering coefficients on the reverberation time. Uwe M. Stephenson (Room Acoust., HafenCity Univ. Hamburg, Nelkenweg 10, Bad Oldesloe 23843, Germany, post@umstephenson.de) and Alexander Pohl (Room Acoust., HafenCity Univ. Hamburg, Hamburg, Germany)

Uncertain scattering coefficients are still a weak point in room acoustical computer simulations and predictions. The definition of the scattering coefficient s, their values as well as the simulation model are uncertain. Not only the roughness of infinite walls but also the edge effect are included in a combined diffusivity coefficient ray tracing models utilize. Most interesting in practical room acoustics is the influence on the uncertainty of the reverberation time (RT), which depends very sensitively on s. However, there is no analytical formula for the RT for only partially diffusely reflecting surfaces. In non-diffuse sound fields, the RT depend on the room shape, the distribution of absorption and especially on the scattering coefficients, too. A crucial example is a long rectangular room with totally absorbing side walls and partially scattering front walls. For this case a semi-analytical approach had been found. These and other typical cases in 2D and 3D have now been investigated numerically by the Sound Particle Simulation Method (SPSM) and the Anisotropic Reverberation Model (ARM). The aim is to find semi-analytically formulae or rules to estimate the uncertainty in the prediction of the RT by the Sabine formula. Is an "equivalent scattering area" a useful concept?

10:35

2aAA7. Uncertainty in scattering coefficient measurements of sintered ceramic tiled surfaces. David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604, dabradley@vassar.edu), Rhett Russo (School of Architecture, New Jersey Inst. of Technol., Newark, NJ), Ariana Sharma, and Jacob Adelgren (Phys. + Astronomy, Vassar College, Poughkeepsie, NY)

The reflection of acoustic energy in an enclosed space can sometimes lead to undesirable effects, particularly if the reflection is relatively large in amplitude or delayed in time. To mitigate these effects, surfaces with non-planar geometries, which are known as diffusers, can be employed in an architectural space to improve the acoustical qualities of the space by attenuating these harsh reflections and by producing a more evenly distributed sound field. One of the standardized quantifiers used to characterize the effectiveness of a diffuser is the scattering coefficient, which is defined as the ratio of non-specularly reflected energy to total reflected energy. Measuring the scattering coefficient requires a carefully controlled acoustic testing facility known as a reverberation chamber. These chambers often have attributes that can be difficult to control (e.g., humidity) or that do not adhere to the standardized specifications (e.g., size). The current study explores the uncertainty associated with a series of measurements focused on testing the effectiveness of a new type of diffuser, which has been created using a novel ceramic sintering technique. The scattering coefficients of several of these ceramic surfaces have been measured in three different reverberation chambers with varying results.

10:55

2aAA8. Characterization of the uncertainty and error propagation in sound field diffusion measurements. Jin Yong Jeon, Muhammad Imran, and Hyung Suk Jang (Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133791, South Korea, jyjeon@hanyang.ac.kr)

ISO 3382-1 describes instructions for conducting acoustical measurements in auditoria. This standard defines how to measure and derive general properties of the acoustic conditions. By applying the general instructions in ISO, a modified Np was investigated by counting the number of peaks from room impulse responses in an auditorium to characterize the sound field diffuseness. The accuracy of the parametric value in a space depends on the uncertainties involved in all stages of the measurements and analyzing processes. Uncertainties related to source/receiver characteristics, directional aspects, and computational techniques are considered in computing the modified Np. Therefore, we discuss the extent to which we are uncertain about the modified Np and the factors affecting its accuracy in all processes involved in measurements and computations.

11:15

2aAA9. Uncertainty in acoustic metrics due to spatial variations of the non-diffuse sound field measured in a variable acoustics classroom. Zhao Peng, Matthew G. Blevins, 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)

Acoustic metrics are commonly expressed as single numbers in classroom acoustical designs, often neglecting the physical quantity's uncertainty due to the non-diffuse sound field in the seating area. A database of measured monaural and binaural room impulse responses (RIR) was previously gathered from a fully furnished mock-up classroom. Different wall and ceiling absorption configurations were used to alter the mid-frequency reverberation times (RT) in five scenarios between 0.4 and 1.1 s. The middle three RT scenarios were additionally created from two different material configurations. For each material configuration (eight in total), two furniture orientations were utilized. RIRs were measured at 9 to 10 receiver positions for each material/furniture configuration to document the spatial variation in the resulting sound field. Diffuseness has been calculated for each receiver position utilizing the measured RIRs by following Hanyu's (2013) method using normalized decay-canceled impulse responses. Variations in diffuseness and in the assorted acoustic metrics calculated from the measured RIRs are investigated across different receiver positions. These acoustic metrics, pertinent to classroom acoustical designs, include RT, speech transmission index, clarity, and interaural cross-correlation. Means to quantify uncertainty in these metrics due to spatial variation in the non-diffuse sound field will be discussed.

11:35

2aAA10. Uncertainty versus parameter in room acoustics. A case study. Miguel Arana, Abel Arregui, Jorge Machin, and Ricardo San Martin (Phys., Public Univ. of Navarre, Campus de Arrosadia, Pamplona, Navarra 31006, Spain, marana@unavarra.es)

An exhaustive characterization of the auditorium of Baranain (Navarre, Spain) has been carried out. All acoustic parameters (both monaural and binaural) in many seats (96 for monaural and 48 for binaural) have been measured for three source positions on the stage. For acoustic characterization, a countless results can be obtained in accordance (in all cases) with the views of the ISO-3382 for the presentation of the results. The spatial dispersion for each source position and combinations thereof will be shown. Accuracy on the acoustic evaluation of the room will be discussed from a statistical point of view.

TUESDAY MORNING, 6 MAY 2014 554 A/B, 8:00 A.M. TO 11:25 A.M.

Session 2aAB

Animal Bioacoustics and Signal Processing in Acoustics: Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals II

James A. Simmons, Chair
Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912

Invited Papers

8:00

2aAB1. Characteristics of echolocation system in Japanese house bat, *Pipistrellus abramus*. Hiroshi Riquimaroux (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

Japanese house bats, *Pipistrellus abramus*, emit harmonically structured downward FM sweeps for echolocation where the fundamental frequency changes from 80 to 40 kHz. Previous studies with an on-board telemetry microphone revealed that they conduct echo-amplitude compensation to stabilize amplitude of returning echoes. When they hunt preys in the field, emitted pulses are prolonged, and the fundamental frequency little change where the terminal fundamental frequency is fixed at about 40 kHz. About 40% of neurons in the inferior colliculus are tuned best to a frequencies range between 35 and 45 kHz, where higher frequency resolution than other frequency ranges is implied and suited for detecting wing beats of insects. However, their audiogram has not yet been known. The present study constructed their audiograms between 4 and 80 kHz by using the auditory brainstem responses evoked by tone pips. Results show that threshold around 40 kHz is lower than other frequencies except for frequency regions around 20 kHz. Findings suggest that Japanese house bats appear to have high sensitivity around 40 kHz, the terminal frequency, and around 20 kHz, corresponding to their communication frequencies. Characteristics of their cochlear microphonics will be discussed. [Work supported by ONR.]

8:20

2aAB2. Adaptive changes in acoustic characteristics of echolocation pulses emitted by Japanese house bats under artificial jamming conditions. Shizuko Hiryu, Eri Takahashi, Kiri Hyomoto, Yoshiaki Watanabe, Hiroshi Riquimaroux, and Tetsuo Ohta (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, shiryu@mail.doshisha.ac.jp)

The echolocation behavior of *Pipistrellus abramus* during exposure to artificial jamming sounds during flight was investigated. Echolocation pulses were recorded using a telemetry microphone mounted on the bats' backs, and their adaption based on acoustic characteristics of emitted pulses was assessed in terms of jamming-avoidance responses (JAR). In experiment 1, FM jamming sounds mimicking echolocation pulses of *P. abramus* were prepared. All bats showed significant increases in the terminal frequency (TF) by an average of 2.1–4.5 kHz when the TF of the jamming sounds was lower than the bats' own pulses. This frequency shift was not observed using jamming frequencies that overlapped with or were higher than the bats' own pulses. These findings suggest that JAR in *P. abramus*are sensitive to the TF of jamming pulses and that the bats' response pattern was dependent on the slight difference in stimulus frequency. In experiment 2, when bats were repeatedly exposed to a band-limited noise, the bats in flight more frequently emitted pulses during silent periods between jamming sounds. These results demonstrate that bats could rapidly adjust their vocalized frequency and emission timing to avoid frequency and temporal overlap with jamming sound even during flight. [Research supported by JSPS.]

Contributed Papers

8:40

2aAB3. Dynamics of biosonar signals of free swimming dolphins searching for bottom targets. Whitlow W. Au, Adrienne Copeland (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu), Stephen W. Martin (Navy Marine Mammal Program, Spa War System Ctr., San Diego, CA), and Patrick W. Moore (National Marine Mammal Foundation, San Diego, CA)

Measuring on-axis biosonar signals from a free swimming dolphin performing a sonar task is extremely difficult without having a special device that the animals carry. A bite-plate device which had a hydrophone directly in front of the dolphin at a fixed location along the beam axis of the biosonar beam was constructed as a part of the Navy Marine Mammal Research Program in San Diego. A data acquisition and storage unit was a part of the Biosonar Measurement Toolbox (BMT) and hung below the bite plate. The outgoing signal was measured by the omnidirectional hydrophone while two disk hydrophones measured the echoes. The device was used with two Atlantic bottlenose dolphins (*Tursiops truncatus*) as the animals conducted biosonar searches for specific objects on the ocean bottom. The outgoing signals were parameterized into the following, peak-to-peak source levels, source energy flux density, center frequency, rms bandwidth, rms duration, and interclick intervals. The parameters were analyzed with both a cluster analysis and a principle component analysis to determine the grouping of the parameters and the relationship between parameters. Both dolphins used different biosonar search strategy in solving the problem and their biosonar signals reflect the difference in strategy.

8:55

2aAB4. Origin of the off-axis double-pulse in an echolocating bottlenose dolphin. Dorian S. Houser, Brian Branstetter, Jason Mulsow, Patrick Moore (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), and James J. Finneran (SSC Pacific, San Diego, CA)

It has been proposed that two pairs of phonic lips might be utilized by a dolphin to adaptively manipulate the frequency content and directionality of the echolocation beam in response to acquired target information. The presence of two pulses appearing off-axis of the echolocation beam and separated in time has been proposed as evidence supporting this hypothesis. An array containing 35 hydrophones was used to measure the beam pattern of a bottlenose dolphin performing a phantom echo-change detection task. Simulated target ranges varied from 2.5 to 80 m and clicks were measured at 5–10° resolution from +/150°. At recording angles beyond +/30°, the click appeared as two distinct pulses that declined in amplitude and distorted as the off-axis angle increased. A simple model utilizing the time difference of arrival for the two pulses was used to compare the direct source-receiver path to one of two source-reflector-receiver paths. Assuming a range of constant sound speeds, distances traveled were compared to a CT scan of the same animal to predict anatomical regions potentially contributing to the second pulse. Results suggest that the second pulse is due to reflections from internal structures of the dolphin head and not a second sound source.

9:10

2aAB5. Can you hear me now? Sensitive hearing and limited variation in wild beluga whales (*Delphinapterus leucas*). T. Aran Mooney (Biology Dept., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu), Manuel Castellote (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, Seattle, WA), Lori Quakenbush (Arctic Marine Mammal Program, Alaska Dept. of Fish and Game, Fairbanks, AK), Roderick Hobbs (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, Seattle, WA), Caroline Goertz (Alaska SeaLife Ctr., Seward, AK), and Eric Gaglione (Georgia Aquarium, Atlanta, GA)

Odontocetes use sound for vital biological functions such as foraging, navigation, and communication, and hearing is considered their primary sensory modality. Yet, hearing abilities within wild species or populations are essentially unknown. Hearing sensitivities and variations can greatly influence biosonar signals and performance. Here we present the hearing abilities of seven wild beluga whales. Data were collected in a capture-release event in Bristol Bay, AK. The goal was to establish the mean audiogram and variation within a subset of animals, defining what wild belugas hear and examining how sound-sensitivities may differ between individuals. Hearing was measured using auditory evoked potentials, from 2 to 150 kHz. All belugas tested could hear up to at least 128 kHz, and two heard up to 150 kHz, exceptionally high for belugas. Regions of "best" hearing (<60 dB) were found between 22 and 100 kHz, showing generally broad sensitivity. Greatest variation (>40 dB) was at regions of best sensitivity and the highest frequencies. This substantial variation was less than that of bottlenose dolphins suggesting differences between populations and the need to collect comparative data on new species. While generally sensitive, the hearing variability suggests that multiple analyses better describe the maximum sensitivity and population variance for odontocetes.

9:25

2aAB6. Echolocation strategy for multiple target-preys by foraging bats investigated by field measurement and mathematical modeling. Emyo Fujioka (FIRST, Aihara Innovative Mathematical Modelling Project, JST, 4-6-1-Cw601 Komaba, Meguro-ku, Tokyo, Japan, Tokyo 153-8505, Japan, emyo.fujioka@gmail.com), Ikkyu Aihara (Brain Sci. Inst., RIKEN, Saitama, Japan), Shotaro Watanabe (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), Miwa Sumiya, Shizuko Hiryu, Yoshiaki Watanabe, Hiroshi Riquimaroux (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan), and Kazuyuki Aihara (Inst. of Industrial Sci., The Univ. of Tokyo, Tokyo, Japan)

Using a super microphone-array system, 3-D flight paths of echolocating Japanese house bats, *Pipistrellus abramus*, across whole foraging area were successfully measured in the field, together with the directional aim of their sonar beams. Sonar sounds were sometimes rapidly alternated between its direction and other one or two particular directions during search phase. Especially, when the bats consecutively captured multiple insect preys, their emissions were directed toward not only the current target prey and also the next target. These suggest that the bats process multiple echo streams by time-sharing manner and plan the flight path to efficiently capture multiple target preys. In order to examine whether the bats select the efficient flight path to consecutively capture multiple targets, the bats' 3-D flight behavior while approaching two target preys was then modeled. The modeling analysis suggested that the echolocating bats select their flight paths to easily direct their sonar beams toward both targets. [This research was supported by the Aihara Project, the FIRST program from JSPS, initiated by CSTP, a Grant-in-Aid for Young Scientists (A) of JSPS, and the Murata Science Foundation.]

9:40

2aAB7. Eigenbeam analysis of the diversity in bat biosonar beampatterns. Philip Caspers, Alexander Leonessa, and Rolf Mueller (Mech. Eng., Virginia Tech, 1110 Washington St., SW, MC 0917, Blacksburg, VA 24061, pcaspers@vt.edu)

A quantitative analysis of the interspecific variability in the biosonar beampatterns of bats has been performed on a data set that consisted of 267 emission and reception beampatterns from 98 different species. The beampatterns were aligned using a pairwise optimization framework defined by a rotation for which a cost function is minimized. The cost function was defined by a p-norm computed over all direction and summed across a discrete set of evenly sampled frequencies. For a representative subset of beampatterns, it was found that all pairwise alignments between beampatterns result in a global minimum that fell near the plane bisecting the mean direction of each beampattern and containing the origin. Following alignment, the average beampattern was found to consist of a single lobe that narrowed with increasing frequency. Variability around the average beampattern was analyzed using principle component analysis (PCA) that resulted in "eigenbeams": The first three "eigenbeams" were found to control the beamwidth of the beampattern across frequency while higher rank eigenbeams accounted for symmetry breaks and changes in lobe direction. Reception and emission beampattern could be differentiated based on their PCA scores using only a small number of eigenbeams.

9:55

2aAB8. Influence of mouth opening and gape angle on the transmitted signals of big brown bats (*Eptesicus fuscus*). Laura N. Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu), Jason Gaudette (Adv. Acoust. Div., Naval Undersea Warfare Ctr., Providence, RI), James Simmons (Neurosci., Brown Univ., Prov, RI), and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

The big brown bat (*Eptesicus fuscus*) produces echolocation sounds in its larynx and emits them through its mouth. The bat is presumed to change the directionality of the emitted echolocation beams by modifying its mouth opening width. We analyzed infrared video sampled at 240 fps synchronized to ultrasonic recordings from a Knowles Electret microphone sampled at 192 kHz. Mouth angles for each emitted echolocation pulse were calculated offline and compared to the pulse's time-frequency characteristics. Our results indicate that the mouth influences both the amplitude and spectral characteristics of the emitted pulse.

10:10–10:25 Break

10:25

2aAB9. Target shape perception and clutter rejection use the same mechanism in bat sonar. Michaela Warnecke (Neurosci., Brown Univ., 185 Meeting St., Providence, RI 02912, michaela_warnecke@brown.edu) and James A. Simmons (Neurosensing and Bionavigation Res. Ctr., Doshisha Univ., Providence, Rhode Island)

Big brown bats (*Eptesicus*) emit multiple-harmonic FM sounds (FM1, FM2) and exploit the relative weakening of higher harmonics in lowpass echoes from the surrounding scene to suppress clutter by defocusing of wideband images. Only echoes from a frontally located targets arrive as unfiltered, focused images. Experiments using electronically generated echoes show that lowpass filtering of masking echoes causes clutter masking to disappear. Lowpass filtering induces amplitude-latency trading, which retards response times at higher frequencies in clutter echoes relative to lower frequencies. Introducing countervailing changes in presentation-times of higher frequencies in electronically generated clutter echoes restores masking. In the big brown bat's inferior colliculus, FM sounds mimicking broadcasts and echoes evoke ~1 spike per sound at each neuron's best frequency; however, amplitude tuning is very broad. To exploit their high acuity for detecting coherence or non-coherence of echo responses, bats work in the latency domain instead, removing background objects through deliberate imposition of response de-synchronization and concomitant inattention on undesired clutter. Overall, the results indicate that big brown bats use neuronal response timing for virtually all auditory computations of echo delay, including clutter rejection. This use of active perceptual processes in biosonar instead of conventional sonar processes opens a new view toward biomimetic design.

10:40

2aAB10. Acoustic tracking of bats in clutter environments using microphone arrays. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp), Alyssa Wheeler, Laura Kloepper, Jason Gaudette, and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

The big brown bat, *Eptesicus fuscus*, uses echolocation for foraging and orientation. Bats can change the echolocation calls dependent on the environments. Therefore, it is necessary to clarify the changes of acoustic characteristics of these calls. In this study, the flight path of the bat were tracked by computing the time differences of arrivals (TDOA) at the microphone array system in the flight room. The acoustic patterns of echolocation calls could be calculated from the measured call data at each microphone by compensating the spread and absorption loss. The head aim and beam pattern at each harmonics were computed from these acoustic patterns of echolocation calls. It was examined whether these acoustics beam patterns were dependent on clutter environment, that is, density of chains. [This research was supported by ONR, NSF, and JST, CREST.]

10:55

2aAB11. Bats (*Tadarida brasiliensis*) jam conspecifics in food competition. Aaron J. Corcoran (Biology, Univ. of Maryland, 4309 Rowalt Dr. #201, College Park, MD 20740, aaron.j.corcoran@gmail.com) and William E. Conner (Biology, Wake Forest Univ., Winston Salem, NC)

We here describe field experiments testing whether bats adaptively produce sounds to interfere with ("jam") the echolocation of other bats. Visual observations, low-light videography, and ultrasound recording with microphone arrays allowing reconstruction of bat flight paths were used to document interactions between Mexican free-tailed bats (*Tadarida brasiliensis*) at two foraging locations in southern Arizona and New Mexico. We tested three sets of predictions based on the jamming hypothesis and two competing hypotheses—cooperative foraging and food patch defense. Bats produced putative jamming calls (termed sinFM calls) that overlapped

temporally and spectrally with "feeding buzz" calls made by conspecifics when attacking insects. Bat capture success decreased by 400% when sinFM calls were present compared to when they were absent. Behavioral sequences consisted of two or more bats sequentially making feeding buzzes within a restricted area while another bats made sinFM calls. After making sinFM calls bats frequently turned toward where the other bat made its feeding buzz and then made a buzz of its own. Together, the results support the hypothesis that bats jam conspecifics in extended bouts of food competition. This is the first known case of echolocating animals adaptively jamming conspecifics.

11:10

2aAB12. Temporal dynamics of echolocation during clutter navigation in *Eptesicus fuscus*. Alyssa Wheeler, Laura Kloepper (Neurosci., Brown Univ., 189 Meeting St., Box GL-N, Providence, RI 02903, Alyssa_Wheeler@Brown.edu), Ikuo Matsuo (Information Sci., Tokohu Gakuin Univ., Sendai, Japan), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

The echolocating big brown bat, Eptesicus fuscus, must both avoid background objects that it may collide with such as vegetation or clutter, and identify insect prey targets. When bats fly through clutter, they emit groups of echolocation sounds in rapid succession called strobe groups. Here, we investigate the limitations of strobe grouping for flight guidance during clutter navigation. We hypothesized that as clutter conditions become extreme, bats will be no longer able to use strobe groups to navigate. We exposed bats to our laboratory flight room cluttered with plastic chains—obstacles that are acoustically similar to vegetation. Bats flew down a corridor 1.0 m wide, 0.7 m wide, or 0.4 m wide, and echolocation sounds were recorded with a 22-microphone array. We used a means clustering analysis to quantify inter-pulse intervals (IPIs) as belonging within a strobe group or between strobe groups. We found that bats made significantly more sounds per flight as the corridor width decreased. Strobe groups in the wide condition contained longer within-strobe IPI, while strobe groups in the narrowest condition had the shortest. We also investigated how many sounds per strobe group were used. These results show a relationship between clutter density and the temporal structure of echolocation calls.

TUESDAY MORNING, 6 MAY 2014 — BALLROOM E, 8:00 A.M. TO 11:30 A.M.

Session 2aBA

Biomedical Acoustics: Brain Therapy and Imaging

Yun Jing, Cochair
Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

Gregory T. Clement, Cochair
Cleveland Clinic, 9500 Euclid Ave., Cleveland, OH 44195-0001

Invited Papers

8:00

2aBA1. Recent advancement in transcranial brain imaging using photoacoustic computed tomography. Mark Anastasio, Kenji Mitsuhashi, Chao Huang, Robert Schoonover, Konstantin Maslov, Alejandro Garcia-Uribe, and Lihong V. Wang (Washington Univ. in St. Louis, One Brookings Dr., St. Louis, MO 63130, anastasio@wustl.edu)

Photoacoustic computed tomography (PACT) holds great promise for transcranial brain imaging. However, the strong reflection, scattering, and attenuation of acoustic waves by the skull present significant challenges for image reconstruction. In this talk, we will review our recent progress on transcranial PACT image reconstruction. Our contributions include the following: (1) development of a methodology to establish a discrete transcranial PACT image model by use of adjunct X-ray CT data; (2) development of image reconstruction methods that can compensate for speed-of-sound and density variations within the skull; and (3) a detailed investigation of the role of shear waves in transcranial PACT. Computer-simulated and experimental data are employed to demonstrate the feasibility of transcranial PACT.

8:20

2aBA2. Passive mapping of acoustic sources within the human skull cavity with a hemispherical sparse array using computed tomography-based aberration corrections. Ryan M. Jones (Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Meaghan A. O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst. , Toronto, ON, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Traditionally, the use of ultrasound (US) in the brain has been limited by the skull bone, which presents unique challenges for both transcranial therapy and imaging due to its attenuating and aberrating effects, which become more prevalent at higher US frequencies. On transmit, these skull-induced aberrations can be overcome through the use of large-aperture phased array transducers with

appropriate driving frequencies, combined with computed tomography (CT)-based bone morphology and numerical models to derive element driving signals which minimize the distortions. Recently, we have demonstrated *in silico* that an analogous approach can be performed during beamforming on receive, to allow for passive acoustic imaging over a large region within the skull cavity [Jones *et al.*, *Phys. Med. Biol.* **58**, 4981–5005 (2013)]. We will present preliminary results obtained from applying this technique experimentally with a hemispherical (30 cm diam.) sparse receiver array (128 piezo-ceramic elements, 2.5 mm diam., and 612 kHz center frequency) to image acoustic sources through an *ex vivo* human skullcap. Images produced using non-invasive CT-based skull corrections will be compared with those obtained through an invasive hydrophone-based correction approach, and with images formed without skull-specific corrections. This technique has promising applications in both cavitation-mediated transcranial focused ultrasound therapies, by providing a method for treatment monitoring and control, as well as in ultrasound angiographic imaging of the brain.

8:40

2aBA3. Photoacoustic imaging of brain cortex in rhesus macaques. Xinmai Yang (Mech. Eng., The Univ. of Kansas, 1560 W 15th St., Lawrence, KS 66045, xmyang@ku.edu)

Functional detection in the brain by photoacoustic (PA) imaging has been an area of interest because of its potential to overcome the limitation of the current available techniques. Both small and large animal brains have been studied by PA imaging. Functional detection in primate brains is particularly of interest because of the similarity between non-human primate brain and human brain and the potential for relevance to a wide range of conditions such as stroke and Parkinson's disease. In this presentation, we show the application of PA imaging technique in detecting functional changes in primary motor cortex of awake rhesus monkeys. Strong increases in PA signal amplitude during forelimb movement indicate an increase in activity in primary motor cortex. The results demonstrate that PA imaging can reliably detect primary motor cortex activation associated with forelimb movement in rhesus macaques. The potentials of PA imaging on clinical human brain imaging will also be discussed.

9:00

2aBA4. Transcranial ultrasound hemorrhage detector. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu), Amber B. Bennoui, Joleigh V. Ferro (Chemistry and Phys., Simmons College, Boston, MA), Charles D. Maneval, Mufaddal A. Jafferji, Nicholas J. Giordano (Biomedical Eng., Boston Univ., Boston, MA), Greg T. Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), and Phillip J. White (Radiology, Brigham and Women's Hospital, Boston, MA)

The present state-of-the-art technology in clinical neurosonography requires an acoustic window to achieve a sufficient signal-to-noise ratio (SNR) for detection of intracranial hemorrhage (ICrH). Conventional clinical practices use ultrasound in the range of 2–10 MHz to image and diagnose patients, a frequency range that is too high to produce high-quality transcranial images. This study examined the use of ultrasound frequencies on the order of 0.5 MHz, applied through the skull bone, to detect ICrH. We demonstrated with *ex vivo* human skull specimens, *ex vivo* animal blood, and soft tissue-mimicking phantoms that a single element ultrasonic transducer can likely be used to detect the presence of ICrH immediately adjacent to the skull bone. The lower frequency design of the transducer together with optimized aperture geometry demonstrated efficient signal transmission through the skull bone with a focal depth optimized for intracranial anatomy. The identification of hemorrhage-brain interfaces was demonstrated with bench-top experiments that mimicked the clinical conditions of acute epi- and subdural hemorrhage. Finally, the device limitations were demonstrated through positive predictive value analysis.

9:20

2aBA5. Transcranial ultrasound-optical transmission correlation. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu), Zun Zar Chi Naing, Felicity A. Meyer (Chemistry and Phys., Simmons College, Boston, MA), Mufaddal A. Jafferji, Chanikarn Power (Radiology, Brigham and Women's Hospital, Boston, MA), Greg T. Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), and Phillip J. White (Radiology, Brigham and Women's Hospital, Boston, MA)

Although the transmission of ultrasound (US) through the skull bone has been demonstrated for both therapeutic and imaging applications, the clinical efficacy of certain transcranial US applications remains limited by the highly attenuating properties of skull bone. For those applications, the ability to pre-procedurally determine those areas of the skull that are less attenuating to US could be a tremendous asset for improving the use of US in the brain. To present a possible solution, we have hypothesized that the optical transmission intensity at points across the skull surface can be correlated with the local US transmission efficiency. The demonstration of this correlation would potentially allow for the use of integrated lasers and photodetectors within a HIFU system to create a patient-specific transmission map of the skull. We have statistically investigated the relationship between transmitted optical and US intensities over multiple points across several *ex vivo* human calvaria to demonstrate this correlation. Along with the results of the analysis, preliminary designs to incorporate optical transmission assessment in transcranial HIFU and echoencephalography will be presented.

9:40–10:00 Break

10:00

2aBA6. Transcranial passive cavitation mapping with a linear array: A simulation study with clinical datasets. Costas D. Arvanitis (Radiology, Harvard Med. School, Brigham and Women, 221 Longwood Ave., Boston, MA 02115, cda@bwh.harvard.edu), Gregory Clement (Dept. of Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), and Nathan McDannold (Radiology, Harvard Med. School, Brigham and Women, Boston, MA)

While numerous investigations that explore the unique properties of acoustic cavitation have recently demonstrated very promising results for a wide range of applications, there is a need to develop new and noninvasive methods that will lead to (1) deeper understandings of the interactions involved, (2) their optimal use for therapy or diagnosis, and (3) ultimately their translation to the clinics. Toward this aim, we developed finite difference time domain (FDTD) simulations to transcranially map and assess the diverging pressure waves generated by oscillating microbubbles. The skull and brain tissue acoustic properties (density, speed of sound, and absorption) were extracted from clinical CT datasets. Point sources derived by microbubble dynamics models were also incorporated and propagated toward a virtual US array that was used to perform passive acoustic mapping (PAM). The FDTD simulations suggest that the microbubbles' pressure waves propagating through the skull lose 97% of their strength as compared to propagation in water. The simulations also indicated that transcranial PAM is possible with an 80 mm aperture linear array; however, at wider apertures (150 mm), significant aberration was introduced. Incorporation of variable speed of sound to the PAM back-projection algorithm corrects the aberrations and significantly improves the resolution.

10:20

2aBA7. On the use of fast marching methods for transcranial beam focusing. Tianren Wang and Yun Jing (Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

In this talk, we will present our recent studies on the use of fast marching methods for transcranial beam focusing. Three topics will be included: beam focusing for transcranial B-mode imaging, beam focusing for transcranial photoacoustic tomography, and beam focusing of a spherical array for therapy. To correct for the phase aberration from the skull, two critical steps are needed prior to brain imaging or treatment. In the first step, the skull shape and speed of sound are acquired by either CT scans or ultrasound scans. In the second step, fast marching methods are used to compute the phase delay based on the known skull shape and sound speed from the first step, and the computation can be completed in seconds even for 3D problems. The computed phase delays are then used in combination with the conventional delay-and-sum algorithm for generating images. They can also be readily used for transcranial beam focusing for therapeutic purposes. Numerical simulation results will be presented to show the robustness of fast marching methods.

10:40

2aBA8. Drug delivery through the opened blood-brain barrier in mice and non-human primates. Elisa Konofagou, Cherry Chen, Hong Chen, Matthew Downs, Vincent Ferrera, Oluyemi Olumolade, Gesthimani Samiotaki, Tao Sun, Shutao Wang, Shih-Ying Wu (Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Over five million U.S. men and women suffer from neurodegenerative diseases. Although great progress has been made in recent years toward understanding of these diseases, few effective treatments and no cures are currently available. This is mainly due to the impermeability of the blood-brain barrier (BBB) that allows only 5% of more than 7000 small-molecule drugs available to treat only a tiny fraction of these diseases. Safe and localized opening of the BBB has been proven to present a significant challenge. Focused ultrasound (FUS), in conjunction with microbubbles, remains the sole technique that can induce localized BBB opening noninvasively and regionally. In the past, our group has focused on cavitation monitoring during BBB opening in both mice and non-human primates, assessment of safety and drug efficacy using behavioral testing, delivery of molecules of variant size through the opened BBB, investigation on the role of the microbubble diameter and use of nanodroplets. We will briefly highlight these past findings as well as introduce newer accomplishments such as its role in enhancement of drugs for neuroprotection and neuroregeneration in the treatment of Parkinson's, the use of alternative routes of systemic administration for larger drug dosage, dependence of the BBB opening size on the acoustic pressure, real-time monitoring of the microbubble perfusion of the brain, cavitation prediction of the timeline of BBB opening, and targeted delivery using adeno-associated viruses.

Contributed Papers

11:00

2aBA9. Effects of diffraction on acoustic radiation force produced by sound beams incident on spherical viscoelastic scatterers in tissue. Benjamin C. Treweek (Dept. of Mech. Eng., Univ. of Texas at Austin, 610 West 30th St., Apt. 126, Austin, TX 78705, btreweek@utexas.edu), Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

The theory for acoustic radiation force on a viscoelastic sphere of arbitrary size in tissue was extended at the spring 2013 ASA meeting to account for nonaxisymmetric fields incident on the scatterer [Ilinskii *et al.*, POMA **19**, 045004 (2013)]. The results were presented in a form that permits inclusion of as many spherical harmonics as needed to describe the field structure. At the fall 2013 ASA meeting, it was shown that for spheres having sizes up to about one wavelength, only four or five spherical harmonics are required for convergence of the solution when plane waves are incident on the scatterer. At the present meeting, the model is applied to diffracting sound beams incident on the scatterer. The analysis is based on angular spectrum decomposition of the incident field, expansion of the resulting plane waves in spherical waves, then a Wigner transformation of the latter back into spherical coordinates with polar axis coinciding with the beam axis, and finally integration over solid angle to obtain the spherical wave amplitudes used in the theory. Results are presented for different radiation patterns illustrating dependence of the radiation force both on beamwidth and on wavelength relative to the size of the scatterer.

11:15

2aBA10. A novel device for guiding ventriculostomy with transcranial ultrasound. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu), Michael A. Persaud, Aaron E. Silva, Abhishek Mundra (Biomedical Eng., Boston Univ., Boston, MA), Greg T. Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), Kirby G. Vosburgh, and Phillip J. White (Radiology, Brigham and Women's Hospital, Boston, MA)

A ventriculostomy is often performed to relieve symptoms of emergent hydrocephalus. This involves the placement of an external ventricular drain (EVD) into the cerebral ventricles to remove excess cerebrospinal fluid. Free-handed EVD cannulation results in high rates of misplacement (~50%), leading to an increased risk of iatrogenic complications. Extant technical approaches to improve ventriculostomy guidance are either too complex or inaccurate. We have investigated the possibility of a novel device to guide EVD placement using transcranial ultrasound. The device uses three specifically aligned transducers delivering pulse-echo 0.5-MHz ultrasound through the skull bone to detect and localize the targeted ventricle. It also incorporates a cannula guide that is registered with the ultrasound FOV to integrate guidance with surgery. Results from the design, fabrication, and testing of the prototype device with *ex vivo* human skulls and brain phantoms will be presented.

TUESDAY MORNING, 6 MAY 2014 550 A/B, 9:00 A.M. TO 11:50 A.M.

Session 2aEA

Engineering Acoustics: Session in Honor of Kim Benjamin

Thomas R. Howarth, Chair
NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841

Chair's Introduction—9:00

Invited Papers

9:05

2aEA1. Reflections of Kim Benjamin as a student, professional colleague, and personal friend. Peter R. Stepanishen (Ocean Eng., Univ. of RI, Narragansett Bay campus, Sheets Bldg., Narragansett, RI 02882, stepanishen@egr.uri.edu)

Kim was a very special person who I was privileged to know as a student, professional colleague, and a close personal friend. After completing his undergraduate degree in Physics at URI in 1975, he was accepted into the Masters Program in Ocean Engineering at the University of Rhode Island where I served as his advisor and major professor. While working on an NIH grant to develop two-dimensional planar arrays of ultrasonic transducers for medical diagnostic purposes, Kim developed his lifelong passion for acoustic transducers. Some of his early "experiences with transducers" will be shared in the presentation. During this time period, Kim also investigated and developed the use of FFT methods for the backward projection of acoustic fields from planar sources. His early contributions in this area laid the foundation for much of the subsequent acoustical holographic work by others over the following several decades and will be briefly reviewed. In the 1980s and 1990, it was clear that Kim was becoming a "transducer guru" as evidenced by his research, reports, and related ASA presentations. I will share some recollections of Kim from this period and our many "extended lunches" during the last several years.

9:30

2aEA2. An overview of Kim Benjamin's U. S. Navy transducer developments. Thomas R. Howarth (NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

Kim Benjamin had already enjoyed a distinguished career as an acoustician at the University of Rhode Island and Raytheon Company before joining the U.S. Navy as a civilian scientist in 1995. From 1995 to 2013, Kim focused primarily on advancing 1–3 piezocomposite materials into unique underwater acoustic devices. Among key accomplishments are the following: design of 1–3 piezocomposite-based beam steered parametric mode transducers with integral high-gain receivers; design of parametric mode sub-bottom profiler transducers; development of U. S. Navy calibration transducer standards F82 and F83; use of 1–3 piezocomposites with area shaded electroding to realize a new class of transduction which maintains a constant beamwidth over a two octave bandwidth; novel use of singly and double curved piezocomposites; design and segment demonstration of a cylindrical array module that is coupled linearly to form a towed line array with 3D spatial discrimination; design and fabrication of a 120 element conical octahedral homing array for high speed (>150 knots) applications. He also was involved in the development of thin, low frequency acoustic sources as he designed a complex set of tooling to accomplish mission objectives. This presentation will overview Kim's Navy contributions with design drawings, photographs, and experimental data.

9:55

2aEA3. Novel, broadband piezocomposite transducers for Navy applications. Brian Pazol and Timothy Mayo (Materials Systems Inc., 543 Great Rd., Littleton, MA 01460, bpazol@matsysinc.com)

Piezocomposites (a two phased material consisting of piezoceramic and polymer) are a widely known form of piezoelectric material with proven advantages over conventional monolithic piezoelectric ceramics. Transducers made with piezocomposite are naturally broadband, have high sensitivity, can be easily shaded for sidelobe suppression, and can be conformed to a variety of shapes. Materials Systems Inc. (MSI) utilizes a low cost injection molding technique that allows large areas to be made by tiling ceramic preforms into large areas at a reasonable cost. There is a large design space that allows better transducer optimization by adjusting the matrix material, active material, and the ratio of active to inactive material (volume fraction). Kim Benjamin recognized the benefits of piezocomposite early in its development. Over the years, he developed many novel and high performance broadband piezocomposite transducers. This paper presents a brief overview of several unique piezocomposite transducers that he developed with MSI. These includes a Constant Beamwidth Transducer (CBT), torpedo homing arrays, and towed high power sources for a variety of Navy applications. Several of Kim's novel fabrication techniques and measured transducer performance will be presented.

10:20–10:30 Break

10:30

2aEA4. Constant beamwidth transducers: A tribute to Kim Benjamin. Dehua Huang (NAVSEA Newport, 43 Holliston Ave., Portsmouth, RI 02871, DHHuang@cox.net)

Constant beamwidth transducer (CBT) is a special acoustic transducer, where acoustic beamwidth is independent of frequency, because of its Legendre polynomial normal velocity distribution on the surface of transducer spherical dome. By elegant design of Legendre polynomial normal velocity distribution profile on the transducer radiation surface dome, acoustic sidelobes can also be controlled and eliminated. To achieve Legendre polynomial normal velocity distribution, electrode area shading is one of important techniques to design a practical CBT. in this paper, the CBT designs, shading patterns, size effects, frequency band limit, build of materials, as well as Mr. Kim Benjamin critical contributions in the field toward a U. S. Navy standard CBT transducer at the Underwater Sound Reference DivNpt (USRD) will be summarized.

10:50

2aEA5. Conformal cymbal array: A broadband, wide beamwidth underwater acoustic projector. James Tressler and Brian H. Houston (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, james.tressler@nrl.navy.mil)

The development of "cymbal"-based acoustic source technology in recent years has been an essential advancement in improving NRL's capability for shallow-water acoustics. The cymbal's unique attributes of high acoustic output from a lightweight, thin profile configurable array is a central technology in target identification programs at NRL. This talk will present a review of cymbal array development over the past ten years. It will start with conventional piezoelectric ceramic (PZT) cymbal elements and end with current research on advanced cymbal designs utilizing new formulations of piezoelectric single crystal materials. In particular, the contributions of Kim Benjamin to the initial development work of the conformal array design will be highlighted. [Work sponsored by the Naval Research Laboratory.]

11:10

2aEA6. Outstanding work of Kim Benjamin on 1–3 composite transducers and acoustic analysis. Kenneth M. Walsh (K&M Eng. Ltd., 51 Bayberry Ln., Middletown, RI 02842, kwalsh4@mindspring.com)

Kim Benjamin was one of the most productive acoustic system researchers that I have known. Kim, myself, and Walter Boober developed acoustic techniques that increased the productivity of the NUWC acoustic facility by a factor of 10. Kim and I produced a number of patents related to the use of 1–3 composite transducers in naval sonars. The last effort with Kim produced a small diameter, broad band transducer that was tested successfully. Kim was a Fellow of the ASA and is greatly missed.

11:30

2aEA7. Transducer cloaking for Kim Benjamin. John L. Butler (Image Acoust., Inc., 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

There is an effort to develop metamaterials for cloaking objects in a way that eliminates backscattering and fills in the shadow zone. The development of this cloaking material for spherical and other shapes would inhibit the acoustic detection of objects, such as mines, torpedoes, UUV's, and, ultimately, submarines by a means which would make them invisible to acoustic waves. It has, however, been pointed out that this form of inactive cloaking could cover a target in a way that shields the target from using its own acoustic sonar means for detecting the source, unless the cloaking could be turned off. We address this issue by presenting an active cloaking transducer system which effectively cloaks the target and yet can also be used as an acoustic sonar system. Equivalent circuits and finite element models are used to demonstrate transducer cloaking. This would have been a good transducer project for Kim Benjamin to implement and I believe he would have enjoyed the challenge and developed one of the best cloaking transducer arrays possible.

Session 2aMU

Musical Acoustics: Where Are They Now? Past Student Paper Award Winners Report

James P. Cottingham, Chair
Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402

Invited Papers

9:40

2aMU1. Where they are, where they have been, and where they are going. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

The ASA Best Student Paper Awards competition began in 1997. From 1997 to 2001, one award winner in Musical Acoustics was selected at each meeting, but beginning in 2002, two awards have been given, with first and second prize winners selected. In all there have been 52 award winners in Musical Acoustics, including three who won an award twice. Some are still active in musical acoustics, but many others are now active in other areas of acoustics or in fields outside acoustics altogether. A brief overview will be presented of the history of the competition and past and current interests of those who have been the award winners. The speakers in this session include winners of the award since the last Providence meeting in 2006. Capsule updates on several award winners who are unable to participate in this session will be presented.

10:00

2aMU2. From musical acoustics to outdoor sound and back. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu)

My musical acoustics student paper award was given in 2009 at ASA San Antonio after an NSF summer research experience at Coe College with Dr. James Cottingham. I was a junior at Murray State University in Kentucky studying the clarinet and mathematics. Though I was not the most qualified applicant, lacking the physics background necessary in this field, I had a passion for acoustics and wanted my future to go in this direction. This summer was my introduction to acoustics research and the reason I was able to continue in the field. Since, I have been attending Penn State in the Graduate Program in Acoustics and have earned my Masters in Acoustics focusing on outdoor sound propagation modeling and now, with the help of an NSF-GRFP, I have found my way back to musical acoustics—clarinet acoustics. Since the San Antonio ASA I have attended seven more ASA conferences, been the musical acoustics student council representative and am now the student council chair. I now split my time between Penn State and Marseille, France, at CNRS-LMA. This talk will detail each step along the path that retuned me to musical acoustics and give a look into my current research as well.

10:20

2aMU3. A systems engineering approach to musical acoustics. Nicholas J. Eyring (Dept. of Phys. and Astronomy, Brigham Young Univ., 2353 W 1700 North, Provo, UT 84604, eyringj@gmail.com)

Research in musical acoustics has benefited greatly from advances in technology; however, the ever increasing complexity of measurement systems may lend to inefficient experimental design and procedure. A systems engineering approach to experimentation can improve the process. Systems engineering involves the design of complex, many element, systems that maximize overall performance, considering all elements related in any way to the system, including human efficiency and the characteristics of each of the system's components. This paper explores how a background in the full product development lifecycle of a Raman spectroscopy based measurement instrument allowed for the rapid development of an automated directivity acquisition system (ADAS) used to measure a concert grand piano. Prior experience also assisted in adapting the ADAS to measure musicians when assessing elements like reliability, logistics, work-processes, and optimization methods. An account of how research in musical acoustics provides applicable experience for employment in a non-acoustics industry position will also be given.

10:40

2aMU4. The science of art, the art of science. Rohan Krishnamurthy (Musicology, Eastman School of Music, 544 Sunrise Circle, Kalamazoo, Michigan 49009, rohan.krishnamurthy@rochester.edu)

I will discuss how my passion for the arts and sciences originated in and developed since elementary school, and led me to my present work as a professional percussionist, educator, researcher, and entrepreneur. Throughout high school, I pursued music and science projects in parallel. My dual interests encouraged me to pursue a double major in music and chemistry at Kalamazoo College. My senior thesis introduced me to acoustical research when I worked with Dr. James Cottingham at Coe College to study the acoustics of a new drum tuning system that I invented and patented. I presented my research at ASA 2007 in New Orleans and other research at subsequent

ASA conferences. After finishing college, I pursued a doctorate in musicology at the Eastman School of Music at the University of Rochester. I will discuss specific examples of my past and current interdisciplinary projects, and how my association with the ASA continues to inspire my endeavors.

11:00

2aMU5. Categorization and lexicon in verbal descriptions of violin quality by performers. Charalampos Saitis (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke Str. West, Montreal, QC H3A 1E3, Canada, charalampos.saitis@mail.mcgill.ca), Claudia Fritz (Lutheries-Acoustique-Musique, Inst. Jean le Rond d'Alembert, Université Pierre et Marie Curie, UMR CNRS 7190, Paris, France), and Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montreal, QC, Canada)

This work aimed to explore how violin quality is conceptualized as reflected in spontaneous verbal descriptions by experienced performers collected while playing in a perceptual evaluation experiment. Participants performed a preference ranking task and justified their perceptions in a free verbalization task. Using the constant comparison analysis from grounded theory, a concept map was developed, which can be useful for future studies aimed at assessing violin qualities. A psycholinguistic analysis of the quality-descriptive lexicon used by violinists further revealed a variety of linguistic devices referring to either the sound of a violin or to the violin itself as the cognitive objects. Adjectives for the description of sound characteristics are largely borrowed from four semantic fields related to texture-temperature (smooth vs. rough), action-presence (resonant vs. muted), size-volume (deep vs. flat), and light (dark vs. bright). These semantic fields indicate what type of dimensions may explain the perception of violin timbre, contributing to the area of violin acoustics research as well as to the broader area of timbre research. Some acoustical interpretations are discussed in the context of finding correlations between measurable vibrational properties of a violin and its perceived quality.

11:20

2aMU6. Robotics, free-reed instruments, and naughty birds: Finding the common thread. Eric A. Dieckman (Sonalysts, Inc., 84 Nicoll St., Unit 1, New Haven, CT 06511, eric.dieckman@gmail.com)

Musical acoustics often provides an accessible starting point for undergraduate research in which the basics are learned and applied to interesting problems. Even if students later study other areas of acoustics, the research experience gained is invaluable. This presentation will touch on a number of research projects encountered since the author's 2006 award for investigations of the acoustic behavior of Southeast Asian free-reed mouth organs. These projects come from a wide variety of acoustic disciplines, from nondestructive evaluation and architectural acoustics to benign bird exclusion and acoustic sensors for mobile robots, with the common thread of signal processing providing a focal point.

11:40

2aMU7. An experience-based approach to auditory perception research. Brian B. Monson (Dept. of Newborn Medicine, Brigham and Women's Hospital, Harvard Med. School, 75 Francis St., Boston, MA 02115, bbmonson@email.arizona.edu)

What and how we hear is determined by our past experience. Thus attempting to quantify human experience becomes the challenge for modeling auditory perception. Based on my recent research at the Duke-NUS Graduate Medical School in Singapore, I will discuss some principles that should guide such an approach to study of auditory neuroscience and perception. One crucial principle is to account for the frequency of occurrence of stimulus patterns to which humans have been exposed over phylogeny and ontogeny (an alternative to explaining perception based solely on peripheral auditory physiology). I will discuss the implications of this research approach and include a brief report on my research progress in my new position at Brigham and Women's Hospital.

Session 2aNSa

Noise: Session in Honor of Kenneth Eldred

Louis C. Sutherland, Cochair
lcs-acoustics, 5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Chair's Introduction—8:55

Invited Papers

9:00

2aNSa1. Kenneth McKechnie Eldred—A distinguished noise control engineer I. William W. Lang (Noise Control Foundation, 29 Hornbeck Ridge, Poughkeepsie, NY 12603, lang1ww@gmail.com) and George C. Maling (NAE Member, Harpswell, ME)

Elected in mid-career in 1975 to the U.S. National Academy of Engineering (NAE) "For outstanding accomplishments in noise and vibration control of air, space, and transportation vehicles and in delineating acceptable noise environments for people." His NAE peers recognized him "...as one of the five best noise and vibration control engineers in the country." Ken's first job after graduating from M.I.T. in 1950 with studies of advanced courses in acoustics was as head of the Boston Naval Shipyard's lab working on the reduction of the noise and vibrations of submarine auxiliary equipment. In 1953 on active duty in the U.S. Air Force, he became Chief of the Bio-Acoustics Branch, Wright Air Development Center. In 1957, he moved to California for a career with Western Electro-Acoustic Laboratory and Wyle Laboratories prior to joining Bolt Beranek and Newman. This paper is primarily devoted to his activities in support of INCE/USA, his role in the passage of the Noise Control Act of 1972, and his activities in connection with the Office of Noise Abatement and Control in the U.S. Environmental Protection Agency.

9:20

2aNSa2. Kenneth McKechnie Eldred—A distinguished noise control engineer II. George C. Maling (NAE Member, 60 High Head Rd., Harpswell, ME 04079, maling@alum.mit.edu) and Eric W. Wood (Acentech Inc., Cambridge, MA)

Ken's accomplishments in noise control engineering cover the spectrum from basic engineering research to recommendations for noise control in six areas: measurement of industrial noise; measurement and reduction of structural vibration in space vehicles; noise radiation from jet flow; noise reduction of jets by multiple nozzles and turbofans; vibroacoustic environmental simulation for aerospace vehicles; and community and transportation noise control. Some of these areas are discussed in consulting reports, and others in papers published in the open literature. He was a consultant at Bolt Beranek and Newman from 1973 until 1982 when he formed Ken Eldred Engineering. His publications appeared in the journal NOISE Control, several NOISE-CON and INTER-NOISE Proceedings, *Noise Control Engineering Journal*, and the *Journal of the Acoustical Society of America*. Representative samples of his accomplishments taken from the above sources will be presented.

9:40

2aNSa3. Ken Eldred—Mentor. Richard Potter (Retired, 129 Wilder Dr., Harvest, AL 35749, dickpotter@bellsouth.net)

I met Ken Eldred at Wyle Laboratories in Alabama in 1963 as a young, green engineer shortly after he formed a staff to support NASA's Apollo program. Later, I was joined by other graduates of the Institute for Sound and Vibration Research, Southampton University, England. Ken led us as we undertook exciting and challenging projects. At Wyle, Ken actively mentored us, encouraging us to develop our investigative skills, adapt to new technologies, and write understandable, readable, and useful reports. He encouraged me to produce ideas and he listened to my suggestions, which he carefully reviewed and then, gently, offered corrections and suggested direction. I, and many of us, owe our successful careers to his mentoring. I will describe some particular work and list some who prospered under his leadership. I moved back to England but then, later, returned to the United States and rejoined Ken at Bolt Beranek and Newman in Cambridge. Again he showed interest and encouragement as I expanded my noise work to other aspects of industrial hygiene, later forming my own companies.

10:00

2aNSa4. Tribute to Kenneth McKechnie Eldred. Louis C. Sutherland (lcs-acoustics, 5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275, lou-sutherland@juno.com)

I first met Ken while at the Boeing Co and he was Vice President at the Western Electro-Acoustics Laboratory in Los Angeles. I met him again at Wright Patterson Air Force Base, Dayton, Ohio, where he was Chief of Physical Acoustics under Henning von Gierke. He recruited me to come to Huntsville, Alabama, to join the new branch of Wyle that Marshall Space Flight Center wanted for rocket noise programs supporting NASA's rocket noise programs. To augment this Wyle staff, Ken recruited several outstanding acoustical scientists from Southampton University in the UK, including the late Martin Lowson. More on this in Richard Potter's paper. Ken's Wyle staff worked with the Federal Aviation Administration, the Boeing Co. and Lockheed Aircraft in pursuing the environmentally failed development of the SST. Ken supported the U.S. Environmental Protection Agency Office of Noise Abatement and Control (ONAC) and guided Wyle in their preparation of key documents on Noise Policy. He left Wyle to join Bolt, Beranek and Newman (BBN) and later left BBN to form Ken Eldred Engineering. He was a Fellow of ASA, received the ASA Silver Medal in Noise in 1994, and was active on ASA Standards committees, the National Research Council, the Society of Automotive Engineers, and the National Academy of Sciences. The other speakers will discuss other aspects of Ken's many contributions, including those for the Institute of Noise Control Engineering (INCE).

TUESDAY MORNING, 6 MAY 2014 — 557, 10:35 A.M. TO 12:00 NOON

Session 2aNSb

Noise: Session in Honor of Harvey Hubbard

Louis C. Sutherland, Cochair
lcs-acoustics, 5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Chair's Introduction—10:35

Invited Papers

10:40

2aNSb1. The administrative leadership of Harvey Hubbard. Charles E. Schmid (ASA, 365 Ericksen Ave., Bainbridge Island, WA 98110, ceschmid@att.net)

Anyone who met Harvey Hubbard knew right away that he was a gentleman. His unassuming manner belied the long list of his accomplishments from the time he left a one-room school in Franklin County, Vermont to his death at age 90 in Newport News, Virginia, in 2012. His endeavors were both technical and administrative, but only the latter will be covered in this presentation, leaving the technical topics to other speakers. The focus will first be on his presidency of the Acoustical Society of America (1989–90). His leadership in selecting and hiring the Society's first executive director (which directly impacted this author) was carried out with his usual professional graciousness. A number of other innovations which occurred during his presidency will be described, such as the establishment of an Investment Committee, which has become vital to the Society's economic well-being. He held many managerial positions in other non-profits, for-profits, and governmental organizations, which will be mentioned to fully understand the breadth of his managerial contributions. [The author would like to acknowledge Elaine Moran for tracking down much of the information which will be presented.]

11:00

2aNSb2. Harvey H. Hubbard and his contributions to wind turbine noise (among other things). Kevin P. Shepherd (NASA Langley Res. Ctr., 2 N. Dryden St., Hampton, VA 23681, k.p.shepherd@nasa.gov)

Following his long NASA career devoted primarily to the understanding and reduction of aircraft noise, Harvey Hubbard came out of retirement to pursue an interesting opportunity that was presented by issues concerning the sound from large wind turbines. This paper will attempt to summarize this pioneering work on wind turbine noise, along with some other accomplishments and recollections.

11:20

2aNSb3. Harvey H. Hubbard's contributions to aircraft noise control during his NACA-NASA career. Domenic J. Maglieri (Eagle Aeronautics, Inc., 732 Thimble Shoals Blvd.'Bldg. C 204, Newport News, VA 23606, sonicboomexpert1@verizon.net)

Following his service in the US Army Air Corps during World War ll, Harvey accepted a position at the NACA Langley Memorial Aeronautical Laboratory in 1945. Propeller aircraft dominated the air transport system at that time and Harvey became one of the first to perform research on the noise associated with propellers. In the next decade jet engine powered aircraft made their appearance and they became the focus of a rapidly growing acoustics research program at Langley. Harvey's pioneering experimental noise studies on propellers and jet engines provided significant insight and understanding of these two concerns. Foreseeing that additional research efforts would be required to address the many new aircraft noise issues, including the sonic boom and airport-community noise concerns of the proposed U.S. supersonic transport, he played a key role in getting NASA to expand its acoustic efforts. As a result, Harvey became NASA's technical focal point for all major acoustical activities. This paper will show the acoustic activities that Harvey was involved in and present some highlights of his research on propeller noise, jet noise, and sonic boom.

11:40

2aNSb4. Harvey Hubbard and the Acoustical Society of America oral histories project. Victor Sparrow (Grad. Prog. Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

Harvey Hubbard was an amazing person who contributed substantially to both the profession of acoustics and to the Acoustical Society of America. This short talk is the story of Harvey Hubbard's ASA oral history. It was a great honor to work with Harvey on his oral history. The transcript of the interview is currently available online through the American Institute of Physics' Center for History of Physics and the Niels Bohr Library and Archives. The interviewer for that oral history will recount some of the highlights from Harvey giving that interview and from some of the superlative life experiences he recounted. Everyone is encouraged to participate in the collection of similar oral histories for the preservation of the history of acoustics and the history of the ASA.

TUESDAY MORNING, 6 MAY 2014 551 A/B, 7:55 A.M. TO 10:10 A.M.

Session 2aPA

Physical Acoustics: Beyond Basic Crystals: Viscoelastic and Piezoelectric Materials

Julian D. Maynard, Cochair
Phys., Penn State Univ., 104 Davey Lab., Box 231, University Park, PA 16802

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

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Resonance ultrasound spectroscopy for studying piezoelectricity and internal friction at elevated temperatures of quartz, langasite, and gallium nitride. Hirotsugu Ogi (Eng. Sci., Graduate School of Eng. Sci., Osaka Univ., Toyonaka, Osaka 560-8531, Japan, ogi@me.es.osaka-u.ac.jp) and Hassel Ledbetter (Mech. Eng., Univ. of Colorado Boulder, Boulder, CO)

Resonance ultrasound spectroscopy (RUS) is a powerful method for measuring elastic constants (Cijkl) of solids. It can be applied to determine the piezoelectric coefficients (eijk) as well, because they also affect the mechanical resonance frequencies. Precise frequency measurements in vacuum and unambiguous mode identification with laser-Doppler interferometry allowed us to determine Cijkl and eijk simultaneously for crystals including alpha quartz, which shows low piezoelectricity. We further developed a noncontact excitation and detection method with antennas through electromagnetic fields and study internal friction and carrier mobility of quartz, langasite, and GaN at elevated temperatures up to ~1200 K. This method is also successfully applied to developing ultrahigh-sensitive biosensors for diagnosis.

8:20

2aPA2. Measuring viscoelasticity of soft tissues using shear and guided waves. Matthew W. Urban, Carolina Amador, Ivan Z. Nenadic, Heng Zhao, Shigao Chen, and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Elasticity imaging has emerged as a viable clinical tool for assisting in diagnosis of certain diseases such as liver fibrosis and cancer. Many methods have been developed for generating and measuring shear waves in soft tissues. The advantage of using shear waves is that the shear wave velocity is proportional to the mechanical material properties of the tissues under investigation. Soft tissues are inherently viscoelastic and considerable effort has been made to quantify the dispersion, or variation of frequency, of the velocity and attenuation of shear waves. This is accomplished by measuring the shear wave motion and using Fourier-based techniques to extract the shear wave velocity and attenuation. Additional considerations are made in tissues where geometric dispersion is also present such as in the heart. Viscoelastic characterization of the shear viscoelasticity of soft tissues *in vivo* such as human liver, human kidney, and swine heart will be shown. We will also demonstrate parameterization of the results by using a model-free approach or by fitting the shear wave velocity dispersion to rheological models. The diagnostic value of the viscoelastic parameters will be discussed for each particular application. [This work was supported in part by NIH Grant Nos. DK092255, DK082408, and EB002167.]

Contributed Papers

8:40

2aPA3. Nonlinear surface acoustic waves on lithium niobate in microfluidic devices. Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Surface acoustic waves (SAW) are used frequently in microfluidic devices. Normally SAWs are generated on the surface of a piezoelectric material. Commonly used PZT is not appropriate for biomedical applications because of its high lead content, over 60% by weight. In this talk, a study of nonlinear SAW propagation in a piezoelectric substrate is presented. Model equations describing nonlinear SAW propagation in a piezoelectric crystal are derived from first principles. Elastic, piezoelectric, dielectric, and electrostrictive properties of a crystal with arbitrary symmetry are taken into account. The derived evolution equations are integrated numerically to illustrate nonlinear distortion of an initially sinusoidal wave of finite amplitude. As an example, SAW propagation along the X axis on single crystal 127.680 YX-cut lithium niobate ($LiNbO_3$), referred to as 128-YX-LN, is considered. This $LiNbO_3$ cut is typically used in microfluidic devices because it provides large mechanical displacements in the substrate. Analysis of the nonlinearity matrix permits quantification of the relative contributions to surface wave distortion from each physical phenomenon. [Work supported by the IR&D program at ARL:UT.]

8:55

2aPA4. Shear wave propagation in worm-like micellar fluids. Josh R. Gladden, Rachel Crim, Amanda Gamble, and Cecille Labuda (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

In viscous Newtonian fluids, support of shear waves are limited to the viscous boundary layer. Non-Newtonian fluids which have shear modulus, however, support shear waves over much longer distances. The restoring force responsible for the shear wave propagation arises from the entanglement of high aspect ratio macromolecules. We report low frequency (30–60 Hz) shear wave studies of aqueous worm-like micellar fluids composed of cetyltrimethylammonium bromide (CTAB) for the surfactant and sodium salicylate (NaSAL) as the salt over a wide concentration range (20–500 mM CTAB). Shear speeds range from 75 to 700 mm/s over this concentration range at room temperature with evidence of two phase transitions at 200 mM and 375 mM CTAB. Shear stress attenuation and temperature resolved measurements between 20 and 40 C will also be presented.

9:10

2aPA5. Using resonant ultrasound spectroscopy on samples with and without conducting coatings to measure piezoelectric constants. Rhiannon E. Vieceli and Julian D. Maynard (Phys., Penn State Univ., 104 Davey Lab., University Park, PA 16802, rev5028@gmail.com)

Piezoelectrics are often used at low temperatures, but among the large number of piezoelectric materials, only one (quartz) has had its properties measured at low temperatures. Because measuring piezoelectric constants with the traditional electrical impedance method has shortcomings, particularly at liquid helium temperatures, it would be advantageous to make measurements with resonant ultrasound spectroscopy (RUS). EerNisse and Holland (1967) established a theoretical basis and Ogi *et al.* (2002) demonstrated an experimental RUS method. A problem with RUS for piezoelectrics is that resonance frequencies are much more sensitive to elastic behavior than to piezoelectric behavior, so that extraordinary precision is required. However, one may make a RUS measurement on a sample twice, once with an electrically conducting coating on sample faces and once without, and analyze the differences in the frequency spectra. Because the effect of the conducting coating depends more on the piezoelectric behavior than on the elastic behavior, analyzing the frequency differences suppresses the dependence on the elastic constants and enhances the measurement of the piezoelectric constants. This paper will present theoretical and experimental results for this RUS method.

9:25

2aPA6. Measurement of dispersion and attenuation in granular media using a filter-correlation method. Caleb O'Connor and Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA, cso5597@louisiana.edu)

A wideband technique for measuring sound dispersion and frequency-dependent attenuation in granular media is presented. The measurements were done on a mono-disperse medium of 2-cm solid polypropylene balls, over the frequency range of 500 Hz–20 kHz, enough to cover both weak- and strong-scattering regimes. A horn driver was used to launch sound into a foam-lined bucket containing the granular medium. The latter was mechanically isolated from the driver so as to minimize direct-contact coupling. The foam isolation was not enough, especially at resonances of the bucket-granular system. To account for the mass loading of the bucket by the granulars, the response of the bucket wall was measured by laser Doppler vibrometry both without and with the granulars. The response of the granular medium itself was extracted from the overall response through successive measurements of the individual responses of the driver, driver + bucket, and driver + bucket + granular. The frequency-dependent wavenumber of the granular is obtained by a filter-correlation method, using the driver response as reference. After successive bandpass filtering, the phase speed and attenuation are obtained within each band, respectively, by signal alignment and amplitude log ratio.

9:40

2aPA7. Design of piezoelectric energy harvesting system using cantilever beam. Jin-Su Kim, Un-Chang Jeong, Sun-Hoon Lee, Jung-Min Jeong, and Jae-Eung Oh (Mech. Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, fermatajin@hanyang.ac.kr)

In this paper, a design for an energy harvesting device using cantilever beam will be investigated and experimental results will be presented to validate the design. The energy harvesting device in the study is 31-unimorph

piezoelectric which was used to convert small amplitude mechanical vibration from a specific machine application into an electrical energy source that could be used for electronic devices with low power requirements. The primary purpose of the design is to illustrate a method to design a cantilever beam that is optimized for attached position of piezoelectric by Experiment and FEM. From the given vibration data a range of frequencies where the energy harvesting device will generate the greatest amount of energy is determined. The device is then designed specifically targeting that frequency range with sinusoidal wave about resonant frequency. And results of this study show the change trend of output voltage according to changing circuit elements. This approach is presented as part of a more general approach to designing energy harvesters for any application. Also, it will be shown how attached position of piezoelectric used for cantilever beam were chosen.

9:55

2aPA8. *In-situ* ultrasonic evaluation of structural/nuclear materials. K. Sakthipandi (Phys., Sethu Inst. of Technol., Pullor, Kariapatti, Tamil Nadu 626115, India, sakthipandi@gmail.com) and V. Rajendran (Ctr. for Nano Sci. and Technol., K S Rangasamy College of Technol., Namakkal, Tamil Nadu, India)

Physical properties of components through ultrasonic non-destructive evaluation play a vital role to understand in quality and strength materials and also help to extend life of the components. Measurements of ultrasonic velocity and attenuation as a function of temperature were used to reveal the structural/phase transitions, initiation and growth of fatigue-induced damages, and life-limiting fatigue crack during the aging of materials. Indigenously designed experimental set-ups was designed for *in-situ* ultrasonic velocities and attenuation measurement over a wide range of temperature from 120 to 300 K and 300 to 1200 K. The ultrasonic velocity/attenuation measurements carried out on AISI316 stainless steel, β-quenched zircaloy-2 specimen and maraging steel were used to explore the formation and resolution/recrystallization of intermetallic and coherent precipitations. Further, the ultrasonic velocity/attenuation measurements carried out in bulk and nano perovskites samples ($La_{1x}Sr_xMnO_3$, $Nd_{1x}Sr_xMnO_3$, $Sm_{1x}Sr_xMnO_3$ and $Pr_{1x}Sr_xMnO_3$) were used to explore the phase transition (TC), charge ordering (TCO), and Jahn-Teller (TJT) temperature. The bulk and nanocrystalline nature of the perovskites were explained based on observed anomalies at transition temperature. The plot of first derivative of temperature dependent ultrasonic parameters was used to reveal the precise information to detect the early stages of microstructural and substructure variations in material.

TUESDAY MORNING, 6 MAY 2014 BALLROOM B, 8:30 A.M. TO 11:25 A.M.

Session 2aPP

Psychological and Physiological Acoustics: Temporal Processing, Compression, and Cochlear Implants: Session in Honor of Sid P. Bacon

Neal F. Viemeister, Cochair
Psychology, Univ. of Minnesota, 75 E. River Pkwy, Minneapolis, MN 55455

Walt Jesteadt, Cochair
Boys Town National Res. Hospital, 444 N. 30th St., Omaha, NE 68131

Chair's Introduction—8:30

Invited Papers

8:35

2aPP1. Modulation masking within and across carriers for subjects with normal and impaired hearing. Brian C. Moore and Thomas Baer (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Sid Bacon was a pioneer in studies of the extent to which the detection of target amplitude modulation (AM) of a carrier is affected by additional (masker) amplitude modulation applied to the same carrier (within-channel modulation masking) or to a different carrier (across-channel modulation masking). Here, these two types of modulation masking were compared for normal-hearing and hearing-impaired subjects. The target was either 4-Hz or 16-Hz sinusoidal AM of a 4000-Hz carrier. The target AM depth was fixed. The masker AM was applied either to the same carrier or to a carrier at 3179 or 2518 Hz. The masker AM rate was 0.25, 0.5, 1, 2, or 4 times the target rate. The masker AM depth was varied adaptively to determine the value needed just to mask the target AM. Preliminary results indicate that within-channel modulation-masking patterns are similar for normal-hearing and hearing-impaired subjects, suggesting that the hypothetical modulation filters are not affected by hearing loss. However, the amount of across-channel modulation masking is lower for normal-hearing than for hearing-impaired subjects, presumably because of the reduced frequency selectivity of the latter. The increased across-channel masking for the hearing-impaired subjects may contribute to their difficulties in understanding speech in background sounds.

9:00

2aPP2. Possible contribution of cochlear compression to amplitude modulation detection. Jungmee Lee (Dept. of Commun. Sci. and Disord., Univ. Wisconsin, 475 Goodnight Hall, 1975 Willow Dr., Madison, WI 53706, jmlee6@msn.com)

Among his many research areas, Sid Bacon's work on auditory temporal processing, and his effort to connect psychophysical phenomena with cochlear compression, contributed greatly to our understanding of auditory system function. Inspired by his work, I will present research suggesting that (1) temporal processing, as measured by amplitude-modulation (AM) detection, is better for people with cochlear hearing impairment (HI) than those with normal hearing (NH) when the age of groups and loudness of the stimuli is matched, (2) AM detection of a target is poorer when an AM masker is presented at frequencies in a region with hearing loss than in a region of normal hearing, (3) masking of AM detection by an AM masker is reduced in HI when the masker is compressed using a low-distortion compressor, but is unaffected for NH, and (4) amplitude modulation of 2f1-f2, Distortion Product Otoacoustic Emissions recorded with amplitude-modulated f1 and steady-state f2, is correlated with AM perception. Taken together, the results provide evidence that cochlear compression plays an important role in auditory temporal processing as measured by AM perception.

9:25

2aPP3. The role of pitch strength in extracting speech from complex backgrounds. Marjorie R. Leek (Res. Dept., VA Loma Linda Healthcare System, 11201 Benton St., Loma Linda, CA 92357, Leekmar@aol.com)

How do people extract a target speech signal from a chorus of other sounds including other speech sounds? And what are the central and peripheral auditory processes that make this possible, and that may fail in people with damaged auditory systems? These critical issues in speech and hearing science were of continuing interest to Sid Bacon throughout his career. As Sid has noted, the problem is multifaceted, and he chose to study many of the individual factors, as well as their interactions. In keeping with that approach, I will discuss how normal-hearing and hearing-impaired listeners use pitch to extract a target from a complex noise background. In one study, auditory stream segregation of iterated rippled noises (IRN) with varying pitch strengths was explored to understand the limits of tonality for separating two patterns of sounds. In a second study, voice pitch strength was investigated as a means to support perceptual separation of target and background speech, with a focus on either spectral or temporal characteristics of the speech sounds. Interactions between degree of tonality in speech and other factors related to perception of speech in background sound will be examined. [Work supported by NIH.]

9:50–10:05 Break

Chair's Introduction—10:05

10:10

2aPP4. Role of temporal fine structure in speech recognition: From psychoacoustics to cochlear implants. Frederic Apoux and Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

The path from fundamental research to real-world application is often long and sinuous. This presentation will depict how psychoacoustic studies performed in Sid Bacon's laboratory have engendered our original view of the role of temporal fine structure (TFS) in speech recognition, and how this view has in turn led to the development of an extremely effective speech-processing strategy for cochlear implants. First, psychoacoustic data showing how normal-hearing listeners can detect amplitude modulation when presented with only the TFS will be described. Second, a series of experiments illustrating the "absence" of acoustic speech cues in the TFS will be briefly presented. Finally, the results of a recent study involving a dual-carrier vocoder will be described. These results suggest that TFS cues are primarily used to assist in identifying which auditory channels are dominated by the target signal so that the output of these channels can be combined at a later stage to reconstruct the internal representation of that target. They also indicate that cochlear implants implementing a speech-processing strategy based on the "dual-carrier strategy" have the potential to restore nearly perfect speech intelligibility in noise. [Work supported by NIH.]

10:35

2aPP5. Understanding the benefits of electric-acoustic stimulation. Christopher Brown (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260, cbrown1@pitt.edu)

The problem of speech understanding in the presence of background noise is a difficult one, especially for users of cochlear implants. Although these users can often perform well on speech tasks in quiet, they typically show rapid declines in background noise. The retention of low-frequency residual acoustic hearing has been shown to provide significant benefit in this regard. Data will be presented on our work on this topic, from characterizing the benefit to exploring ways of providing it to cochlear implant users who do not show a benefit typically.

11:00

2aPP6. Spectral resolution and its effects on temporal analysis in cochlear-implant perception. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Sid Bacon's contributions to auditory science span a wide range of topics but two areas were constant themes throughout his career: temporal modulation perception, and peripheral processing and frequency selectivity. In this study, speech understanding in noise was measured, with a focus on the role of inherent temporal fluctuations in noise maskers. Tonal maskers, presented to cochlear-implant users, were placed at the center frequencies of each frequency channel of the implant, thereby producing the same masker energy as a noise masker in each frequency channel, but without the inherent fluctuations. In contrast to the results from normal-hearing subjects listening through a tone-excited envelope vocoder, cochlear-implant users gained no benefit from eliminating the inherent fluctuations from the maskers. Further experiments suggested that the poor spectral resolution of cochlear implants resulted in a smoothing of the temporal envelope of the noise maskers. The results indicate an important, and potentially overlooked, effect of spectral resolution on the temporal representations of speech and noise in cochlear implants. The results also suggest a new interpretation for why cochlear-implant users, and perhaps hearing-impaired listeners, generally show reduced masking release when additional temporal modulations are imposed on noise maskers. [Work supported by NIH grant R01DC012262.]

TUESDAY MORNING, 6 MAY 2014 553 A/B, 8:20 A.M. TO 11:30 A.M.

Session 2aSA

Structural Acoustics and Vibration, Physical Acoustics, Engineering Acoustics, and Noise: Acoustic Metamaterials I

Christina J. Naify, Cochair
Acoust., Naval Res. Lab., 4555 Overlook Ave. SW, Bldg. 2, 138G, Washington, DC 20375

Michael R. Haberman, Cochair
Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

8:20

2aSA1. Metal based acoustic metamaterials. Andrew Norris, Adam J. Nagy, and Alexey S. Titovich (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Two apparently distinct types of acoustic metamaterial are considered: a metallic phononic lattice structure and an array of metal shells in water. The unifying feature is that metal acts primarily as a stiffness, and by adding material one arrives at a desired effective density. The metal provides a reservoir of stiffness in the sense that a little bit goes a long way toward the effective stiffness of water, or properties close to water. We first describe the Metal Water structure proposed as a generic metamaterial for transformation acoustics, in both 2D and 3D. The structures have isotropic elastic properties with low shear modulus, hence mimicking water. While designed for long-wavelength effective properties the structures also display interesting finite frequency effects, such as negative index properties. The thin shell metamaterial elements achieve the bulk modulus of water at a specific thickness/radius ratio. Simultaneous matching of effective bulk modulus and density is obtained using an internal mass. By design, both types of metamaterials separate stiffness and density, allowing for simple lumped parameters modeling. The use of metal also has implications for optimal cloaking properties, which result from the fact the metallic structure is non-causal. [Work supported by ONR.]

8:40

2aSA2. Transparent acoustic metamaterials for broadband aqueous applications. Theodore P. Martin (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, theodore.martin@nrl.navy.mil), Christina J. Naify, Christopher N. Layman, Charles A. Rohde (National Res. Council, Washington, DC), Abel Thangawng, Michael Nicholas, David C. Calvo, and Gregory J. Orris (Naval Res. Lab., Washington, DC)

Matched impedance between different components is an integral part of many wave transport device applications. Achieving transparency in acoustics has typically been accomplished using bulk materials such as ρc rubbers or resonance phenomena over narrow bandwidths. This talk will focus on broadband acoustic metamaterials that are impedance matched to water and that offer a broad range of locally definable sound speeds. Experimental investigations of a number of device applications will be presented, including a transparent gradient index (GRIN) lens, an omnidirectional focusing coating, and elastic lattices that mimic the material properties of water. Impedance-matching over a wide range of sound speeds is achieved by changing the filling fraction of sub-wavelength acoustic scattering components that are individually impedance-matched to water. Both sonic crystals and pentamode trusses are explored as component lattices to produce fluid-like transport over a broad bandwidth in the homogenization limit of the lattices. Excellent agreement is obtained between predictions using fully elastic homogenization theory and the acoustic multistatic signatures measured in the vicinity of the as-realized devices. [Work supported by the Office of Naval Research.]

9:00

2aSA3. From acoustic metamaterials to functional metasurfaces. Nicholas X. Fang, Jun Xu, Chu Ma, and Navid Nemati (MechE, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, nicfang@mit.edu)

By guiding and controlling the wave path through a deformed space, metamaterial devices that display distinctive response to light, acoustic waves, and heat waves have opened up a new field of considerable interest. However, current challenge issues such as high loss and complex geometries are hampering the development of metamaterial technology. Is it possible to tailor the field information

contained by a complex volumetric object with a thin metamaterial structure, so the observer from afar would not tell the difference? In this invited talk, we will present our research progress toward tailoring the edge rays and creeping rays with acoustic metasurfaces. The tailored metasurface could lead to two pronounced effects: redirecting the reflected rays by spatial impedance gradient and reducing the strength of the edge rays. In fact our recent study suggest such illusional effects by embedding wedges in a transformed acoustic medium, and we will present theoretical analysis and experimental study of such engineered components with acoustic metasurface. The potential application of such novel device concept in underwater communication and medical ultrasound will be also discussed.

9:20

2aSA4. Pressure-invariant non-reflective and highly dissipative acoustic metamaterials. Alireza Amirkhizi (Mech. Eng., Univ. of Massachusetts, Lowell, Perry Hall 331, 197 Riverside St., Lowell, MA 01854, alireza_amirkhizi@uml.edu), Christian Nielsen, Zhanzhan Jia, Wiroj Nantasetphong, Hossein Sadeghi, Kristin Holzworth (Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA), Ankit Srivastava (Mech., Mater., and Aerosp. Eng., Illinois Inst. of Technol., Chicago, IL), and Sia Nemat-Nasser (Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA)

Metamaterials have shown great potential to transform the design of acoustic components in many applications. Many composites with extreme properties have been envisioned, designed, fabricated, and experimentally verified. The next step involves testing such composites in realistic application environments. Oceans are one such environment in which the mechanical effects of water pressure and flow become important factors in any acoustic design, particularly for soft dissipative shells. We have designed a layered metamaterial composite that not only shows very high dissipation but also matches the acoustic impedance of water. Furthermore, we have experimentally verified that the relevant properties of the constituents of this layered design do not change under pressure levels that exist down to significant depths. We are in the process of fabricating this composite to test its acoustic properties under pressure. The metamaterial composites lend themselves naturally to multi-component designs, examples of which as well as gradient media will be presented. Some potential novel applications of gradient and layered components will be discussed.

9:40

2aSA5. Wave propagation in three dimensional crystalline foams. Alessandro Spadoni (Mech. Eng., EPFL, STI-IGM-LOMI, Station 9, Lausanne, 1015 Lausanne, Switzerland, alex.spadoni@epfl.ch)

Recent progress in manufacturing of crystalline foams has introduced cellular solids with very low relative density, the portion of volume occupied by the solid phase. For closed-cell configurations, this means thin films enclosing entrained fluid. While numerous mechanical models for the mechanical properties of 2D, open-cell configurations have been proposed, 3D closed-cell configurations are described by phenomenological models based on powers of the relative density due to their complexity. Elastic wave propagation in such media presents similar challenges and is described by Biot's theory, a model derived from a strain-energy functional defined at the macroscale, based on averaged microstructural quantities. Biot's theory requires equivalent mechanical properties for drained and undrained configurations which are often not available. Exploiting periodicity, we developed a detailed finite-element model to explicitly describe the coupling of fluid and solid. The entrained fluid is compressible, inviscid, and both convection and heat conduction are neglected. Three crystalline foams are considered: Kelvin, rhombic, and Weaire-Phelan configurations. In this talk, I will discuss frequency regimes with a single and two longitudinal pressure wavemodes, and super anisotropy. Dispersion depends on two key frequencies: film resonant frequency, and the natural frequency of a pore with deformable walls.

10:00–10:30 Break

Contributed Papers

10:30

2aSA6. Tapered labyrinthine acoustic metamaterials for coherent controlling of acoustic wave. Yangbo Xie, Adam Konneker, Bogdan-Ioan Popa, and Steven A. Cummer (Duke Univ., 3417 CIEMAS, Durham, NC 27705, yx35@duke.edu)

Acoustic metamaterials with their exotic material properties enable unprecedented control over acoustic wave propagation and reflection. Besides utilizing locally resonating structures or non-resonant composite effective media, non-locally resonating spatial coiling structures have recently been adopted to design negative or high positive refractive index metamaterials. We have in the past experimentally demonstrated the unit cell characteristics of one kind of labyrinthine metamterial (Xie *et al*. PRL 2013) and its impedance matching improved versions (Xie *et al*. APL 2013). In this work, we present several coherent modulation devices based on our recently proposed tapered labyrinthine metamaterials. With thickness of only one or two metamaterial cell layers, we can create an acoustic blazed diffraction grating, a phase conjugation lens, or a flat lens that can perform plane wave-cylindrical wave conversion. The design process and experimental demonstrations will be presented. The coherent controlling devices feature precise phase modulation, high-energy throughput, broad operating bandwidth, and sub-wavelength thickness. Our work demonstrates that labyrinthine metamaterials can be the unit cells of choice for functional coherent acoustic modulation devices.

10:45

2aSA7. Ultrasonic subwavelength focusing above periodic membrane arrays in immersion. Shane Lani, Karim G. Sabra, and F. Levent Degertekin (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30332, shane.w.lani@gmail.com)

Subwavelength focusing and imaging has been a long sought after goal and one that metamaterials can possibly achieve. In 2011, Lemoult *et al.* used time reversal techniques to focus sound to as small as $\lambda/25$ in air by using the evanescent wave field above a gird of soda cans acting as Helmholtz resonators [Lemoult *et al.* Phys. Rev. Lett. **107**, 064301, (2011)]. This paper will demonstrate subwavelength focusing in immersion in the 11–0 MHz frequency range with capacitive micromachined ultrasonic transducer (CMUT) arrays. CMUTs are microscale (10–100 μm wide) membrane arrays, which support evanescent surface waves that derive their dispersive properties not only from the periodic structure of the array, but also from the membrane resonance. Furthermore, CMUTs have embedded electrodes for electrostatic excitation and detection of acoustic waves which allow implementation of time reversal techniques to focus the dispersive evanescent surface waves using only the CMUTs on the same substrate as sources and receivers. Using a finite boundary element method simulation, we demonstrate subwavelength focusing at points in the near-field above a 2D CMUT array in immersion.

11:00

2aSA8. Acoustic metamaterial elements from tunable elastic shells. Alexey Titovich and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey17@eden.rutgers.edu)

Elastic shells are used as elements in novel acoustic metamaterials. Tuning the shells produces unnatural and favorable acoustic qualities in the quasi-static regime. This is achieved by internally stiffening the shell with an axisymmetric distribution of springs which connect the added central mass to the shell. The two parameters: stiffness and mass are carefully optimized for the desired effect. Flexural resonances of the shell dominate the frequency response, but are constrained in the quasi-static regime as shown by the analytical model. As an example of transparency, an aluminum shell of radius 1 cm is tuned to water with and acrylic internal oscillator exhibiting a near-zero scattering cross section up to ka = 0.6. Also, individually tuning each shell in a fluid saturated array is a means of creating devices based on transformation acoustics such as a cylindrical to plane wave lens. Investigations of favorable high frequency effects and active tuning are presented. Another method of changing the effective acoustic properties of a shell is to attach a second shell to the inside creating a composite structure. The thickness of each shell is optimized to yield desired effective medium properties. Tuning to water yields a broad frequency range of transparency.

11:15

2aSA9. Physical constraints on lossy acoustic metamaterials with complex effective properties. Caleb F. Sieck, Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 4021 Steck Ave. #115, Austin, TX 78759, cfsieck@utexas.edu), and Andrea Alù (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Recent theoretical and experimental work on acoustic metamaterials (AMM) has demonstrated materials that demonstrate many properties, such as negative modulus and density, beyond what is capable using conventional materials. In most cases, AMM are assumed to be passive and causal with frequency dependent losses accounted for via complex modulus and density. Despite the maturity of AMM research, literature concerning the physical constraints on the complex effective constitutive properties for passive, causal AMM is very limited. This work presents the physical limits for the real and imaginary effective dynamic mass density and dynamic compressibility via recourse to restrictions placed on the AMM by conservation of energy, passivity, and causality. We further note that constitutive properties are determined from the effective wavenumber and impedance extracted from simulation or experiment. Although care is normally taken to guarantee that passivity holds for the wavenumber and impedance, assumptions implicit in various homogenization schemes can result in constitutive properties that do not satisfy passivity and causality. This work will therefore also discuss implications on AMM homogenization and extraction of properties due to constraints based on the foundational concepts of conservation of energy, passivity, and causality. [This work was supported by the Office of Naval Research.]

TUESDAY MORNING, 6 MAY 2014 BALLROOM A, 8:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Speech Perception I (Poster Session)

Sayako Earle, Chair
Dept. of Speech, Lang. and Hearing Sci., Univ. of Connecticut, 123 Davis Rd., Storrs, CT 06268

All posters will be on display from 8:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

2aSC1. Gradient perception of within-category nasality in English vowels. Georgia Zellou and Delphine Dahan (Linguist, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu)

Nasality encodes a phonemic consonant contrast in English, and its acoustic correlates affect adjacent vowels. The phonological status of nasality predicts that people encode vowel nasality as a discrete and binary feature, i.e., as the presence or absence of velum lowering. The present study examined whether listeners can also infer the degree of velum lowering encoded in a vowel. We resynthesized 20 word pairs that contrasted in vowel nasality only (e.g., bet vs. bent [produced with no nasal murmur]) and created, for each pair, a continuum of seven stimuli varying linearly in their degree of vowel nasality. The 140 stimuli were used to assess listeners' categorization and discrimination. First, participants categorized each stimulus as oral or nasal. They were then presented with two stimuli selected from the same continuum and judged whether they consisted of physically identical or different tokens. Using linear-regression modeling, we demonstrated that discriminating between two stimuli is determined by their acoustic distance in velum height above and beyond their category assignment. Thus, people can encode velum height in stimuli they categorize identically. This finding adds to the growing body of evidence that listeners track properties of speech that go beyond establishing lexical contrasts.

2aSC2. Arrays of subcritical width rectangular speech bands maintain intelligibility at high intensities. Richard Warren, James Bashford, and Peter Lenz (Psych., Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, rmwarren@uwm.edu)

Speech intelligibility declines at high intensities for both normally hearing and hearing impaired listeners. However, it appears that this rollover can be minimized and intelligibility preserved by reducing speech in high frequency regions to an array of noncontiguous bands having vertical filter slopes (i.e., rectangular bands) and widths substantially narrower than a critical band. Normally hearing listeners were presented with sentences consisting of a 500-Hz lowpass pedestal band and an array of ten 4% bands spaced at 1/3-octave intervals from 1000 Hz to 8000 Hz. The pedestal band was fixed at 70 dB and the subcritical-band array varied from 55 to 105 dB in peak level. Desired sub-ceiling intelligibility ranged from 80 to 89% and was statistically asymptotic for levels from 75 to 105 dB. The largest intelligibility difference across that range, found between 85 and 105 dB, was just 1.6%. For that same contrast in levels, Molis and Summers [ARLO **4**, 124–128 (2003)] obtained a significant intelligibility loss of 26.7% for spectrally continuous highpass speech. It is suggested that subcritical-width bandpass

filtering reduces rollover by limiting firing rate saturation to a subset of fibers comprising individual critical bands. Implications for hearing aid construction will be discussed. [Work supported by NIH.]

2aSC3. Asymmetries in vowel perception: Do they arise from focalization, perceptual magnets, or both? Matthew Masapollo and Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca)

While directional asymmetries are ubiquitous in cross-language studies of vowel perception, their underlying mechanisms have not been established. One hypothesis is that listeners display a universal perceptual bias favoring vowels with greater formant frequency convergence, or focalization (Polka and Bohn, 2011). A second, but not mutually exclusive, hypothesis is that listeners are biased toward prototypical vowel exemplars in their native language (Kuhl, 1993). In a test of these hypotheses, English listeners discriminated synthesized English /u/ and French /u/ vowels presented in pairs. While the French /u/ tokens exhibit greater formant convergence (between F1 and F2), English listeners have previously been shown to rate the English /u/ tokens as "better" instances of the category (Molnar, 2010). Preliminary results demonstrate that the degree of focalization affects vowel discrimination. When discriminating vowel changes presented in the direction going from the more focal (French) to less focal (English) /u/ vowels, English listeners' reaction times were slower, relative to the same changes presented in the reverse direction. These results suggest that listeners treat the more focal vowels as perceptual reference points. Additional data collection with French listeners is ongoing. The implications of these findings for theories of vowel perception will be discussed.

2aSC4. Stream segregation of concurrent speech and the verbal transformation effect: Influence of fundamental frequency and lateralization cues. Marcin Stachurski, Robert J. Summers, and Brian Roberts (Psych., School of Life & Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk)

Listening to a repeating recorded word produces verbal transformations (VTs); perceptual re-grouping of phonetic segments may contribute to this effect. The influence of fundamental frequency (F0) and lateralization grouping cues was explored by presenting two concurrent sequences of the same word resynthesized on different F0s (100 and 178 Hz). In experiment 1, listeners monitored both sequences simultaneously, reporting for each any change in stimulus identity. Three lateralization conditions were used—diotic, interaural time difference ($\pm$680-μs ITD), and dichotic. The results were similar for the first two, but fewer forms and later transformations were reported in the dichotic condition. This suggests that large lateralization differences between the two sequences per se have little effect—rather, there are more possibilities for perceptual re-grouping when each ear receives both sequences. For all conditions, VTs reported on one sequence were mainly independent of the other. Experiment 2 investigated the effect of number of sequences presented and monitored. The most forms and earliest transformations were reported when two sequences were presented but only one was monitored, indicating that high task demands reduce reporting of VTs for concurrent sequences. Overall, these findings support the idea that perceptual re-grouping contributes to the VT effect. [Work supported by EPSRC.]

2aSC5. Does a perceived intensity cause pitch change in noise-vocoded vowels? Marina Takabayashi (Sensory and Cognit. Neural System Lab., Faculty of Life and Medical Sci., Doshisha Univ., 15-11 Okenoi-cho,Takeda, Fushimi-ku, Kyoto-shi 612-8421, Japan, bmk1086.splash@gmail.com), Kohta I. Kobayasi, and Hiroshi Riquimaroux (Sensory and Cognit. Neural System Lab., Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Noise-vocoded speech sound (NVSS) is the synthesized speech sound whose frequency information is greatly reduced while the amplitude envelope information remains preserved. In NVSS the fundamental frequency does not exist, and a change in amplitude is perceived as a change in not only loudness but also pitch. Original speech having physically same amplitude is not always identical. The phenomenon might occur also in NVSS. The purpose of this study is to examine whether change in not amplitude but loudness creates pitch change in NVSS. Subjects listened to paired original speech Japanese vowels and judged whether the second vowels were perceived louder or softer than the first ones. And they listened to paired noise-vocoded vowels to evaluate whether pitch of the second sounds rises or falls from the first ones. Results show that changes in loudness seem to cause pitch change in NVSS. After this, loudness perception in NVSS will be investigated. The data will show whether pitch of the second sounds change from the first ones or not when loudness of the first and the second sounds are same. And loudness perception in noise-vocoded vowels will be compared with loudness perception in the original speech.

2aSC6. Perceived emotional valence in clear and conversational speech. Shae D. Morgan and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

In the laboratory, talkers asked to speak as though talking to an individual with hearing loss modify their speech from their everyday conversational style to a "clear" speaking style. In the real world, individuals with hearing loss sometimes complain that their frequent communication partners seem to be shouting at them, while the communication partners insist that they are just trying to speak more clearly. Acoustic analyses have contrasted angry speech with neutral speech and clear speech with conversational speech. A comparison of these analyses reveals that angry speech and clear speech share several acoustic modifications. For example, both clear speech and angry speech show increased energy at high frequencies. The present study will explore whether clear speech sounds angry to listeners. Young adult listeners with normal hearing will be presented with conversational and clear sentences from the Ferguson Clear Speech Database (Ferguson, 2004) and asked to assign an emotion category to each sentence (anger, sadness, happiness, fear, disgust, or no emotion). The resulting data will show whether clear speech is more likely to be judged as sounding angry than typical conversational speech.

2aSC7. Error analysis and modifications to the short-time speech transmission index. Karen Payton and Matthew Ferreira (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, kpayton@umassd.edu)

The Speech Transmission Index (STI) predicts the intelligibility of speech degraded by noise and reverberation. Recently, Payton and Shrestha [J. Acoust. Soc. Am. **134**, 3818–3827 (2013)] reported on a short-time speech-based STI (ssSTI) to predict time-varying intelligibility of speech using analysis windows shorter than 1 s. While the ssSTI generally tracked a theoretical STI calculated using octave-band signal-to-noise ratios, it deviated from the theoretical calculation for windows shorter than 0.3 s. The current work analyzes and improves the performance of the ssSTI for speech degraded by stationary speech-shaped noise. Using a cluster analysis, the time-varying standard deviation was determined to be inversely proportional to window length, octave band and speech envelope variance. Two ssSTI modifications are proposed to reduce the 0.3 s window limitation: A silence detection algorithm eliminates non-zero ssSTI values that occur during silence and a modified envelope extraction scheme reduces the standard deviation by increasing envelope bandwidth. Using the 0.3 s window as a performance benchmark, new octave-band specific window limitations, ranging from 151 ms to 21 ms, were established. The modified ssSTI also works with common octave-band window lengths as short as 30 ms when full envelope bandwidths are used in combination with the silence detector.

2aSC8. Using the short-time speech transmission index to predict speech reception thresholds in fluctuating noise. Matthew Ferreira and Karen Payton (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, mjferreira128@gmail.com)

The Speech Transmission Index (STI) predicts the intelligibility of speech degraded by noise and reverberation. Recently, Payton and Shrestha [J. Acoust. Soc. Am. **134**, 3818–3827 (2013)] reported on the ability of a short-time speech-based STI (ssSTI) to predict the intelligibility of speech

in the presence of fluctuating noise using analysis windows shorter than 1 s. They found the ssSTI highly correlated with theoretical STI calculations using windows as short as 0.3 s. In the current work, extended versions of the ssSTI were investigated for their ability to improve speech intelligibility prediction in the presence of fluctuating noise; a condition for which the long-term STI incorrectly predicts the same intelligibility as for stationary noise. No STI metric predicts a normal-hearing listener's improved ability to perceive speech in the presence of fluctuating noise as compared to stationary noise at the same signal-to-noise ratio. The investigated technique used window lengths that varied with octave band, based on human auditory temporal resolution as in the Extended Speech Intelligibility Index [Rhebergen and Versfeld, J. Acoust. Soc. Am. **117**, 2181–2192 (2005)]. An extended sSTI using speech-shaped noise instead of speech as a probe predicted published speech reception thresholds for a variety of conditions.

2aSC9. Gradient coding of voice onset time in posterior temporal cortex. Nathaniel D. Anderson (Beckman Inst., Univ. of Illinois, 1412 Beckman Inst., 405 N. Mathews Ave., Urbana, IL 61801, nandrsn3@illinois.edu), Joseph C. Toscano, Monica Fabiani, Gabriele Gratton, and Susan M. Garnsey (Beckman Inst., Univ. of Illinois, Urbana-Champaign, IL)

The issue of whether early stages of speech processing are influenced by category has been central to work in speech perception for decades. We present the results of an experiment using fast diffusive optical neuroimaging (Gratton and Fabiani, 2001, Int. J. Psychophysiol.) to address this question directly by measuring neural responses to speech with high temporo-spatial resolution. We found that changes in voice onset time (VOT) along a /b/-/p/ continuum evoked linear changes in neural responses in posterior superior temporal gyrus (pSTG) 100 ms after stimulus onset. This is the first non-invasive observation of such responses in humans. It is consistent with results from recent event-related potential (Toscano *et al*., 2010, Psychol. Sci.) and fMRI (Blumstein *et al*., 2005, J. Cognit. Neurosci.) studies, and provides evidence that those results reflect listeners' early encoding of speech sounds in pSTG, independently of phonological categories. Thus, the results provide evidence that speech perception is based on continuous cues rather than discrete categories. We discuss these results in light of recent intra-cranial EEG studies reporting either categorical effects in pSTG (Chang *et al*., 2010, Nature Neurosci.) or evidence that pSTG maintains fine-grained detail in the signal (Pasley *et al*., 2012, PLoS Biol.).

2aSC10. Effects of linguistic structure on perceptual attention given to different speech units. Shinae Kang and Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, UC Berkeley, Berkeley, CA 94720-2650, sakang2@berkeley.edu)

Listeners can shift their attention to different sizes of speech during speech perception. This study extends this claim and investigates if linguistic structure affects this attention. Since English has a larger syllable inventory than Korean and Japanese, each phoneme plays a larger functional role. Also, listeners have different levels of phonological awareness due to the differences in the orthography. We focus on the effect of perceptual attention on the perceptibility of intervocalic consonant clusters (VCCV) and whether it varies cross-linguistically by these structural factors. We first recorded eight talkers saying VC- and CV-syllables and spliced the syllables to create non-overlapping VC.CV-stimuli. Listeners in three language groups (English/Korean/Japanese) participated in a 9-Alternative-Forced-Choice perception task. They identified the CC as one of 9 alternatives ("pt", "pk", "pp", etc.) and in an attention-manipulated condition did the same task while also monitoring for target talkers. The preliminary result shows that Korean listeners showed less perceptual sensitivity to clusters than English listeners. Also, the English listeners showed better perception of syllable coda when prompted to focus on coda only. The result indicates that the linguistic structure of a language can potentially affect the level of perceptual attention that its users give to a linguistic unit.

2aSC11. Perception of dialectal prosody in Taiwan Mandarin. Mao-Hsu Chen (Linguist, Univ. of Pennsylvania, 4200 Spruce St., Apt. 310, Philadelphia, PA 19104, chenmao@sas.upenn.edu)

This pilot study aims at answering whether prosodic cues alone can account for the differences among three regional dialects in Taiwan Mandarin, Northern, Central, and Southern, which all belong to Mandarin Chinese. Assumed that prosodic cues alone can be used for distinguishing among different Taiwan Mandarin dialects, a perception experiment was conducted. The Northern dialect was best recognized while the identification rates of the Central and the Southern dialects were slightly below chance level. Results showed the tendency that it was easier for listener to identify his or her own dialect than it was to detect other regional dialects. Gender effect was observed to play a role in the recognition of the three dialects, which led to the following-up production experiment intended for exploring the acoustic differences among these three dialects. Preliminary results examined the descriptive tonal patterns of all four lexical tones in three Taiwan Mandarin dialects. The tonal registers produced by the Northern dialect speakers were more prominent than those of the Central and the Southern dialects in that they had the highest normalized F0 values for T1 tokens and greater pitch range, or steepest slope, for the other three contour tones, compatible with the result of the perception experiment.

2aSC12. Unpredictable and unintelligible: Individual differences in predictability-based reduction affect speech intelligibility. Rory Turnbull (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43215, turnbull@ling.osu.edu)

Words spoken in high predictability (HP) contexts tend to be phonetically reduced and less intelligible than words spoken in lower predictability (LP) contexts. However, individual differences in degree and strategies of reduction, and their effects on intelligibility, are largely unexplored. This study examined the role of autistic traits in speech intelligibility. Sixteen participants completed a speech in noise word identification task at 2dB SNR. Stimuli were recordings of words spoken in HP and LP contexts, extracted from sentences produced by talkers ranging in autism-spectrum quotient (AQ) scores. After the identification task, listeners also completed the AQ questionnaire. A higher AQ score indicates a greater prevalence of autistic traits in one's cognitive style. Logistic mixed effect regression modeling revealed an expected effect of duration, such that longer words were more likely to be correctly identified. Further, talkers with higher AQ scores were less intelligible than talkers with lower AQ scores. Unexpectedly, no effect of predictability condition was observed: LP words were not consistently more intelligible than HP words. No effect of individual differences in listener AQ scores was observed. These results suggest that predictability-based enhancement and reduction strategies vary between individuals and are not necessarily for the benefit of the listener.

2aSC13. Multidimensional scaling and order asymmetry of the acoustic change complex to voiceless fricatives in Polish, English, and Finnish listeners. Emma Brint, Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, 1 High Rd., London E182QN, United Kingdom, emma.brint.11@ucl.ac.uk), and Anita Wagner (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen , Groningen, Netherlands)

The acoustic change complex (ACC) is a P1-N1-P2 onset response to changes between sounds measured using EEG, which when measured at a central location, is not thought to be affected by the language experience of the listener. Recent research has shown that with a pair of stimuli, there is an asymmetry between the ACC recorded when they are presented in one order compared to the other. This study used eight voiceless fricatives to measure the ACC to all possible pairs from EEG recordings of 15 English, Finnish, and Polish listeners. The ACC magnitude was used to perform multidimensional scaling to create a two dimensional perceptual space for each language that is driven by the spectral characteristics of the stimuli.

Asymmetry occurred in all three languages depending on how peaky the first and second stimuli were. This asymmetry effect was greater for the Polish group, indicating that the Polish listeners' language experience increases their sensitivity to the spectral peakiness of fricatives. These results show that the ACC is in fact susceptible to native language effects, enhancing its application as a test for auditory speech processing.

2aSC14. The perception of postvocalic English stops in diphthongs and monophthongs using gating experiment. Siriporn Lerdpaisalwong (of Linguist, Univ. of Wisconsin-Milwaukee, 1915 E. Kenilworth Pl., Mailbox no. 83, Milwaukee, WI 53202, siriporn@uwm.edu)

There have been many studies on the acoustic cues in the preceding vowels for the perception of the final consonants (Hillenbrand *et al*., 1983; Warren and Marslen-Wilson, 1988). Yet, none of the studies has been conducted to see whether there are any differences in listener perception of a final consonant in areas of diphthongs versus monophthongs. Since the average duration of diphthongs /eɪ/ and /oʊ/ is longer than that of monophthongs /i/ and /u/ (Hillenbrand *et al*., 1995), this study investigates whether a listener perceives postvocalic English stops /p, t, k/ in monophthongs /i/ and /u/ faster than in diphthongs /eɪ/ and /oʊ/ using the gating paradigm. Fifteen American English speakers participated in this study (6F, 9M; 18–44 y.). The stimuli consist of 24 CVC words and noncewords with final /p, t, k/. Vowels of 24 words were chopped into ten gates each. The results show that the listeners perceived the final stops at about the same gates (areas) in both diphthongs and monophthongs in both word types. The results also show no significant difference among the perception of three stops, except between /p/ and /k/ in the real words with monophthongs.

2aSC15. Sonority of adjacent segments in the perception of durational distinctions. Olga Dmitrieva (Purdue Univ., 100 North University St., Beering Hall, Rm. 1289, West Lafayette, IN 47907, odmitrie@purdue.edu)

The distributional typology of length contrasts in consonants suggests that the sonority of adjacent segments may be relevant for the perception of duration differences in consonants. Across languages, short consonants typically contrast with long consonants intervocalically or next to high sonority consonants, such as glides or liquids. It has been proposed that surrounding sonorants facilitate the perception of durational distinctions by providing clear acoustic cues to the beginning and the end of the target consonant, which makes it easier to estimate the target's duration (Bradley, 2001; Padgett, 2003). The present study investigates this hypothesis by examining the effect of adjacent segments' sonority on the perception of durational differences in alveolar voiceless stops. Target stops of two lengths were placed in the environment of a preceding or following vowel, liquid, nasal, fricative, or non-homorganic stop and presented to listeners in a discrimination experiment. Initial results indicate an important role for syllable structure and its sonority profile in duration discrimination: Listeners appear to be more sensitive to length distinctions when stops targets are in the onset position preceded by a higher-sonority coda (e.g., al.ta) compared to stop targets in the coda position followed by a higher-sonority onset (e.g., at.la).

2aSC16. On the gradience in perceptibility of word-final voicing contrast in Russian. Mayuki Matsui (Linguist, Hiroshima Univ., 1-2-3 Kagamiyama, Higashi-Hiroshima-shi, Hiroshima 739-8522, Japan, matsui-ma@hiroshima-u.ac.jp)

Russian is one of the languages in which the underlying voiced obstruents devoice in word-final position, resulting in voicing neutralization (e.g., /rok/ [rok] "fate" vs. /rog/ [rok] "horn"). However, recent studies have shown that word-finally devoiced (i.e., underlying voiced) obstruent and the underlying voiceless one are acoustically different (Chen 1970, Dmitrieva *et al*. 2010, among others). That is, word-final devoicing shows a case of incomplete neutralization. Also, those acoustic differences are in some degree perceptible for listeners (Matsui 2011, Kharlamov 2012). This paper presents a perceptual analysis of incompletely neutralized obstruents in Russian. Pseudo-nouns produced by the native speakers of Russian were presented to the listeners as auditory stimuli. Sixteen native listeners identified what they heard in a forced-choice identification task. The most striking result to be reported in this paper is that the listeners' sensitivity to the underlying voiced and voiceless stimuli is different between obstruent types: the underlying voicing contrast in stops is harder to perceive than that in fricatives for listeners. Other than obstruent type, the effects of the stimuli presentation type and of the magnitude of the acoustic difference will also be discussed.

2aSC17. Discriminability and perceptual saliency of acoustic cues for final consonant voicing in simulations of cochlear-implant and electric-acoustic stimulation. Ying-Yee Kong (Dept. of Speech Lang. Pathol. & Audiol., Northeastern Univ., 226 Forsyth Bldg., 360 Huntington Ave., Boston, MA 02115, yykong@neu.edu), Ala Mullangi (Bioeng. Program, Northeastern Univ., Boston, MA), Matthew Winn (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI), and Gail Donaldson (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Multiple redundant acoustic cues can contribute to the perception of a single class of speech sounds. This study investigates the effect of spectral degradation on the discriminability and perceptual saliency of acoustic cues for phonetic identification of word-final fricative voicing in "loss" versus "laws," and possible changes that occur when low-frequency acoustic cues are restored. Three acoustic cues that contribute to the word-final /s/-/z/ contrast (first formant frequency [F1] offset, vowel-consonant duration ratio, and consonant voicing duration) were systematically varied in modified natural speech syllables. The first experiment measured listeners' ability to discriminate differences among stimuli within a single cue dimension. The second experiment examined the extent to which listeners make use of a given cue to label a syllable as "loss" versus "laws" when multiple cues are available. Normal-hearing listeners were presented with stimuli that were either unprocessed, processed with 8-channel noise-band vocoder to approximate CI spectral degradation, or low-pass (LP) filtered to simulate low-frequency residual hearing. They were tested in four listening conditions: unprocessed, vocoder alone, LP alone, and vocoder + LP where low-frequency fine-structure cues could enhance F1 perception and voicing cues. The impact of listening condition on discriminability and weighting of different acoustic cues will be discussed.

2aSC18. Defining spectral and temporal resolutions of information-bearing acoustic changes for understanding noise-vocoded sentences. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu) and Matthew Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Information-bearing acoustic changes (IBACs) in the speech signal are highly important for speech perception. This has been demonstrated for full-spectrum sentences using cochlea-scaled entropy (CSE; Stilp and Kluender, 2010 *PNAS*) and noise-vocoded sentences using an adapted metric (CSE_{CI}; Stilp *et al*., 2013 *JASA*). While IBACs appear fundamental to speech perception most broadly, Stilp *et al*. tested a single set of vocoder parameters, obscuring the breadth and depth of perceptual reliance upon these acoustic changes. Here, TIMIT sentences were noise-vocoded with variable spectral resolution (4-24 spectral channels spanning 300–5000 Hz), variable temporal resolution (4–64 Hz amplitude envelope cutoff frequency), or combinations therein. High-CSE_{CI} or low-CSE_{CI} sentence intervals were replaced with speech-shaped noise. As spectral resolution decreased, IBACs became more important for sentence understanding, especially at 6–12 channels. At high spectral resolutions, performance nearly overcame replacement of low-CSE_{CI} intervals but not high-CSE_{CI} intervals. Importance of IBACs at different temporal resolutions was largely driven by overall intelligibility, suggesting CSE_{CI} has sufficient temporal resolution at low modulation frequencies. Data exploring spectral-temporal tradeoffs will also be presented. Peak-picking strategies in CIs select and stimulate channels according to their amplitudes, but results suggest additional perceptual benefit may be offered by encoding IBACs as well.

2aSC19. Sentence intelligibility during segmental interruption and masking by speech-modulated noise. Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC), William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, Charleston, South Carolina), and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Amplitude modulation from a competing talker can interfere with a listener's ability to process informative speech cues. The current study investigated how noise modulated by the wideband temporal envelope of a single competing talker impacts the contribution of consonants and vowels to the intelligibility of the target sentence. High variability speech materials were used, including talkers from different dialect regions. Young normal-hearing, older normal-hearing, and older hearing-impaired listeners completed speech recognition tests. All listeners received spectrally shaped speech matched to their individual audiometric thresholds to ensure sufficient audibility. Preliminary results demonstrated similar performance among the listener groups. When the modulated masker was interrupted, performance in consonant and vowel conditions was similar. However, sentence intelligibility in continuous single-talker-modulated noise was higher when vowels in the target sentence were preserved, as compared to consonants. Thus, masker continuity may facilitate source segregation for vowels more than for consonants. A second experiment that varied masker modulation rate for younger adults with normal hearing showed that modulation rate had more effect on processing cues conveyed by consonants than by vowels. Implications for hearing-impaired and older adults will be discussed. [Work supported by grants from NIH/NIDCD and ASHA.]

2aSC20. Acoustic properties of multi-talker babble. Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu), Kenneth de Jong, Kirsten Regier, Aaron Albin (Linguist, Indiana Univ., Bloomington, IN), and Yen-Chen Hao (Modern Foreign Lang. and Literatures, Univ. of Tennessee, Knoxville, TN)

Multi-talker babble can function as an excellent masker for speech stimuli in perception experiments. It has a higher degree of ecological validity than other maskers (e.g., white noise, speech-shaped noise), as it is a type of noise that many listeners encounter on a regular basis in everyday life. In addition, maskers constructed from speech have, by definition, acoustic properties similar to that of the signal. While multi-talker babble is used extensively in speech perception research, relatively little work has been done on the fine-grained acoustic properties of multi-talker babble. We present analyses of a number of acoustic properties of multi-talker babble generated by randomly combining phonetically balanced utterances (e.g., amplitude modulation depth, amplitude modulation frequencies, spectral properties, and spectro-temporal variability). In order to gain a fuller understanding of the nature of multi-talker babble, we analyze how the acoustic properties of babble vary as a function of the number (2–20), gender, and native language (English vs. Mandarin) of the speakers constituting the babble components. Future extensions of this work will (a) focus on how these acoustic variables affect speech perception, and (b) provide the foundation for a web-based system for generating customized samples of multi-talker babble noise for speech perception researchers.

2aSC21. The relationship between fluency, intelligibility, and acceptability of non-native spoken English. Mengxi Lin (Linguist, Purdue Univ., West Lafayette, IN 47907, lin211@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Non-native accented speech is typically less intelligible and less fluent than native speech, but it is unclear how these factors interact to influence perceived speech quality. To investigate this question, the speech of 20 non-native speakers of English varying in proficiency and native language was evaluated. Subjective measures of speech quality (listening effort, acceptability, and intelligibility) were compared to objective measures of word recognition by native listeners, and to acoustic measures of fluency and of segmental and suprasegmental properties related to intelligibility. Results showed that subjective quality measures were highly related to one another and to word recognition and were most strongly predicted by measures of fluency. Segmental and suprasegmental measures did not predict word recognition or subjective speech quality. There was also an interaction between the effects of proficiency and speaker's native language on word recognition, but this did not extend to subjective measures. Finally, listeners who first heard high-proficiency speakers gave overall lower subjective quality ratings but there was no interaction between proficiency and presentation order. Multivariate analyses suggest that factors related to speaking rate, including pause duration, have the greatest effect on measures of acceptability, intelligibility, and listening effort. [Work supported by Purdue Linguistics and Purdue Research Foundation.]

2aSC22. Speech intelligibility can improve rapidly during exposure to a novel acoustic environment. Sofie Aspeslagh (MRC/SCO Inst. of Hearing Res. – Scottish Section, Glasgow Royal Infirmary, 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, sofie@ihr.gla.ac.uk), D. Fraser Clark (Dept. of Computing, Univ. of the West of Scotland, Paisley, United Kingdom), Michael A. Akeroyd, and W. Owen Brimijoin (MRC/SCO Inst. of Hearing Res. – Scottish Section, Glasgow, United Kingdom)

In natural listening environments, the background noises and the acoustic spaces they occupy can vary greatly, both in their characteristics and in their impact on speech intelligibility. It has been suggested that listeners build up a statistical representation of ongoing noises; however, listeners can move around a room or move from one room to another, constantly changing both the background noises and the reverberation around them. If listeners are to make use of such statistics to aid in speech intelligibility, they must engage in a constant adaptation. We asked normal and hearing-impaired listeners to identify in real-time a stream of random target words in a contiguous series of novel acoustic environments lasting 13 s each, thus measuring the extent and time-course of changes in speech intelligibility that listeners experience in a new environment. The results from our task demonstrate that most listeners experience a rapid increase in speech intelligibility over several seconds of exposure to certain acoustic environments, but not to others. This suggests that there are classes or types of noises and reverberations that can be adapted to over a short time course and others that cannot. [Work supported by the MRC and the CSO.]

2aSC23. Sizing down the competition: Speaking style and word recognition. Kristin Van Engen (Psych., Washington Univ., 176 Landa St. #514, New Braunfels, Texas 78130, kj.vanengen@gmail.com)

Identifying words with many phonological neighbors is more challenging for older adults than for younger adults. This difference has been attributed to reductions in inhibitory control associated with aging, which impair older adults' ability to cope with competition from similar-sounding words (Sommers and Danielson, 1999). Many difficulties in speech identification can be alleviated, however, when speech is produced clearly (i.e., the style adopted naturally by speakers when they perceive that their interlocutors are having difficulty understanding them). The current study investigates whether the acoustic-phonetic enhancements of clear speech can also reduce the inhibitory challenge of word recognition. If so, it is predicted that listeners will receive a greater benefit from clear speech when identifying lexically difficult words (i.e., words with many neighbors) vs. lexically easy words (i.e., words with fewer neighbors). Younger and older adults performed a word-recognition task in noise. Results to date show that the clear speech benefit is indeed greater for lexically difficult words than for lexically easy words for both groups of listeners. This pattern of results suggests that clear speech reduces the inhibitory demands associated with word recognition by increasing the perceptual difference between phonological neighbors.

2aSC24. Subcritical width rectangular bands of vocoded speech reveal the nature of envelope processing. James A. Bashford, Richard M. Warren, and Peter W. Lenz (Psych., Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, bashford@uwm.edu)

Bandwidth requirements for temporal envelope processing were examined using sentences that were reduced to arrays of sixteen rectangular bands (4800 dB/octave rolloff) having center frequencies ranging from 250

Hz to 8000 Hz, spaced at 1/3-octave intervals, and having subcritical bandwidths ranging from 4% to 0.5%. Envelopes extracted from the speech bands, without smoothing, were used to modulate white noise or tones centered at the speech band center frequencies. When the bandwidth of the modulated signals matched that of the speech, tone-vocoded arrays were more intelligible than noise-vocoded arrays and less intelligible than parent speech-band arrays. However, doubling the bandwidths of tone-vocoded arrays, which passed the upper and lower modulation sidebands fully, increased intelligibility to that of the parent speech array. Moreover, when the widths of modulated noise-band arrays were expanded to either an ERBn or 1/3-octave, their intelligibility equaled that of the speech for parent bandwidths of 4% and 2% and greatly exceeded speech array intelligibilities for parent bandwidths of 1% (58% vs. 25%) and 0.5% (28% vs. 3%). These and other findings indicate that optimal temporal envelope processing of speech requires that envelope cues stimulate a majority of fibers comprising critical bands. [Work supported by NIH.]

2aSC25. What explains perceptual weighting strategies of children with CIs: Auditory sensitivity or language experience? Susan Nittrouer, Amanda Caldwell-Tarr, and Joanna H. Lowenstein (Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, nittrouer.1@osu.edu)

Cochlear implants (CIs) have tremendously improved speech perception for deaf children, but problems remain. To examine why, this study compared weighting strategies of children with CIs and children with normal hearing (NH), and asked if these strategies are explained solely by the degraded spectral representations they receive through their implants, or if diminished opportunity to hear the ambient language accounts for some of the effect, as well. One hundred 8-year-olds (49 with NH and 51 with CIs) were tested on four measures: (1) labeling of a final-voicing contrast with one duration and one formant-transition cue; (2) labeling of a fricative-place contrast with one stable spectral and one formant-transition cue; (3) duration discrimination; and (4) glide discrimination. Children with NH and CIs weighted the duration cue similarly, suggesting children with CIs have sufficient experience to acquire language-appropriate strategies when cues are salient. Differences in weighting of spectral cues (both stable and time-varying) were found, but were not entirely explained by auditory sensitivity. The conclusion was that more salient cues facilitate stronger weighting, but individuals differ in how salient cues need to be to capture perceptual attention. Stimulus familiarity (speech or non-speech) affects how reliably children attend to acoustic cues, as well.

2aSC26. A phonetic basis for the sonority of [X]. Sarah Bakst and Jonah Katz (Linguist, Univ. of California Berkeley, 1915 Bonita Ave., Studio A, Berkeley, CA 94704, bakst@berkeley.edu)

Although after voiceless stops French rhotics are realized as voiceless uvular fricatives [X], they pattern phonologically as high-sonority liquids and are the only obstruent allowed between a consonant and a vowel. This poses a problem for the sonority hierarchy. This experiment tests whether [X], which has apparent approximant-like formant structure (Yeou and Maeda, 1995) patterns like the fricative [f] or the approximant [l] in its ability to convey information in formant transitions from a preceding consonant. In an AX burst detection task, native English speakers heard syllables of the form ClV, CXV, and CfV (spoken by a native French speaker) with and without a burst. Pilot data ($n = 8$; 410 trials each) suggests participants are more likely to respond "same" for [X] and [l] trials than for [f] trials ($p < 0.001$), suggesting that [X] carries more redundant information from a preceding consonant than [f] does and thus behaves more like approximants than other fricatives do. This suggests that the sonority of [X] is grounded in acoustics and perception, rendering an abstract account of sonority unnecessary. This result predicts that other fricatives with long front cavities should also be able to function as high-sonority segments.

2aSC27. A listener-based account for dispersion effects in sound change. Thomas Denby (Linguist, Northwestern Univ., 1525 W Estes Ave., Apt. B2, Chicago, IL 60626, tdenby@u.northwestern.edu), Grant McGuire, and Jaye Padgett (Linguist, Univ. of California-Santa Cruz, Santa Cruz, CA)

Phonetic dispersion has been proposed as the driving force behind a number of closely related sound-change phenomena. Listener-based accounts of dispersion (Labov, 1994, 587; Wedel, 2006; Denby, 2013) posit that phonetically unambiguous productions influence future productions of the listener more than ambiguous productions. The mechanism that drives this is a filter by which ambiguous productions are not stored, and thus do not update the phonemic categories of the listener. In turn, they are not reflected in that listener's future productions. In a new experiment, subjects heard words in noise and were asked to identify them by responding using a keyboard, following Goldinger (1996). Stimuli were from monosyllabic stop-initial minimal pairs differing in initial voicing, e.g., pat/bat. Half of these pairs were unambiguous productions, while the stop-initial VOT of the other half were manipulated to be somewhat ambiguous. If subjects store ambiguous words normally, their accuracy should improve with every exposure. If however, they do not store ambiguous productions, their accuracy should improve less than it does for unambiguous productions. Using d' scores, a repeated-measures ANOVA confirmed differences in improvement for unambiguous and ambiguous conditions were significant. A follow-up study is being implemented to test and expand these results.

2aSC28. Evaluation of web speech and lab speech for automatic classification of prosody. Jonathan Howell (Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, howellj@montclair.edu)

This study evaluates the performance of two very different sources of speech data in prosodic classification tasks: naturally occurring speech collected from the web (e.g., podcasts) and speech elicited in a laboratory. Speech from the web contains useful variability from which to generalize, e.g., variations in speaker and context; however, this variation also includes much statistical "noise," which may obscure the correct generalizations, e.g., different dialects or overlapping speech and other acoustic artifacts. We trained machine learning algorithms (support vector machines and linear discriminant analysis) to detect prosodic prominence in utterances of the comparative construction, e.g., "than I did." From 217 web-harvested utterances and 394 lab-elicited utterances (16 items, 27 subjects), we extracted more than 300 acoustic values, including measures of duration, F0, F1, F2, intensity, amplitude, voice quality, and spectral tilt. Both the web and lab training sets yielded similarly high-accuracy classifiers. The best performing algorithm achieved 87.6% accuracy ($p < 0.05$) when trained on web data and tested on lab data and 90.6% accuracy ($p < 0.05$) when trained on lab data and tested on web data. Significance values were calculated using a permutation-achieved empirical distribution.

2aSC29. Effects of perceptual anchors on nasality ratings in speech. Kristine E. Galek (Speech Pathol. and Audiol., Univ. of Nevada, Reno, P.O. Box 193, Carnelian Bay, CA 96140, kegh70@gmail.com) and Thomas Watterson (Speech Pathol. and Audiol., Univ. of Nevada, Reno, Reno, NV)

To study the effects of perceptual anchors on nasality ratings in speech. Speech samples were obtained from 95 hypernasal children and 5 normal controls. Samples were randomized and duplicated (6 sets of 100 samples). Six listening groups ($N = 129$) rated nasality on a seven-point scale ("1" normal nasality to "7" severe hypernasality). Anchors were located at different points along the continuum for each group. A single anchor located at scale value "4" or at scale values "3" and "5" educed an assimilation of ratings. Anchors placed at scale values "1" and "7" educed a contrast effect in that the distribution of ratings shifted away from the anchor sites. A single anchor placed at scale value "7" educed a systematic regression of the distribution of ratings away from the anchor site. Three anchors placed at scale values "1", "4" and "7" educed a more even distribution of ratings across the entire scale than the other five anchored conditions. Three of the five anchor groups consistently rated the first sample of the rating task a scale value "2" (based on median findings) even though the first sample was a normal control sample. The degree of nasality in speech is influenced by perceptual anchors.

2aSC30. Locus of phonological deficits in adults with dyslexia. Stephanie N. Del Tufo, Joslynn Noyes, Rebecca Sylvia, Sarah Montanaro, and Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269, stephanie.del_tufo@uconn.edu)

Dyslexia is a developmental disorder that has traditionally been viewed as the consequence of impaired phonology. However, recent evidence suggests that the phonological deficits observed in dyslexia may reflect phonetic impairment earlier in the processing stream. The goal of the current work is to test this hypothesis. We use a phoneme-monitoring task to evaluate the relative use of two sources of information for making phonemic decisions (e.g., deciding if "dog" begins with /b/ or /d/). One source comes from a prelexical analysis of the speech signal (phonetic information) and the other source comes from a postlexical analysis (phonological information). Research has shown that healthy listeners flexibly shift between these sources based on experience with the acoustic signal. Specifically, phonemic decisions for native speech reflect phonetic information whereas phonemic decisions for foreign-accented speech reflect the use of phonological information. The interpretation is that phonetic processing is difficult for the foreign-accented speech, leading to a greater reliance on a postlexical analysis. If adults with dyslexia have impairment in phonetic processing, then we predict that phonemic decisions will reflect the use of phonological information for both types of speech, suggesting that the native speech signal is being processed as if it were accented.

2aSC31. Auditory and phonetic contributions to the neural mechanisms underlying vowel perception. Jeremy Burnison (Neurosci. Graduate Program, Univ. of Kansas, 1200 Sunnyside Ave., 4115 Haworth, Lawrence, KS, jburnison@ku.edu) and Jonathan Brumberg (Speech-Language-Hearing, Univ. of Kansas, Lawrence, KS)

One aspect of speech perception is our ability to identify the vowel sounds of spoken language. Identification accuracy of acoustic vowel targets has been shown to decrease as the first two formant frequencies of target sounds become more similar, suggesting that a vowel discretization exists in the F1-F2 auditory perceptual space. This is supported by prior work showing neural processes during vowel perception reflect recognition of discrete phonemes. In this study, we investigated the differential contributions of phonetic and acoustic factors on vowel perception using an event-related potential (ERP) protocol. Subjects first identified synthesized acoustic presentations of American English monopthong vowel sounds by selecting one of eight representative hVd words. Next, we presented synthesized sounds along the trajectory between two neighboring vowels in a mismatched negativity (MMN) oddball paradigm with one exemplar vowel as the standard stimulus. Preliminary results show statistically significant MMN amplitudes attenuate with standard-deviant vowel similarity, but include an additional MMN amplitude reduction at 50% vowel identification accuracy based on the behavioral responses. These results further clarify the neurological processing of vowel perception as a combination of auditory and phonetic factors in which acoustic differences elicit graded MMN responses that are augmented by shifts across phonetic boundaries.

2aSC32. Consonant confusability and its relation to phonological similarity. Sameer ud Dowla Khan (Linguist, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, sameeruddowlakhan@gmail.com)

Gradient similarity avoidance patterns in Bengali echo reduplication suggest that the most similar consonants to /t/ are, in order, /t, t^h, d, t̪, s, t̪^{h}, k, .../. Converted to confusability, this ranking predicts that aspiration is most confusable, followed by voicing, minor place, continuancy, major place, and sonority. To confirm this, 24 native speakers identified syllables masked with babble, noise, or nothing ("clear"). Results indicate that confusability reflects similarity as predicted by avoidance patterns. In clear speech, most errors involved voicing and aspiration. Other errors reflected dialect-specific alternations. Noise introduced the percept of a loud burst: fricatives were often heard as affricates, non-alveolar coronals as alveolars, and non-coronals as coronals. Babble largely resembled noise. This pattern suggests that voicing is the most confusable feature, followed by aspiration, minor place and continuancy, major place, and lastly sonority. This ranking seems Bengali-specific, as studies of English find place to be significantly more confusable than manner and voicing. These results suggest that at least for Bengali, phonological alternations and perceptual confusability are argued to be better representations of how speakers judge similarity, rather than patterns in the lexicon or metrics such as shared natural classes metric of Frisch *et al*. (2004).

2aSC33. Information structure guides prominence perception. Jason Bishop (City Univ. of New York, 2800 Victory Blvd., Staten Island, NY 10314, jason.bishop@csi.cuny.edu)

The present study investigates the effect of information structure on the perception of prosodic prominence in English. In particular, we probed for top-down effects related to the size of the focus constituent (broad VP focus versus narrow object focus) in simple subject-verb-object sentences using a naïve prosody "transcription" task. In this task, listeners heard the same productions of a sentence, but in different information structural (i.e., question) contexts, and provided self-report decisions about the prominence of words using a Likert scale. Two primary questions were asked. First, does information structural interpretation produce expectation-based prominence perception? In this case, it was predicted that the presence of focus would induce perceived prominence independent of the signal (via expectations based on experience with production patterns). Second, does information structural interpretation modulate signal-based prominence perception? In this case, it was predicted that the presence of focus would enhance sensitivity to signal-based cues (via the modulation of attentional resources). Results are presented that show evidence for both types of effects, demonstrating a multifaceted influence of sentence-level semantic/pragmatic meaning on the perception of the signal.

Session 2aSP

Signal Processing in Acoustics: Session in Honor of William M. Carey I

James Lynch, Chair
Woods Hole Oceanogr., MS # 11, Bigelow 203, Woods Hole, MA 02543

Chair's Introduction—8:25

Invited Paper

8:30

2aSP1. A scientific crossroad: Carey's influence in affecting shallow water acoustics research after the cold war. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu) and David P. Knobles (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Before the end of cold war most of the U.S. Navy's underwater acoustics research was focused on deep water. However, in the late 1980s and early 1990s, a shift toward shallow water environments was emphasized. With a body of knowledge from Ewing, Worzel, and Pekeris to Weston, Bill Carey took it upon himself to test a set of realistic goals in understanding complicated shallow water acoustics problems. His major field experiments in 1988–1993 produced results directing the community to address the issue of bottom attenuation and array coherence in a much more serious manner. In addition, due to his work, the community also considered the important problem of the effects of the water column, such as those associated with internal waves, on the acoustic propagation in these regions. This paper presents a summary of Carey's research in shallow water acoustics in light of the broader picture of the community's progress and direction in this field.

Contributed Papers

8:45

2aSP2. Laboratory measurements of compressional and shear wave speed and attenuation in muddy sediments. Megan S. Ballard, Kevin M. Lee, and Thomas G. Muir (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

As demonstrated by William M. Carey's field measurements in Dodge Pond, muddy sediments are characterized by a slow compressional wave speed and a low compressional wave attenuation [Carey and Pierce, POMA (5), 7001, 2009]. To gain insight into the measured data, a theoretical treatment of muddy sediments, named the Card House theory [Pierce and Carey, POMA (5), 7002, 2009], was developed. According the theory, isomorphous substitution causes each mud platelet to carry a net negative charge, and the resulting electrical interaction between platelets is responsible for creating a card-house structure. In this work, we examine laboratory measurements of compressional and shear wave properties in mud. Compared to the Dodge Pond measurements, which were affected by the size and distribution of gas bubbles present in the mud, the volume of gas in the laboratory samples was reduced by applying a vacuum. The estimated compressional wave speed is consistent with predicted values for a relatively gas-free mud. The estimated shear wave speed compares favorably with predicted values from the Card House theory. The electrochemical basis of the Card House model and its acoustical implications are also investigated. [Work supported by ARL:UT IR&D.]

9:00

2aSP3. Bill Carey and Bob Urick. David Bradley (Penn State Univ., PO Box 30, State College, PA 16870, dlb25@psu.edu)

Principles of Underwater Sound, written by Robert J. Urick, and last published in 1983, has been a staple on everyone's bookshelf. Bill Carey took on the job of updating that book, with the premise of not changing what was written, but simply updating each chapter with new information obtained over the succeeding three decades. One of the primary reasons the text was so popular is its ease in reading and grasping the import of the data and simplified ideas presented and their respective use(s) in the sonar equation. Work continues with Bill's firm imprint on the next edition, which is the focus of this discussion.

Invited Papers

9:15

2aSP4. Environmental factors in shallow water active sonar design. Peter Cable (135 Four Mile River Rd., Old Lyme, CT 06371, petercable@att.net)

In 1991, when William Carey was at DARPA, he initiated an effort under the Adverse Environments Program to both establish and define the limits of low frequency active sonar performance in shallow water and littoral regions, and to demonstrate a system concept that could achieve limiting performance. In support of that objective sea tests were conducted in the West Florida Shelf (Gulf of Mexico), Northeast U.S. Continental Shelf and Korea Strait (Area Characterization Tests I, II, and III), each in ~100m deep water with sandsilt bottoms under downward refracting sound conditions. The design, conduct and subsequent analyses of those tests, including consideration of broadband sound transmission (100 Hz–1 kHz), signal dispersion and coherence, reverberation, bottom scattering strength, and clutter, will be traced with particular emphasis on Bill's leadership role in the endeavor. The key results of the program were sufficiently robust to guide the design of tactically significant shallow water active sonar, which they ultimately did.

9:30

2aSP5. Model-based underwater signal processing—The Carey factor. James V. Candy (Eng., Lawrence Livermore National Security, PO Box 808, L-151, Livermore, CA 94551, tsoftware@aol.com) and Edmund J. Sullivan (Prometheus, Inc., Newport, RI)

The sustained encouragement and belief in the model-based approach to underwater processing has always been one of the favorite topics of Bill Carey's conversation. Not only by his direct encouragement, but also his contributions in the form of well-executed, well-controlled and well-documented experiments in the Hudson Canyon area off of the New Jersey coast that has become known as the best and most complete sets of oceanic data available for signal processors to apply their latest algorithms. It has become affectionately known as the "canonical" oceanic signal generator and often stated by many signal processors that "if your algorithm is not capable of performing well on the Carey Hudson Canyon data, then it is not worthy of pursuing it further." His experimental work is a major contribution to the underwater processing area. In this paper, we briefly discuss the Hudson Canyon data set and show the performance of a model-based processor that was applied to localize a source using a 23-element hydrophone array in shallow water.

Contributed Papers

9:45

2aSP6. Bill Carey and sound attenuation in marine sediments. Ross Chapman (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

Experiments carried out by Bill Carey in the Hudson Canyon off the New Jersey coast provided a great wealth of data for the study of low frequency sound propagation in shallow water. His analysis of the transmission loss data, reported in a series of papers dating from the mid 1990s, indicated a non linear frequency dependence of sound attenuation in the sediment material. This work provided a large base of experimental evidence for the non linear dispersion predicted by the Biot theory of sound propagation in porous media, and it stimulated new studies on sound propagation in marine sediments by many researchers, including Bill Carey himself. This paper reviews the results obtained by Carey from the Hudson Canyon experiments and places them in the context of new results from more recent experiments using different techniques and observables. Analysis of results from the new work indicates that Carey's observation of non linear frequency dependence in sediment material in the Hudson Canyon applies to attenuation of sound in different types of marine sediments.

10:00–10:15 Break

10:15

2aSP7. Spatial variation of seabed acoustic bulk properties. Nicholas P. Chotiros, Marcia J. Isakson, James N. Piper, and Andrew R. McNeese (Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

The seabed is modeled as a poro-elastic medium with a rough interface. The spatial variation of bulk properties, along with the interface roughness, are important contributors to the acoustic scattering strength of the seabed. Their effects are often indistinguishable. While roughness may be measured directly, the variability in the bulk properties is more difficult to obtain. In a recent experiment over a sandy seabed off Panama City, FL, known as the target and reverberation experiment of 2013 (TREX13), the seabed roughness and the normal acoustic reflection loss were simultaneously measured using a laser profiler and a short range acoustic sounder deployed aboard a remotely operated vehicle (ROV). Using the measured roughness statistics, the fluctuations in acoustic reflection loss due to roughness were estimated. Subtracting the roughness contribution from the total measured reflection fluctuations, the component due to bulk property changes was estimated, from which the fluctuation in the bulk properties may be inverted. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

Invited Paper

10:30

2aSP8. In memory of Bill Carey. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

Bill Carey has been a strong and influential mentor to many young and not-so-young acousticians and engineers. He had a profound effect on my career, encouraging me to publish my work and to attend Acoustical Society Meetings. His work in bottom acoustics, noise directionality, and surface effects, in particular, provided significant input to my research. He was also instrumental in nominating me for Fellowship in the Acoustical Society. I am deeply grateful and appreciate having an opportunity to remember him in this session.

Contributed Papers

10:45

2aSP9. Measuring seismic waves using a towed underwater acoustic array. Jon M. Collis (Colorado School of Mines, 1500 Illinois St., Golden, CO 80401, jcollis@mines.edu) and Allan D. Pierce (Retired, East Sandwich, MA)

This paper discusses the possibility of detecting shear and interface (Scholte) wave effects in the ocean using a towed hydrophone array. The shear field will be evanescent in the water and so may only be detected near to the ocean bottom interface. A benefit of measuring the acoustic field with a towed array is that a Hankel transform can be used to construct the horizontal wavenumber spectrum. If a shear or interface wave is measured, then it will be visible in the horizontal spectrum. The possibility of detecting the shear field will be strongly dependent on the shear wave speed in the sediment and this will also affect the detection of the more difficult to detect Scholte wave, which travels at about 90 % the shear wave velocity. The Scholte wave has a circular polarization where the shear wave may be vertical, horizontal, or a combination of the two polarities and may not be detectable for all frequency and source depth configurations. [Work supported by ONR.]

11:00

2aSP10. Bill Carey and the development of our understanding of ocean acoustic coherence. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

One of the many topics for which Bill Carey had an enormous passion was coherence. Unlike many researchers who elect to stay in either the observational or theoretical sides, Bill jumped into both areas with zeal, uncovering important navy data and working with the theorists of the day. This talk will summarize some of Bill's most seminal work in the area of coherence, and the talk will demonstrate how Bill's ideas live on in recent developments on the subject.

11:15

2aSP11. A brief history of the modeling of sound propagation in bubbly liquids. Craig N. Dolder, Preston S. Wilson, and Mark F. Hamilton (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

William M. Carey is well known for his interest in sound propagation through bubbly liquids. He was also a champion of re-attributing the low frequency effective medium model widely known as Wood's law to its original author Arnulph Mallock, who published a paper titled "The Damping of Sound by Frothy Liquids" in 1910. In the same spirit, this presentation will discuss the evolution of theories involving sound propagation through bubbly liquids over time from Mallock to modern day. Since bubble pulsations can exhibit strong nonlinearity, the presentation will conclude by reintroducing another often-overlooked modeling advance, at least in the western literature, that of Zabolotskya and Soluyan [Sov. Phys. Acoust. **13**, 254–256 (1967)] describing the nonlinear propagation of sound in bubbly liquids. [Work supported by ONR.]

Invited Paper

11:30

2aSP12. From Wood to Carey to Mallock: A review of Bill Carey's work associated with the Mallock-Wood equation and the acoustics of bubbly liquids and gas-bearing sediments. Preston S. Wilson, Craig N. Dolder (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, pswilson@mail.utexas.edu), Ronald A. Roy (Dept. of Eng. Sci., The Univ. of Oxford, Oxford, United Kingdom), and Allan D. Pierce (Dept. of Mech. Eng., Boston Univ., Boston, MA)

One of Bill Carey's many scientific interests throughout his career was the acoustics of bubbly liquids. Many underwater acousticians know of Carey's work associated with bubble clouds and more recently, gas-bearing sediments, but Bill got his start with the subject earlier in his career, studying flow in a boiling water reactor while employed at Argonne National Laboratory. Here, the acoustic velocity of bubbly liquid is of interest because of the possibility of supersonic flow, at comparatively low flow rates, in the high-void-fraction mixture within the reactor's piping system. In this talk, an overview of Bill's work with the acoustics of bubbly liquids will be presented, including scattering from bubble clouds, and sound propagation within bubbly liquid and gas-bearing sediments. Finally, Bill's campaign to rename a famous equation (Wood's Equation) in honor of its forgotten originator (Mallock) will be reviewed. [Work supported by ONR.]

Contributed Paper

11:45

2aSP13. A dipole source in a free surface and its representation as a near-surface monopole. Richard B. Evans (College of Eng., Boston Univ., 99F Hugo Rd., North Stonington, CT 06359, rbevans@99main.com)

A delta function impact on a free surface creates a dipole radiation pattern. This idealized source is of interest in the study of underwater ambient noise, in connection with noise caused by breaking waves, spray, and rain. The radiation pattern, due to the delta function impact, is derived and identified with the partial derivative of a fundamental monopole solution, or Green's function. Underwater acoustical models usually employ a monopole source. The representation of the surface dipole by a near-surface monopole is, therefore, a convenient approximation. The approximation of the dipole by a near surface monopole is an application of numerical differentiation. The reason for a distance of one quarter of a wavelength in the finite difference is described.

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems,
and
IEC/TC 29, Electroacoustics

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

M.A. Bahtiarian, Chair, U.S. Technical Advisory Group for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

W. Madigosky, Chair of the U.S. Technical Advisory Group for ISO/TC 108 Mechanical vibration, shock and condition monitoring
MTECH, 10754 Kinloch Road, Silver Spring, MD 20903

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

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

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

D.J. Vendittis, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machine systems
701 Northeast Harbour Terrace, Boca Raton, FL 33431

P.J. Battenberg, U.S. Technical Advisor for IEC/TC 29, Electroacoustics
3M Personal Safety Division, Detection Solutions, 1060 Corporate Center Drive, Oconomowoc WI 53066

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

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S1, which will be held on Monday, 5 May 2014 from 5:15 p.m. - 6:30 p.m.

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

Tuesday, 6 May 2014 11:00 a.m.-12:30 p.m. ASC S12, Noise
Tuesday, 6 May 2014 2:00 p.m. - 3:30 p.m. ASC S3, Bioacoustics
Tuesday, 6 May 2014 3:45 p.m. - 5:00 p.m. ASC S3/SC 1, Animal Bioacoustics

Accredited Standards Committee S2, Mechanical Vibration and Shock, is not scheduled to meet in Providence.

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SCI** Noise	ASC S12
M.A. Bahtiarian, Chair	**ISO/TC 43/SC 3**, Underwater acoustics	ASC S1, ASC S3/SC 1 and ASCS12
W. Madigosky, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
W.C. Foiles, Co-Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
P.J. Battenberg, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

Meeting of Accredited Standards Committee (ASC) S12 Noise

W.J. Murphy, Chair, ASC S12
NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

S.J. Lind, Vice Chair, ASC S12
The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse WI 54601-7599

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 6 May 2014.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

TUESDAY AFTERNOON, 6 MAY 2014 555 A/B, 1:05 P.M. TO 5:10 P.M.

Session 2pAAa

Architectural Acoustics: Uncertainty in Describing Room Acoustics Properties II

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

Ingo B. Witew, Cochair
Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany

Chair's Introduction—1:05

Invited Papers

1:10

2pAAa1. Measuring speech intelligibility using impulse responses: Impact of the decay range on the speech transmission index. Constant Hak (Bldg. Phys. and Services, Eindhoven Univ. of Technol., Den Dolech 2, Eindhoven 5600 MB, Netherlands, c.c.j.m.hak@tue.nl) and Remy Wenmaekers (Level Acoust., Eindhoven, Netherlands)

IEC 60268-16 describes how to measure the Speech Intelligibility Index STI and its simplified derivatives such as STITEL and STIPA, using two measurement techniques. The first and oldest technique is based on a set of modulated noise signals used as a stimulus. The second method uses impulse responses obtained from maximum length sequence (MLS) or swept sine stimuli, deconvolution and Schroeder's Modulation Transfer Function (MTF). This technique is gaining more and more ground through advancing technology. The ability to modify the background noise component of a measured impulse response is one of the advantages of this technique, but requires a certain minimum impulse response quality. A measure of the quality of an impulse response is its decay range. The influence of the impulse response decay range on the calculated STI value is investigated. As in a previous study on the ISO 3382-1 parameters, this is done by using the Impulse response to Noise Ratio INR as an estimator for the decay range. The result is a proposal for the minimum required decay range to accurately measure the STI, based on the Just Noticeable Difference JND and the INR.

1:30

2pAAa2. Uncertainties in speech transmission index measurements. Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

A detailed study has been carried out into the uncertainties associated with Speech Transmission Index (STI) measurements. The uncertainties and measurement errors are shown to be either systematic or random in nature. Systematic errors were found to include limitations of the technique itself as well as uncertainties related to measurement hardware and software implementations. Systematic errors were found to be caused by a range of issues including measurement microphone properties, test signal generation and replay errors, talker loudspeaker directivity and frequency response variations, and measurement system algorithms. Some forms of digital signal processing are also shown to affect the measured result and were found to be highly dependent upon the nature of the test signal and its processing. Measurement uncertainties due to random errors were found to include out of band, high sound pressure level, low frequency modulations or overloading of the microphone preamplifier stages, the pseudo random nature of the STI test signal itself, and the sensitivity of some test signals, such as maximal length sequences, to short term environmental and acoustic changes. Changes and fluctuations in the background noise level during measurements were also found to be a significant cause of error and uncertainly.

1:50

2pAAa3. Evaluation and improvement of a model to predict the measurement uncertainty due to the directivity of room acoustical sound sources. Ingo B. Witew, Mark Mueller-Giebeler, and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, ingo.witew@akustik.rwth-aachen.de)

With the aim to reduce the necessary efforts to empirically determine the uncertainty in room acoustical measurements, in previous work, a model was developed that can predict the uncertainty a directivity of a sound source introduces to a measurement. As part of the validation extensive series of scale measurements have been conducted. In this contribution, the predicted uncertainty based on simulations and the empiric data is compared to each other. The results were used to improve the model. Concluding it will be discussed whether the model is suitable for a reasonable measurement uncertainty discussion.

2:10

2pAAa4. Measurement repeatability of late lateral energy level and lateral energy fraction. David A. Dick and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dad325@psu.edu)

Late lateral energy level (GLL) and lateral energy fraction (LF) are two room acoustics measures that have been shown to correlate with certain aspects of spatial impression in concert halls. The purpose of this study was to investigate the repeatability of GLL and LF measurements. A custom microphone stand was built that can be adjusted in each spatial dimension separately to allow for accurate and precise microphone placement. Room impulse responses (IRs) were measured at six receiver locations in a 2500-seat auditorium using two different methods to obtain the lateral energy IR: the beamformed dipole response from a spherical microphone array, and a studio-grade figure-of-eight microphone. Three sets of IR measurements were taken at each receiver location. In between sets, the microphone stand was removed and the various adjustment points were randomly repositioned. The stand was then replaced in the same position to re-measure the IRs. The variability between the measurements at each receiver location was found to be relatively low for GLL (the standard deviation ranged between 0.22 and 0.73 JNDs for the 125–1000 Hz sum), and higher for LF (the standard deviation ranged between 0.49 and 2.80 JNDs for the average over the 125–1000 Hz octave bands). [Work supported by NSF Grant 1302741.]

2:30

2pAAa5. Using narrowly defined hypotheses to extract meaningful results from broad data sets. Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

Data sets come in many forms, from modeled or calculated results, to *in-situ* measures, to bodies of work that provide collected sets of evidence. The collective data set can often be overwhelming, and can provide conflicting results, or present a level of uncertainty in analysis. Narrowly defining the questions to ask of the data based on the strengths and weaknesses of each tool seemingly enables discovery of the needle(s) in the haystack. Listening experiences in live settings provide the hypotheses that steer the analysis utilizing all of the tools at our disposal. Experience with targeted analysis of very specific phenomena are discussed from recent projects.

2:50

2pAAa6. Uncertainty in room acoustics: A consultant's perspective. Benjamin Markham and Jonah Sacks (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Faced with time and budget constraints on most projects, acoustical consultants make a series of choices when asked to evaluate the acoustics of an existing space. Measure the impulse response of the room with sophisticated technology (MLS, sine-sweeps, etc.), or pop a balloon? Use a dodecahedral loudspeaker (or several), a more standard portable amplified loudspeaker, or the house sound system (or pop a balloon)? Use a single omni-directional microphone, a binaural dummy head, or a B-format soundfield microphone? How many source locations, and how many receiver locations? Whither measure? There is a growing body of literature that identifies uncertainty and inconsistency in reverberation time and other room acoustics metrics and the tools used to obtain them. The authors will add modestly to this evidence from consulting experience, and then focus on how some consultants in architectural acoustics choose what data to gather, how to gather it, and to what extent. Considerations of time, budget, and logistics factor in those choices, as do determinations regarding the essence of the client's needs and concerns, subjective listening to determine what measurements could be most useful, and our own expectations for how we might use the measurements we make, now and in the future.

3:10–3:25 Break

3:25

2pAAa7. Evaluation of acoustic diversity of religious buildings; case study from churches and mosques in Turkey. Filiz B. Kocyigit (Architecture, Atilim Univ., Kizilcasar Mah. Incek, Ankara 06560, Turkey, filizbk@gmail.com), Gamze Akbaş, Gokhan Ozal, Can Yerli (Fine Art Design and Architecture, Atilim Univ., Ankara, Turkey), Meral Gunduvan (Fine Art Design and Architecture, Atilim Univerity, Ankara, Turkey), and Feyyaz Demirer (English Education, Harran Univ., Urfa, Turkey)

In this study, by analyzing the reverberation time and noise insulation features of religious structures, effects of differences on the forms, materials, and functions on interior acoustics are discussed. For this purpose, calculations were made working on the projects of the selected sample mosques and churches built by different materials and their interior reverberation times were measured and compared. Two different types of areas were compared using RT60 reverberation time calculations and T30 and T20 frequency band analysis measurement systems. In the comparison, interior functions are evaluated. In this evaluation, it was observed that in mosques since there is excessive speaking, medium frequency resulting from people was more dense whereas in churches high frequency was also involved with rite taking place alongside the medium frequency resulting from talking. Therefore, interior frequency band width was observed to increase. Selected churches have cross and basilic type plan and mosques have square type plan. It is observed that selected traditional mosque roofs have circular forms and churches have sharp forms. Hereby variations of the acoustic features of the sites with different forms and functions used for similar purposes were compared.

3:40

2pAAa8. The development and analysis of a large variable acoustics space. Jay Bliefnick (Acoust., Columbia College, 5001 River Rd., Apt. 1S, Schiller Park, IL 60176, jay.bliefnick@loop.colum.edu), Andrew Hulva, and Dominique Cheenne (Acoust., Columbia College, Chicago, IL)

A new, large-scale variable acoustics space has recently been added to the Audio Arts & Acoustics department at Columbia College Chicago. Built within the current Motion Capture studio, this facility will provide students and faculty the ability to perform tests in an acoustically controlled environment, without the influence of small-room effects. The construction involved the creation of nearly 300 2 in. x 2 in. reversible boxes: one side diffusive and the other absorptive. These line three full walls of the space at a height of 10 in., totaling ~1200 ft^2 of acoustically adjustable surface area. This allows the room to convert from a very absorptive space, to one that is much more acoustically active. Multiple specular reflector panels are also available for the creation of "hot spots," allowing for even more diverse applications. This study focused on the construction and initial testing of this innovative new space. To analyze the effectiveness of the additions, frequency, time, and reverberation responses for the entire room were sampled in a variety of configurations: fully absorptive, fully diffusive, empty, etc. These objective metrics were then analyzed against perceptual data to determine the correlation between what could be measured and what could be heard.

3:55

2pAAa9. Evaluating methods of acoustic analysis in a small listening room. Jennifer Levins (SoundSense, LLC, 2669 E Thompson St., Philadelphia, PA 19125, jenlevins@gmail.com)

Although common, octave band analysis of decay times provides limited information on the behavior of sound in a room. In spaces for critical listening, additional analysis is required in the design phase. Even in existing spaces, measurements of reverberation time do not always reveal room anomalies such as echoes and standing waves. The acoustic experience of the space is not always indicated by reverberation time. In rooms for critical listening, additional methodologies such as modal analysis, ray tracing, and early reflection times are used to determine additional information about a room's acoustic characteristics. A case study will be presented to demonstrate the applicability of these methods in small listening rooms.

4:10

2pAAa10. Experimental verification of computer modeled loudspeaker sound level performance based on excitation signal. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Vahid Naderyan (Phys., Univ. of MS, University, MS)

An investigation of a loudspeaker sound level performance in a simple geometry to verify the EASE sound level definition comparing pink noise and multi-tone signal utilizing field tests of loudspeakers.

4:25

2pAAa11. Efficient computational modeling of Platonic solid loudspeaker directivities. Jeshua H. Mortensen and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, meako490@gmail.com)

Dodecahedron loudspeakers are commonly used in architectural acoustics measurements as quasi-omnidirectional sources of sound. Other Platonic solid loudspeakers may also be used for this purpose, but the geometrical properties that optimize their omnidirectional behaviors are not well understood. Because of the difficulties in constructing large numbers of loudspeakers and boundary element models for related investigations, a computational method has been developed for the MATLAB environment to rapidly predict radiated fields and observe general directivity trends as geometrical properties vary. The method is based on related spherical enclosure geometries. It enables one to easily assess the effects of altered driver diameters, positions, numbers, vibrational patterns, and enclosure volumes. This presentation discusses the tool and presents several computational findings. Its output is presented as animated frequency-dependent balloon plots and area-weighted spatial standard deviations. The results are found to agree well with similar predictions for actual Platonic solid geometries from the boundary element method and experimental measurements.

4:40

2pAAa12. A closer look at the Hopkins-Stryker equation. Timothy W. Leishman and Zachary R. Jensen (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, twleishman@byu.edu)

The Hopkins-Stryker equation has long been used to represent sound fields in semi-reverberant rooms. However, its implementation could be improved if users were more familiar with its origins, assumptions, and potential applications. The directivity factor, distance from the acoustic center, room constant, and locally averaged energy density are all key elements of the equation that merit special attention. This presentation explores these quantities and their theoretical underpinnings. It also introduces the use of generalized energy density as a means of simplifying averaging requirements. Selected numerical examples serve as illustrations to clarify the concepts.

4:55

2pAAa13. A two-point method for direct measurement of the room constant. Zachary R. Jensen and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., ESC, Provo, UT 84604, zjens1@gmail.com)

The room constant is a key frequency-dependent value that is widely used to characterize reverberant fields. It is typically estimated from room boundary properties, viz., total surface area and average absorption coefficient. Unfortunately, these properties are often difficult to ascertain with sufficient accuracy. While reverberation times may be adequately measured using modern methods, the effective surface areas and volumes of many practical rooms are elusive. Furthermore, several formulations for the room constant exist without general agreement as to their best usage. This presentation introduces a two-point energy-based method that enables acousticians to feasibly measure the room constant without knowledge of the room volume, surface area, or average absorption coefficient. Numerical simulations and measurements illustrate the benefits of the approach. Resulting values are compared with those approximated using common formulations for validation and clarification.

Session 2pAAb

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition (Poster Session)

Norman H. Philipp, Chair
Geiler and Associates, LLC, 1840 E. 153rd. Cir., Olathe, KS 66062

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2014 Student Design Competition that will be professionally judged at this meeting. The 2014 design competition involves the design of a fine arts building for a high school of moderate size primarily for a school's strong opera program. The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1,250 will be made to the submitter(s) of the design judged "first honors." Four awards of UDS$700 each will be made to the submitters of four entries judged "commendation."

TUESDAY AFTERNOON, 6 MAY 2014 554 A/B, 1:00 P.M. TO 5:00 P.M.

Session 2pAB

Animal bioacoustics: Acoustics as a Tool for Population Structure I

Kathleen Stafford, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Shannon Rankin, Cochair
Southwest Fisheries Science Ctr., 8901 La Jolla Shores Dr., La Jolla, CA 92037

Chair's Introduction—1:00

Invited Papers

1:05

2pAB1. Geographic variation in acoustic signals of freshwater fishes. Carol Johnston (Fisheries, Fish Biodiversity Lab., Auburn Univ., Swingle Hall, Auburn, AL 36830, johnsc5@auburn.edu), Catherine Phillips (US Fish and Wildlife Service, Panama City, FL), and Patty Noel (Fisheries, Auburn Univ., Auburn, AL)

Although well studied in other taxa, geographic variation in signal structure has been poorly studied in fishes. Signal divergence in marine fishes depends on larval form; pelagic larvae disperse more widely than demersal forms, limiting opportunities for isolation and subsequent divergence. Freshwater fishes, especially those restricted to headwater habitat, are isolated by drainage. There are many examples of species radiations in groups of North American freshwater fishes within drainage networks, including darters, minnows, and catfishes, some of which are restricted to single streams. Our data demonstrate divergence of acoustic signals among populations of stream and riverine fishes at multiple scales, and often in the absence of apparent morphological variation. Two model species, Longear Sunfish and Whitetail Shiner, differed in the temporal components of calls, while darter and sturgeon models showed variation in both temporal and spectral call components. In the case of the sturgeon, the populations were genetically distinct. Furthermore, data for Whitetail Shiner suggest that calls associated with courtship were strongly associated with geographic isolation, while divergence in those characteristics associated with aggression may be driven by genetic drift. We suggest that variation in acoustic signal structure may be common in freshwater fishes and discuss implications for mate choice.

1:25

2pAB2. Sources of acoustic variation in the advertisement vocalizations of Neotropical singing mice. Bret Pasch (Dept. of Integrative Biology, Univ. of Texas at Austin, Austin, TX 78712, bpasch@utexas.edu), Polly Campbell (Dept. of Zoology, Oklahoma State Univ., Stillwater, OK), Mustafa Z. Abbasi, Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, Austin, TX), Steven M. Phelps, and Michael J. Ryan (Dept. of Integrative Biology, Univ. of Texas at Austin, Austin, TX)

Patterns of variation in communication systems provide important insight into the processes that shape phenotypic evolution. Although studies in anurans, birds, and insects indicate that diverse selective and stochastic forces influence acoustic signals, factors that shape variation in mammalian vocalizations are poorly understood. Neotropical singing mice (*Scotinomys*) are diurnal, insectivorous rodents distributed throughout montane cloud forests of Middle America. Males commonly emit species-specific advertisement vocalizations that are used in mate attraction and male-male aggression. To explore factors contributing to vocal variation, we summarize data from a diversity of studies at disparate scales and levels of analysis. We highlight the importance of genetic drift in shaping population differentiation, the role of androgens in modulating the performance of physically challenging displays, the influence of social context in shaping posture and vocal parameters, and the impact of the ambient environment in affecting sound propagation. Neotropical singing mice are emerging as an important model that enables us to draw parallels to vocal communication systems in traditionally more tractable species.

1:45

2pAB3. Vocal diversity and taxonomy of white-cheeked crested gibbons. Julia Ruppell (Biology, Pacific Univ., 3920 SW Alice St., Portland, OR 97219, Ruppell@pacificu.edu)

Previous research suggests that gibbon song repertoire is genetically determined and song characteristics are useful for assessing systematic relationships. The taxonomy and distribution of crested gibbons (genus *Nomascus*) had not been studied previously. In addition, crested gibbons face several threats to extinction such as habitat loss, the pet trade, the domestic and international trade of wildlife, and unsustainable harvest of wildlife for subsistence consumption. Additionally, rice field expansion and poaching pose significant threats to gibbons and their habitats, especially because most populations are very small. I studied vocal diversity among different wild populations of *Nomascus* in Vietnam and Laos to assess their taxonomic relationships and to examine whether their vocal patterns correspond to forms previously described. Linear discriminant analysis, classification trees, and multidimensional scaling revealed distinct populations based on song characteristics and species (or subspecies) boundary locations were recognized. In addition, population sizes in different areas were estimated based on vocal analysis. The recognition of previously unknown diversity within *Nomascus* and ability to locate potential species boundaries aided in implementation and adaptation of gibbon conservation strategies and the development of gibbon management plans for protected areas. In addition, this project improved the monitoring of a poorly known and understudied ape, by working with provincial staff and local people.

2:05

2pAB4. Behavioral and phylogenetic differentiation in a potential cryptic species complex, the canyon treefrog. Katy Klymus, Carl Gerhardt, and Sarah Humfeld (Univ. of Missouri, 302 ABNR, Columbia, MO 65201, klymusk@missouri.edu)

Detection of genetic and behavioral diversity within morphologically similar species has led to the discovery of cryptic species complexes. We tested the hypothesis that the canyon treefrog (*Hyla arenicolor*) may consist of cryptic species by examining mate-attraction signals among highly divergent lineages defined by mitochondrial DNA (mtDNA). Unexpectedly, calls exhibited little variation among the three U.S. lineages despite large mtDNA sequence divergences. We re-analyzed intraspecific and interspecific phylogenetic relationships by sequencing both mitochondrial and nuclear genetic markers among populations and a closely related, but morphologically and behaviorally different species, the Arizona treefrog (*H. wrightorum*). Discordance between mitochondrial and nuclear datasets suggests multiple instances of introgression of *H. wrightorum's* mitochondrial genome into populations of *H. arenicolor*. Furthermore, intraspecific population structure based on nuclear markers shows better congruence with patterns of call variation than population structure based on the mitochondrial dataset. Although the U.S. lineages do not appear to represent cryptic species, Mexican lineages do show biologically relevant call differences as assessed through female preference tests. Our results suggest that call variation can indicate genetic structure of populations; however, a multilocus approach should be used in defining genetic structure, as using only mtDNA may lead to erroneous conclusions.

2:25

2pAB5. Population structure of humpback whales in the western and central South Pacific Ocean determined by vocal cultural exchange. Ellen C. Garland (National Marine Mammal Lab., AFSC/NOAA, AFSC/NOAA, 7600 Sand Point Way NE, Seattle, WA 98115, Ellen.Garland@noaa.gov), Michael J. Noad (Cetacean Ecology and Acoust. Lab., School of Veterinary Sci., Univ. of Queensland, Gatton, QLD, Australia), Anne W. Goldizen (School of Biological Sci., Univ. of Queensland, St. Lucia, QLD, Australia), Matthew S. Lilley (SecuritEase Int.., Petone, New Zealand), Melinda L. Rekdahl (Cetacean Ecology and Acoust. Lab., School of Veterinary Sci., Univ. of Queensland, Gatton, QLD, Australia), Claire Garrigue (Opûration Cûtacûs, Noumea, New Caledonia), Rochelle Constantine (School of Biological Sci., The Univ. of Auckland, Auckland, New Zealand), Nan Daeschler Hauser (Cook Islands Whale Res., Avarua, Rarotonga, Cook Islands), M. Michael Poole (Marine Mammal Res. Program, Maharepa, Moorea, French Polynesia), and Jooke Robbins (Provincetown Ctr. for Coastal Studies, Provincetown, MA)

Male humpback whales (*Megaptera novaeangliae*) produce a continually evolving vocal sexual display, or "song," which is shared by all males within a population. The rapid cultural transmission of this display between distinct but interconnected populations within the western and central South Pacific region presents a unique opportunity to investigate population connectivity based on a vocal display. Quantitative analyses were conducted on eleven years of data to investigate vocal groupings based on song types shared between populations, to produce an acoustically derived population structure for the region. Four distinct vocal groupings resulted; the western group contained a single population, off eastern Australia, the central group was comprised of whales around New Caledonia, Tonga and American Samoa, and finally the whales of the eastern region were split into two groups, one around the Cook Islands and the other in the waters of French Polynesia. These groupings broadly agree with results obtained using genetic and photo-identification methods, and confirm that humpback whales are likely to form separate breeding populations rather than panmictic subpopulations. This study demonstrates the utility of using culturally transmitted vocal patterns as a way of defining populations, at least in this species.

2:40

2pAB6. Using passive acoustics to investigate seasonal and diel trends in acoustic behavior of North Atlantic right whales (*Eubalaena glacialis*). Leanna P. Matthews, Jessica A. McCordic, and Susan E. Parks (Biology, Syracuse Univ., 227 Life Sci., 107 College Pl., Syracuse, NY 13244, lematthe@syr.edu)

The North Atlantic right whale (*Eubalaena glacialis*), an endangered baleen whale species, produces a variety of stereotyped acoustic signals. One signal, the "gunshot" sound, has only been recorded from adult males and is thought to function for reproduction, either as advertisement for females or an agonistic signal toward other males. This study uses remote acoustic monitoring to analyze the presence of gunshots over a two-year period at two sites on the Scotian Shelf to determine if there is evidence that right whales use these locations for breeding activities. Seasonal analyses at both locations indicate that gunshot production is highly seasonal, with an increase in autumn. One site had significantly more gunshot sounds and exhibited a clear diel trend in signal production. The other site also showed a seasonal increase during autumn, but did not show any significant diel trends. This site difference indicates variation either in the number or the behavior of whales at each location. The timing of the observed seasonality in gunshot production is consistent with the current understanding of right whale breeding season, and these results demonstrate that detection of gunshots with remote acoustic monitoring can be a reliable way to track seasonal mating activities.

2:55–3:15 Break

3:15

2pAB7. Variation in the acoustic behavior of right whale mother-calf pairs. Susan Parks (Dept. of Biology, Syracuse Univ., 107 College Pl., Syracuse, NY 13244, sparks@syr.edu), Lisa Conger (NOAA Fisheries, Northeast Fisheries Sci. Ctr., Woods Hole, MA), Dana Cusano (Dept. of Biology, Syracuse Univ., Syracuse, NY), and Sofie Van Parijs (NOAA Fisheries, Northeast Fisheries Sci. Ctr., Woods Hole, MA)

North Atlantic right whale mother-calf pairs are a critical segment of the population for recovery of this endangered species and therefore their protection is paramount. Passive acoustics, including the development of real-time buoys, has increasingly played a role in the detection of right whale presence in high vessel traffic areas. The vocal behavior of North Atlantic right whale mothers and their young calves has not been well described and may not be well represented by calls produced by other individuals in the population that are commonly utilized for passive acoustic monitoring. Therefore, it is critical to determine the call types and rates of sound production by mother-calf pairs to assess the efficacy of passive acoustic monitoring for their detection. We conducted behavioral focal follows coupled with acoustic recording of right whale mother calf pairs off the coast of Florida and Georgia in January–March, Cape Cod Bay in April, and the Bay of Fundy in August–September from 2011 to 2014. Results show modifications in both call structure and call rate with increasing calf maturity and independence. These data are necessary to better utilize passive acoustic monitoring for management purposes in this species.

3:30

2pAB8. Western and central North Atlantic fin whale (*Balaenoptera physalus*) stock structure assessed using geographic song variations. Julien Delarue (JASCO Appl. Sci., 202 - 32 Troop Ave., Dartmouth, NS B3B 1Z1, Canada, julien.delarue@jasco.com), Robert Dziak, David Mellinger (PMEL/NOAA, Newport, OR), Jack Lawson (Fisheries and Oceans Canada, St. John, NF, Canada), Hilary Moors-Murphy (Fisheries and Oceans Canada, Dartmouth, NS, Canada), Yvan Simard (Fisheries and Oceans Canada, Mont-Joli, QC, Canada), and Kathleen Stafford (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Variations in calls or songs between areas are increasingly acknowledged as a way to assess stock structure. We present the results of an analysis of 163 fin whale (FW) songs recorded in seven areas of the North Atlantic (NA): Irminger Sea, Davis Strait, Grand Banks of Newfoundland, Southern Newfoundland, Gulf of St. Lawrence, eastern Scotian Shelf, and the waters off Delaware Bay. Song measurements included inter-note intervals (INI), notes' peak frequency and bandwidth and note type (classic, backbeat, and high-frequency) proportion. Seasonal patterns of classic-classic INI provided the highest level of differentiation between areas and revealed the existence of six acoustic stocks. Classification trees revealed that other parameters distinguished between regions over larger spatial scales, grouping some of the recording areas together. These results suggest that (1) there are four distinct stocks in the western NA; (2) the range of a presumed central NA stock includes southwestern Iceland, both sides of Greenland and appears to extend south along the Mid-Atlantic Ridge, at least in recent years; (3) two stocks are present off West Greenland. These results bring new information on potential FW stock delineations in the NA. The latter will be compared to those derived using other stock assessment metrics.

3:45

2pAB9. Do spectral features of Risso's dolphin echolocation clicks vary geographically? Melissa Soldevilla, Lance Garrison (NOAA Southeast Fisheries Sci. Ctr., 75 Virginia Beach Dr., Miami, FL 33149, melissa.soldevilla@noaa.gov), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), Danielle Cholewiak, Sofie Van Parijs (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), Lynne Hodge, Andrew Read (Duke Univ. Marine Lab., Beaufort, NC), Erin Oleson (NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), and Shannon Rankin (NOAA Southwest Fisheries Sci. Ctr., La Jolla, CA)

The ability to classify odontocetes to species and population from acoustic recordings leads to improvements in stock identification, abundance and density estimation, and habitat-based density modeling, which are crucial

for conservation and management. Risso's dolphins off Southern California have distinctive peaks and valleys in their echolocation clicks, which allow researchers to easily distinguish them from other species in passive acoustic recordings. However, Risso's dolphin echolocation clicks from other geographic areas have not been described and it remains unknown whether they have similarly distinctive click spectra and whether stocks are acoustically distinct. We investigate the potential for using acoustics to identify populations by quantifying the acoustic structure of Risso's dolphin echolocation clicks recorded over wide-ranging geographic regions including the U.S. waters of the North Atlantic Ocean (north and south of Cape Hatteras), Gulf of Mexico, and North Pacific Ocean (Eastern Tropical Pacific and Southern California). Several distinctive peak and valley patterns are found and we evaluate these in light of variability within individuals, groups, and regions as these acoustic differences may indicate differences in age, sex, or stock composition, foraging behavior, or ambient noise environment of the dolphin schools.

4:00

2pAB10. Counting odontocetes from click train detections using multiple independent autonomous acoustic sensors. James A. Theriault (Ocean and Ecosystem Sci., Bedford Inst. of Oceanogr. , Dartmouth, NS, Canada), Craig Sheppard, Joey Hood (Akoostix, Inc., 10 Akerley Blvd., #12, Dartmouth, NS B3B 1J4, Canada, jhood@akoostix.com), Hilary B. Moors-Murphy (Ocean and Ecosystem Sci., Bedford Inst. of Oceanogr. , Dartmouth, NS, Canada), and Matthew Coffin (Akoostix, Inc., Dartmouuth, NS, Canada)

Passive acoustic monitoring (PAM) is often suggested as an effective technology to mitigate impacts from anthropogenic activities; however, the ability to reliably and efficiently detect, locate, and count cetaceans using PAM is still in development. One particularly useful application of PAM is species density estimation, which requires an estimate of the number of individuals involved in a detection event. Efforts have been undertaken to develop methods to directly count the number of vocalizing animals during acoustic detection events. For odontocetes, discrete clicks are almost indistinguishable between individuals, making it more difficult to determine the number of vocalizing animals as the number increases. Using recordings from multiple closely spaced ($\approx$200 m) GuardBuoy sensors deployed on the Canadian Scotian Shelf, cross-sensor correlograms were produced to estimate the number of individual sperm whales, and, as a more challenging case, the number of vocalizing delphinids. Using feature-based multipath reflection discrimination, the raw time series were reduced to a synthetic time series of binary click detections with the multipath arrivals removed. The synthesized click-detection time series were used for the cross-sensor correlograms to generate improved estimates of the number of vocalizing animals as compared with using the raw time series.

4:15

2pAB11. The acoustic structure of whistles as a tool for identifying evolutionary units in dolphins. Elena Papale, Marta Azzolin (Dept. of Life Sci. and Systems Biology, Univ. of Torino, via accademia albertina 13, Torino 10123, Italy, elena.papale@unito.it), Irma Cascao (Centro do Instituto do Mar (IMAR) da Universidade dos Açores, Departamento de Oceanografia e Pescas, Universidade dos Açores, Horta, Portugal), Alexandre Gannier (Groupe de Recherche sur les Cétacés, Groupe de Recherche sur les Cétacés, Antibes, France), Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kanehone, HI), Julie N. Oswald (Bio-Waves, Bio-Waves, Encinitas, CA), Monica Perez-Gil (Society for the Study of Cetaceans in the Canary Archipelago, Puerto Calero, Spain), Monica Silva (Centro do Instituto do Mar (IMAR) da Universidade dos Açores, Departamento de Oceanografia e Pescas, Universidade dos Açores, Horta, Portugal), and Cristina Giacoma (Dept. of LIfe Sci. and Systems Biology, Univ. of Torino, Torino, Italy)

Acoustic signals are expressions of phenotypic diversity and their variation could provide important information on differentiation patterns within species. Due to a number of selective pressures acting on signals, the contribution of genetic drift is often complex to outline. This study aims at evaluating if an examination of the acoustic structure of communication signals can allow the identification of evolutionary units in species capable of vocal learning. We quantified and compared parameters of whistles emitted by three dolphin species (*Stenella coeruleoalba*, *Delphinus delphis*, and *Tursiops truncatus*) to examine the hypothesis that acoustic signals permit the recognition of differentiation between populations from the Atlantic Ocean and the Mediterranean Sea. In the three species, recordings were correctly assigned to their basin of origin with a percentage higher than 82% by DFA. Frequency parameters were the most stable within each species. Where gene flow has been shown, i.e., within Atlantic Ocean, significant differences were found principally in modulation parameters. We hypothesize that these parameters are influenced by social and behavioral factors and that similar ecological conditions led to convergent acoustic features. Results of this study suggest that is it possible to recognize evolutionary units based on acoustic data.

4:30

2pAB12. Long-term acoustic surveying of bats. Annemarie Surlykke, Tûrur Andreassen (Biology, Univ. of Southern Denmark, Campusvej 55, Odense DK-5230, Denmark, ams@biology.sdu.dk), and John Hallam (Maersk-McKinney Moller Inst., Univ. of Southern Denmark, Odense, Denmark)

Increasing concern about decline in biodiversity has created a demand for population surveys. Long-term unmanned automatic monitoring may provide unique unbiased data from a whole season, but the large amount of data presents serious challenges for automatic processing. A two-month study of echolocating bats at 500 kHz sampling rate provided 236 GiB of data at full bandwidth. We used a Support Vector Machine (SVM) classier based on a combination of temporal and spectral analyses to classify events into bat calls and non-bat events. Duration, energy, bandwidth, and entropy were used to identify bat calls and reject short noise pulses, e.g., from rain. The SVM classifier reduced our dataset to 162 MiB of candidate bat calls with an estimated accuracy of 96% for dry nights and 70% when it was raining. The automatic survey revealed correlation between bat activity and rain, temperature, and sunset/sunrise. There were calls from two species new to the area, as well as an unexpected abundance of social calls. Future applications aim at higher accuracy in classifying bat calls and using trajectory-tracking to determine flight paths to correct for the bias toward loud bats inherent in acoustic surveying.

4:45

2pAB13. Small-scale soundscapes over coral reef ecosystems: Latitudinal variation in the Northwestern Hawaiian Islands. Simon E. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 3725 Miramar St. Apt. C, La Jolla, CA 92037, sfreeman@ucsd.edu), Lauren A. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA), Marc O. Lammers (Oceanwide Sci. Inst., Honolulu, HI), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

Two-dimensional seafloor "maps" of near-field ambient sound produced by biological sources in a coral reef environment were obtained from four spur-and-groove shallow water reef environments along a latitudinal gradient in the Papahanaumokuakea Marine National Monument, Northwestern Hawaiian Islands. Acoustic data were collected in conjunction with SCUBA based ecological surveys in an effort to establish correlation between components of the acoustic field and the ecological state of each field site. Simultaneous acoustic measurements and remote underwater photographs taken during the day and at night allowed for comparisons of biological activity with recordings over time. Using a bottom-mounted L-shaped array of hydrophones, the spatial distribution, frequency, and temporal characteristics of sounds produced by small-scale biological processes were estimated within a 40 m by 40 m region around the array. A fast cross-correlation guidance technique lessened the computational burden imposed by conventional and white noise constrained adaptive focusing (curved-wavefront beamforming). Likely sources of biological sound and variation of the acoustic field within and between the sample sites will be discussed. The variation of sound field properties over latitude and correlation with ecological information obtained at each site may be useful in underwater biological surveys that utilize passive acoustic recording.

Session 2pEA

Engineering Acoustics: Transduction

Dehua Huang, Chair
NUWC, 43 Holliston Ave., Portsmouth, RI 02871

Contributed Papers

1:30

2pEA1. Excitation of multi-modes of vibration using tangentially polarized thin walled cylinders. Sairajan Sarangapani (Rowe Technologies, Inc., 12136, Via Milano, San Diego, CA 92128, ssairajan@yahoo.com) and David A. Brown (Electro-Acoust. Res. Lab., Adv. Technol. and Manufacturing Ctr. and ECE Dept., BTech Acoust. LLC, Univ. of Massachusetts Dartmouth, Massachusetts, MA)

Tangentially polarized thin-walled striped-electroded piezoelectric cylindrical transducers are used in several electromechanical and electroacoustic applications. This study is an extension of the previous work [J. Acoust. Soc. Am. **133**, 2661 (2013); J. Acoust. Soc. Am. **132**, 3068 (2012)] and involves the study of electromechanical excitation of multi-modes of vibration using tangentially polarized cylinders. A numerical finite difference method (FDM) was used to analysis the nonuniform electric field in the tangentially polarized cylinder under the assumption that the piezoelectric element is fully polarized. The electromechanical properties including the effective electromechanical coupling coefficient, effective piezoelectric modulus, the effective compliance, and effective relative dielectric constant were calculated for a tangentially polarized cylinder for several modes of vibration and compared with results of cylinders using traditional transverse piezoelectric effect.

1:45

2pEA2. Statistical properties of random arrays. Jenny Au, Charles Thompson, and Ololade Mudasiru (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, Jenny_Au@student.uml.edu)

In this work, the statistical properties of the randomly placed linear arrays is considered. The performance of random, random-bin and nearest-neighbor constraint placement of transducer elements is considered. Transducer placement locations drawn from the transducer density function satisfying the van der Maas objective function is of particular interest. It is shown that the variance in the response of the sidelobe region decays asymptotically as 1/N, where N is the number of transducers.

2:00

2pEA3. Construction of a new underwater low-frequency projector based on clarinet acoustics. Andrew A. Acquaviva (Acoust., Penn State Univ., 871 Willard St., State College, PA 16803, acquavaa@gmail.com)

Understanding theoretical models of novel designs is a crucial component of developing early prototypes. This presentation discusses a new underwater low-frequency sound source. This system is based on a model of a clarinet-like source that is designed to work underwater, such that the complete assembly incorporates all of the necessary components required for a real clarinet to produce sound. The resulting device can, with proper set up and understanding of clarinet acoustics, produce a stable tone at significant amplitudes. Furthermore, the harmonic content substantiates the validity of the model on which it is based.

2:15

2pEA4. Design of piezo-micro-electro-mechanical systems for low frequency energy harvesting. Swapnil D. Shamkuwar and Kunal N. Dekate (Electronics, G.H.Raisoni College of Eng.,Nagpur, 96 Naik Nagar, Post Parvati Nagar, Nagpur, Maharashtra 440027, India, shamkuwarswapnil@gmail.com)

The exibility associated with piezoelectric materials makes them very attractive for power harvesting. Piezoelectric materials possess a large amount of mechanical energy that can be converted into electrical energy, and they can withstand large strain magnitude. The critical physical dimensions of MEMS devices can vary from well below one micron on the lower end of the dimensional spectrum, all the way to several millimeters. While the functional elements of MEMS are miniaturized structures, sensors, actuators, and microelectronics, the most notable (and perhaps most interesting) elements are the microsensors and microactuators. Microsensors and microactuators are appropriately categorized as "transducers," which are defined as devices that convert energy from one form to another. In the case of microsensors, the device typically converts a measured mechanical signal into an electrical signal. Mechanical compression or tension on a poled piezoelectric ceramic element changes the dipole moment, creating a voltage. Compression along the direction of polarization, or tension perpendicular to the direction of polarization, generates voltage of the same polarity as the polling voltage. Hence, by changing device physics, we may get a sensor with higher output with low power consumption and reduced size of the device. The proposed piezoelectric sensor will be designed in COMSOL software and respective characteristics analysis will be observed.

2:30

2pEA5. Ultrasound communication for body senor network. Meina Li, Canghee Hyoung, Junghwan Hwang, Sungweon Kang, and Kyunghwan Park (Human Interface SoC, Electronics and Telecommunications Res. Inst., 218 Gajeongno, Yuseong-gu, Daejeon 305700, South Korea, limeinajl85@etri.re.kr)

Radio frequency (RF) waves have been the dominant part in the field of wireless communication. However, RF waves has the limitation when transmit through the human due to the most of body composition is water. Therefore, the ultrasound that has good propagation in the water has been proposed as the new communication wave for the body sensor network (BSN). Two ultrasonic sensors are used in the communication system. One was placed under skin as the transmitter (Tx) and the other one as the receiver (Rx) was placed right toward the Tx on the top of skin. The Tx can detect and monitor the physiological signal inside of body then transmit the information to the Rx. After the Rx received the signal, it can transmit the information to the doctor also can release the drug to the patient. The modulation of ultrasonic wave has been experimented by three common modulation ASK, PSK, FSK for digital communication. The results showed the ultrasound wave is a possible communication method for body sensor network.

2:45

2pEA6. An experimental and computational study of beam-steering of parametric array. Kyunghun Been, Yub Je, and Wonkyu Moon (Mech. Eng., Pohang Univ. of Sci. and Technol., POSTECH, San 31, Hyojadong, Namgu, Pohang 790-784, South Korea, khbeen@postech.ac.kr)

A parametric array is a nonlinear conversion process that can generate a highly directional sound beam with a small aperture. It is expected that electrical beam steering of directional sound beams generated by the parametric array may be useful in many applications such as ultrasonic raging sensors or directional loudspeakers in air. One of the major issues of beam steering of the parametric array is to precisely predict the steered difference frequency wave field in the medium. In this study, beam steering of the parametric array is computed by using a time-domain numerical code that solves the Khokhlov-Zabolotskaya-Kuznetsov equation. Since it is impossible to compute the exact difference wave field due to a complex primary source distribution in the medium, a simplified numerical model is proposed. The computed result is compared with the experimental result. For experimental study, 16-channel piezoelectric micromachined ultrasonic transducer array, which consists of two resonant type unit drivers to generate bi-frequency primary waves (f1 = 100 kHz and f2 = 140 kHz), was designed, fabricated, and tested. The beam patterns of the primary and difference frequency waves were measured and compared with the computed result while applying complex weighting to each channel. [Work supported by ADD (UD130007DD).]

3:00

2pEA7. A micro-machined hydrophone based on piezoelectric-gate-of-field-effect-transistor for low frequency sounds detection. Min Sung, Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), PIRO 416, Hyoja, Namgu, Pohang City, Kyungbuk, 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 miniaturization of conventional piezoelectric hydrophone is known to have limits in low frequencies due to high sensor impedance of micro-sized piezoelectric body. In this study, a new transduction mechanism is devised and named as piezoelectric gate of field effect transistor (PiGoFET) so that its application could solve the sensitivity limitation of a miniaturized hydrophone with a tiny piezoelectric body less than 1 mm. The PiGoFET transduction can be realized by combination of a field effect transistor and a small piezoelectric body on its gate. We connect a micromachined membrane 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 strained piezoelectric body modulates the channel current of FET at any frequency less than high limit of transistor; thus, the sound pressure may be transferred to the source-drain currents even at very low frequencies irrespective of the size of piezoelectric body. Under the described concept, a small hydrophone was fabricated by micromachining and calibrated using the comparison method in low frequencies to investigate its performance as a low frequency sensitive hydrophone. [Research funded by MRCnD.]

3:15

2pEA8. A piezoelectric micro-cantilever acoustic vector sensor. Wonkyu Moon, Sungkwan Yang (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 405, POSTECH, San31, Hyoja-dong, Namgu, Pohang City, Kyungbuk 790784, South Korea, wkmoon@postech.ac.kr), Joh Cheeyoung, and Kyungsub Kim (Underwater Sensor Lab., Agency for Defense Development, Changwon, South Korea)

An acoustic vector sensor measures the direction of wave propagation as well as the acoustic pressure. We investigate feasibility of using a piezoelectric micro-cantilever (PEMC) as an acoustic vector sensor in water at the frequencies range below 500 Hz. In order to measure the propagation direction, we try to devise the properties of a PEMC so that its deflection is proportional to the particle velocity due to acoustic waves. We found that the desired property can be obtained with PEMC if it is designed to be flexible enough. Under the assumption that the PEMC affects little on the wave propagation, we have developed a simple lumped parameter model to predict the relationship between the acoustic pressure of a progressive wave and the deflection of PEMC. The developed model shows that the deflection of PEMC is dependent on the magnitude and direction of the incoming progressive wave. In addition, the frequency of the wave is also found to affect the responses of PEMC. Based on the developed simple lumped parameter model, a PEMC acoustic vector sensor was designed with PEMC of 400 μm long 200 μm wide 5.25 μm thick for operating around 200 Hz. The designed PEMC acoustic vector sensor was fabricated by micro-machining and packaged inside cater-oil-filled rubber housing so that it can be tested in the water. The expected dependence of the fabricated PEMC on the direction and acoustic pressure can be observed in the experiments at the target frequency range.

Session 2pMU

Musical Acoustics: Acoustics of the Organ

Uwe J. Hansen, Chair
Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374

Chair's Introduction—1:00

Invited Papers

1:05

2pMU1. Sound choices: Designing a pipe organ to suit its acoustical environment. Matthew M. Bellocchio (Andover Organ Co., P. O. Box 36, Methuen, MA 01832, mbellocchio@andoverorgan.com)

A key to the effectiveness of any pipe organ is to design it to be acoustically compatible with its environment. Churches and concert halls all vary in size, plan, seating arrangement, resonance, reflectivity, and reverberation. A successful instrument should have a balanced sound in the room throughout its frequency range, be capable of both warmth and brilliance, and have a variety of tonal resources suitable for both solo performance and accompaniment. The choices an organbuilder makes regarding placement, physical size and layout, tonal specification (choice of stops), pipe materials and dimensions, wind pressures, and voicing style, all contribute to an instrument's tonal signature and acoustical effectiveness. These decisions are usually based on historical precedents, traditional knowledge, personal taste, and experience. The four-manual Casavant organ, Opus 3145, in the Cathedral of Saints Peter and Paul in Providence, Rhode Island, illustrates Lawrence Phelps' tonal design for a heroically sized mechanical action instrument intended for a large, acoustically live space. This organ, built in 1972 at the height of the Baroque Revival period in American organbuilding, was intended to be a legacy instrument for the designer, the builder, and the client.

1:25

2pMU2. Room acoustics and pipe organs: Putting a STOP to common misunderstandings. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 Lhirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

Misunderstandings about room acoustics exist among architects, building committees, musicians, organ consultants, and organ builders. These range from opinions with no acoustic validity to simple confusion of acoustic principles by those not versed in our science. Common and repeated mistakes this author has encountered while serving as an acoustic consultant to numerous worship houses are discussed along with a rational basis to correct such unfounded beliefs.

1:45

2pMU3. A survey of pipe organ reed research emphasizing recent developments. George Plitnik (Phys., Frostburg State Univ., 120 Compton Hall, Frosrburg, MD 21532, gplitnik@frostburg.edu)

As experimental and theoretical acoustics advanced during the 19th century several prominent researchers attempted, typically unsuccessfully, to understand the physics of pipe organ reeds. Centuries of organ building experience had advanced the art of reed voicing to a consummate skill passed down from masters to apprentices, with no understanding of, and no desire to learn, the underlying physical principles. This presentation will give a cursory survey of two centuries of organ reed investigations and then highlight the research during the past two decades which has unlocked some of the conundrums of the past to render these complicated instruments amenable to scientific understanding. The following reed parameters and their influence tone will be examined: reed thickness, reed length, imposed air pressure, reed curvature, shallot type, shallot filling, and the type of metal used for the reed as well as the provenance and method of its manufacture.

2:05

2pMU4. The influence of the shallot shape on the sound of Trompete reed pipes. Judit Angster, Kai Dolde (Acoust., Fraunhofer IBP, Nobel Str. 12., Stuttgart, 70569, Germany, Judit.Angster@ibp.fraunhofer.de), Rucz Peter (3Budapest Univ. of Technol. , Budapest, Hungary), and Andras Miklos (Steinbeis Transfer Ctr. of Appl. Acoust., Stuttgart, Germany)

The focus of investigations was the influence of certain shallot parameters on the sound of reed pipes. Since the dimensioning of organ pipes is mainly based on the experience of organ builders at present, the aim is to provide scientific data for organ builders explaining the influence of various shallot parameters on the sound. Due to this knowledge, it would be possible to realize the desired ideas of sound of organ pipes by targeted adjustment of these parameters during the dimensioning of the pipes. The investigated Trompete shallots differ in the termination angle of the shallot. To analyze the parameter of the shallot termination, angle measurements were carried out at a simple shallot model, which provide findings on the dependence of the reflection properties on various termination angles. The results show that the findings due to the model comply with the analyses of the sound spectra of the Trompete shallot.

2:25–2:40 Break

2:40

2pMU5. Effect of generators and resonators on musical timbre in coupled systems. Jonas Braasch (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th, Troy, NY 12180, braasj@rpi.edu)

The majority of Organ pipes are typical tone-generator/resonator systems, where the tone is either produced by an air jet or reed. My previous work on free reeds in pipe organs has raised my interest in how different sound generators affect timbre in terms of spectral balance and attack phase. Most recently, I have started to explore different tone generators with the soprano saxophone using custom adapters, including a brass mouthpiece from a cornetto, a bassoon reed and a free reed from the Chinese Bawu. Since the resonator remains the same, the role of the sound generator can be easily determined using a fast Fourier transformation and onset analysis of partial tones. With the brass mouthpiece, the soprano saxophone becomes an outward striking mechanism instead of a reed-based, inward striking mechanism. As a consequence, the instrument vibrates above the natural frequency of the conical resonator, and the B-flat instrument becomes a B instrument. It is important not to underestimate the range of the embouchure. In the case of the cornetto mouthpiece, it took some practice to attain the higher harmonics. Naturally, the spectrum with the cornetto mouthpiece is very similar to that of a regular soprano saxophone. [Work supported by NSF #1002851.]

3:00

2pMU6. Sound production in the reed organ and harmonium. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

The free reed instruments of European origin, which originated around 1800 and were developed over the next 50–75 years, include the reed organ and the harmonium. These keyboard instruments enjoyed a period of great popularity beginning around 1840 and lasting until the early 20th century. They were widely used in homes and churches, and appeared in a variety of instrumental ensembles, including the salon orchestra. They are close relatives of the harmonica and the accordion-concertina family, but unlike these instruments they are not in wide use today. This paper discusses the fundamental mechanisms of sound production in the instruments as well as means used to alter the tone quality, which include the design of the wind system, effects of the chambers in which the reeds are mounted, and details of the reed tongue design. The presentation will include audio examples of the reed organ and the harmonium employed in various musical contexts.

3:20

2pMU7. The Musical Instrument Digital Interface (MIDI): The digital organ for organists and non-organists. Paul Wheeler (Elec. & Comput. Eng., Utah State Univ., 1595 N. 1600 E., Logan, UT 84341, paul.wheeler@usu.edu)

The Musical Instrument Digital Interface (MIDI) is a technical protocol (standardized in 1983) to connect a wide variety of electronic musical instruments. Digital organs, due to increased sound quality and lower pricing, have become a popular alternative to pipe organs. They also provide an advantage of controllability through MIDI. Using MIDI, digital organs can be connected to a variety of sound modules, greatly increasing the number of stops available for organists. For the non-organist composer, organ music can be written in notation software (such as Finale) and played on an actual digital organ. The challenge in writing music for playback through MIDI is to incorporate organ techniques (such as shortening of repeated notes, legato playing by finger crossing or finger substitution, and finger glissandos) so that the result does not sound like a pianist (or a computer) playing the organ. A MIDI capable digital organ can be advantageous for organists and non-organists alike.

3:40

2pMU8. Bach with sampled sounds. Uwe J. Hansen (Chemistry& Phys., Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu) and Norman C. Pickering (Pickering Res., East Hampton, NY)

A number of years ago Norman used a digital recorder to record sound samples from a Casavant organ with two manuals plus pedals, with a total of 13 stops in 16 ranks of 882 pipes. He did this by recording all pipes for every other key. He included the initial transient for each sample. Additional organ stops were synthesized from multiple oscillators of various waveforms, based on FFT analysis of actual organ stops. All these sounds were mapped onto the appropriate keys of two synthesizer keyboards. Using these sampled sounds, Norman recorded nearly the entire Bach Orgelbuechlein and several other major works including six Schuebler Chorales, a concerto, BWV 594, and a Prelude and Fugue, BWV 552. We will listen to a brief excerpt from “Wachet auf” to hear the effect of the initial transients. A portion of the Prelude of BWV 552 will give an impression of the full organ.

Session 2pNS

Noise and Architectural Acoustics: Acoustics During Construction

Norman H. Philipp, Cochair
Geiler and Associates, LLC, 1840 E. 153rd. Cir., Olathe, KS 66062

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

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. The integration of acoustics in construction management education. Norman H. Philipp (School of Construction, Pittsburg State Univ., 1840 E. 153rd Circle, Olathe, KS 66062, nphilipp@geileracoustics.com)

An overview of the importance and need for acoustics and proper noise control mitigation in assemblies and during construction being integrated into construction management education. Items discussed include the impact of noise on a construction site, workers, and neighboring areas; importance of considering acoustical considerations in project scheduling; and methodologies for enabling student understanding of acoustics in construction.

1:25

2pNS2. Mitigation of construction noise at operating hospital and university facilities. Kerrie G. Standlee (Daly-Standlee & Assoc., Inc., 4900 SW Griffith Dr., Ste. 205, Beaverton, OR 97005, kstandlee@acoustechgroup.com)

Hospitals and universities often have occasions where new or remodeling-related construction must occur during times when existing facilities are in use. Noise caused by construction at an active facility often requires that steps be taken to minimize the impacts associated with the construction. This paper discusses the findings made during the course of several projects undertaken by Daly-Standlee & Associates, Inc., to determine mitigation measures that could be used to minimize noise impacts from construction at operating hospital and university facilities.

1:45

2pNS3. Noise control for an impact pile driver in an urban environment. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Soil conditions at a construction site for a new seven-story office building in downtown Seattle, Washington, dictated the use of an impact pile driver to set 50 to 60-ft long concrete piles into the ground, many less than 200 ft from existing offices. Exterior and interior noise measurements revealed that the pile driving activity exceeded the city's noise ordinance. A portable acoustic shield was designed, constructed, and implemented to reduce noise levels below the legal limits. Measured sound levels before and after will be presented along with details and photos of the shield in action.

2:05

2pNS4. Construction issues affecting the intended acoustical environment of a new court tower. Robert M. Brenneman (Acoust., McKay Conant Hoover, Inc., 7435 E Stetson Dr., Ste. A, Scottsdale, AZ 85251, rbrenneman@mchinc.com)

The proficiency to readily identify and resolve construction issues in the field is vital to realizing intended acoustical design goals. In court facilities, inattention to such construction details can result in problematic reductions in speech privacy for confidential deliberations, negotiations, and attorney discussions, structure-borne noise intrusion from holding areas to noise-sensitive spaces, and diminished speech intelligibility in courtroom proceedings. This paper considers examples of acoustics and noise control issues arising during the construction of a new court tower, discussion of their potential effects on the acoustical environment, and methods of addressing these field conditions.

2:25

2pNS5. Preparation, execution, and documentation of construction site visits. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 Lhirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

Construction site visits, referred to as "work observation" by the American Institute of Architects, are a critical factor in project delivery. The unique nature of acoustic detailing requires visual and sometimes field testing to verify correct installation of products and construction assemblies. Preparation for site visits begins during the Construction Documents design phase. A list of site inspections, keyed to project milestones, should be issued to the Contractor at the start of the Construction Administration phase. Site visit objectives should be planned and the Contractor notified prior to arriving on site. When on site it is necessary to conduct inspections in a methodical manner, document potential deviations from design intent, and debrief the Contractor. Checklists can improve efficiency and avoid overlooking key objectives. Work observation reports become part of the project record and are important for tracking corrective measures by the Contractor. Acoustic measurements may be part of inspections to check quality of work or determine compliance with Specifications. Well-planned and executed site visits contribute to project success and boost our profession's esteem with the building construction industry.

2:45

2pNS6. A potpourri of acoustical issues arising during construction administration. Joseph F. Bridger and Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd.,Ste. 101, Raleigh, NC 27607, joe@sacnc.com)

This is a collection of our firm's most noteworthy experiences during construction administration. First, they illustrate the many ways what is built can go astray from what was recommended. In other cases, we do not get involved with the particular area of scope until construction administration. Sometimes in such cases, design professionals attempt to do acoustics themselves but lack the knowledge to do so correctly. At other times, design professionals are not even aware that an acoustics issue should be addressed, until it is noticed during construction. Of course, there is the contractor whom did not understand or simply did not do what the drawings and specifications called for. Lastly, there can simply be things that cannot/should not be built the way they were designed (often the contractor notices these things first) and the inevitable "now what do we do?" Of course anything that comes up during construction is on a short and critical time schedule, and fraught with the challenge of how to get things fixed quickly and with least added cost. It is our hope that these case studies will help building professionals and acoustical consultants on future projects.

3:05–3:20 Break

3:20

2pNS7. On the importance of experienced acoustical inspectors during and following the construction process and installation of noise mitigation measures. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

During the installation of recommended noise mitigation measures that appear on architectural drawings or that are presented in project acoustical reports, demolition, renovation, office building, institutional facilities, apartment buildings, and condominium construction trades may or may not omit and/or improperly install the recommended noise control systems. This paper presents several examples of such improper practices that, had they not been discovered, would have resulted in significant reductions in the planned noise control performance. These cases result in undesired living and/or working conditions. The selected examples of such construction problems are discussed and successful remedies are presented, some of which avoided threatened litigation.

3:40

2pNS8. Issues and opportunities during construction of a new courthouse building. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

A new courthouse building was recently completed, housing 31 courtrooms and the full complement of ancillary spaces, and using a delivery system called performance-based infrastructure (PBI) that enhanced the need for high durability and long-term functionality, as well as a fast-paced schedule. An interesting issue arose concerning unexpectedly loud levels of toilet flush noise in a public restroom adjacent to a judge's chamber, whose resolution resulted in 20 dBA reduction at the source without need for reconstructed partitions in over a dozen similar adjacencies. An interesting opportunity arose when on opening day it was determined that a cueing room was noisy, and one of the architects emailed audio files of balloon pops recorded on his cell phone, in a manner that simulated measurements that he had witnessed several times before, and which confirmed our calculations sufficiently to engender agreement on treatment without need for additional site visits. These and perhaps other issues and opportunities will be discussed.

4:00

2pNS9. Direct and flanking airborne sound transmission for hotel guestrooms; results and comments regarding three field tests. Jim X. Borzym (Borzym Acoustics LLC, 2221 Columbine Ave., Boulder, CO 80302, acoustics@columbine.netr)

A "four-and-a-half star" hotel was constructed within an existing historic structure. Among the design/build team there were differences of opinion of what was needed for peace and privacy for hotel guests. Hotel guidelines suggested a moderately high airborne sound transmission class rating for the primary wall partition. Three field tests were conducted during an early phase of construction. A strong flanking transmission path in a corridor wall assembly was discovered during the first test. An error in preparation for this test caused an important opportunity for valuable field test data development to be lost regarding a flanking path via a common ceiling, which points to the kinds of difficulties encountered in conducting field tests. The second test showed improvement in results due to rectification of the corridor wall flanking path. The third test gave information about flanking transmission via the common floor. Comments will be made regarding field testing and flanking sound transmission.

4:20

2pNS10. Construction monitoring—A lack of standardization. Tyler Rynberg (VACC, Inc., 490 Post St, Ste. 1427, San Francisco, CA 94102, tyler@va-consult.com)

Construction noise and vibration monitoring is being required more and more frequently, especially for large infrastructure projects. Work on these civil projects often involves tunneling, piling, large-scale earthworks, and even blasting—all in close proximity to commercial or residential population centers. Aside from the human-related annoyance effects, there are effects on the environment (endangered species) and the possibility of damage to structures. Because construction monitoring for noise and vibration is a relatively young field, there is little guidance for agencies to consult; the result is a multitude of differing monitoring requirements on these projects. From project to project, the monitoring requirements differ in subtle ways that do not necessarily correspond to the true needs of different projects. This presents serious problems in terms of developing consistent protocols (allowing apples to apples comparisons) and also in instrumentation, as most instrument vendors are too inflexible to meet the range of requirements or are too expensive for the flexibility they do offer (or both). A discussion to develop more consistent standards in construction monitoring is needed.

Contributed Papers

4:40

2pNS11. Method to improve warning sound detection for hearing protectors. Eric Bernstein, Anthony J. Brammer, and Gongqiang Yu (Univ. of Connecticut Health Ctr., 263 Farmington Ave., Farmington, CT 06030, eric.r.bernstein@gmail.com)

Hearing protection devices (HPDs) provide both desirable attenuation of environmental noise and undesirable attenuation of auditory warning alarms, such as vehicle backup alarms. Both warning alarm detection and localization performance are affected. A digital cross-correlation based method is described to identify a pre-selected warning alarm to bypass the attenuation of the HPD while maintaining attenuation of environmental noise outside the bandwidth of the alarm. This method can be integrated into existing digital HPD designs. Computer simulation of the algorithm demonstrates that an alarm signal can be detected at signal-to-environmental noise ratios as low as 30 dB for the military and industrial noise sources investigated. Implementation of the method using a modified commercial HPD demonstrates a 7 dB improvement in warning alarm detection threshold compared with an unmodified HPD. Alternative methods for presenting the alarm signal to the user will be discussed as well as modifications to expand the method to accommodate multiple alarm signals.

4:55

2pNS12. Reduction of construction noise while building a recording studio. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

While a new recording studio was being built within a large Audio and Video complex, construction noise had to be adequately controlled, in order to allow for the already existent audio recording studios and TV sets to continue with their normal activities along the building process. The noise was reduced by making a double wall system in the façade overlooking the construction area. Use was made of the already existing external wall, adjacent to the construction site, as the first wall of the system, and a second one was added in the exterior side of that wall in order to complete the double wall system, which contributed with a few dB's in the low frequency range, while getting a considerable noise level reduction in the high frequencies range to the overall noise reduction of the original wall. Producers and performers who were actually working in the existing studios were satisfied by the results obtained once the adaptation was complete. Details of the wall and its expected acoustical behavior are presented and discussed.

Session 2pPA

Physical Acoustics and Education in Acoustics: Demonstrations in Acoustics

Murray S. Korman, Chair
Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402

Chair's Introduction—12:55

Invited Papers

1:00

2pPA1. Apparatus for demonstrating evanescent waves in acoustic waveguides. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu) and Daniel O. Ludwigsen (Dept. of Phys., Kettering Univ., Flint, MI)

A physical demonstration apparatus, inspired by [K. Meykens, *et. al*., Am. J. Phys., **67**(5), 400–406 (1999)] is used to demonstrate evanescent sound waves in a rectangular waveguide. The apparatus consists of a rectangular waveguide, approximately one meter in length, with one optically transparent wall. The waveguide is driven at one end by a pair of loudspeakers whose polarity may be switched. The other end of the waveguide is open. The sound field inside the waveguide may be interrogated using a microphone and an oscilloscope. The apparatus will be used to demonstrate several features of acoustic wave guides, including the propagation of plane waves, the cutoff frequency for non-plane waves, the propagation of non-plane waves above the cutoff frequency, and the exponential decay of evanescent non-plane waves below the cutoff frequency. The educational and teaching applications of this apparatus will be discussed.

1:20

2pPA2. Fun with foggers. R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu)

Many physical acoustics phenomena can be demonstrated with a so-called "pond fogger." The easy availability and affordable price of these acoustic transducer systems allows one a lot of freedom in the demonstrations one can do. Three phenomena, acoustic streaming, standing waves, and Faraday waves, will be demonstrated in this talk. The audience will be invited to propose and investigate others during the talk.

1:40

2pPA3. Use of the hammered dulcimer to demonstrate physical acoustics principles. Cameron T. Vongsawad, Kent L. Gee, Tracianne B. Neilsen, and Benjamin Y. Christiansen (Dept. of Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604, cvongsawad@byu.net)

In 1636 AD, Marin Mersenne described the law of vibrating strings that relates frequency to length, tension, and density in *L'Harmonie Universelle*. One of the instruments described by Mersenne is the *psalterion* or hammered dulcimer. The dulcimer is a versatile struck-string instrument, based on the circle of fifths, in which the sound generation and radiation are linked to many physical acoustics principles. Some of these principles are basic: how string properties affect fundamental frequency; how the soundboard and body cavity give rise to different resonances; how hammer construction and excitation affect the spectrum; etc. Because of the ease of demonstrating acoustical principles with the hammered dulcimer, a 9/8 backpack-size dulcimer has been included as part of the new ASA outreach workshops held in conjunction with semiannual meetings. A description of these efforts and a short demonstration of the hammered dulcimer will be given. More advanced concepts to be discussed include high-speed video results of the hammer-string interaction as well as near-field acoustical holography analyses on a 16/15 Songbird® dulcimer. The holography shows how the sound holes, traditionally thought to be merely decorative, significantly influence the radiated sound.

2:00

2pPA4. Demonstration of an extremely directional acoustic source. Preston S. Wilson, Craig N. Dolder, and Mark F. Hamilton (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, pswilson@mail.utexas.edu)

Highly directional light sources such as flashlights and lasers are well known to most people. In contrast, highly directional acoustic sources, or in other words, sources of sound that are audible in only a very narrow region of space, are far less common. Many people have never experienced such a source, and the phenomenon is not found in nature. A highly directional source of sound known as a parametric array is used underwater for sonar applications, but the frequency (pitch) of the sound is often above the human hearing range. Similarly, highly directional, focused sound sources are regularly used in medical applications, but again, the frequency is too high to be heard. The narrowness of the acoustic beam cannot be experienced by human listeners. Recently, parametric array technology has been commercialized for use in air at frequencies in the human auditory range. These devices produce extremely narrow (on the order of 2 degrees) beams of audible sound. When pointed directly at one listener, the sound is virtually inaudible to another listener only a few feet away. Such a device will be demonstrated and the basic physics behind its operation will be explained.

2:20

2pPA5. Acoustics concepts you can demonstrate with a coffee mug. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

The simple coffee mug can be used to illustrate many important concepts in acoustics. Adding water to the mug shows the effect of mass on the frequency of oscillation. Striking on the handle of the mug compared to striking the side of the mug shows the vibrational mode degeneracy lifting. Adding instant coffee or hot chocolate mix to a mug full of water heated in a microwave oven shows the effect of bubble density on the speed of sound. Each of these concepts are easily demonstrated and can also be used as starting points for student exploration in the laboratory.

2:40–3:00 Break

3:00

2pPA6. Nonlinear experiments of tuning curve resonances for the vibrational modes of a weakly stretched circular membrane. Benjamin W. Lloyd and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

Nonlinear oscillations of a pre-stretched 11.5 cm diameter latex circular membrane clamped rigidly around the boundary were investigated. The relaxed membrane thickness was 0.4 mm. An 8 cm diameter loud speaker drove the membrane from below using a swept sinusoidal tone. Low drive tuning curve sweeps demonstrated resonant frequencies corresponded to the first three radially symmetric linear drum modes (56, 147.6, and 240.5 Hz) sweeping from 50 to 300 Hz. Regression yielded a transverse wave speed of 10.6 m/s. Incrementally increasing the speaker amplitude (after each sweep) showed the lowest resonant frequency at first decreases, then increases with increasing acoustic drive pressure. Higher modes exhibited frequency increases with drive level. Tuning curve responses were measured using a 6 mm diameter microphone 5 mm above the center of the membrane. Nonlinear tuning curves hysteresis effects occur for higher drive levels sweeping between 50 and 70 Hz. Here, the amplitude sharply falls from a high to a low microphone response at f_a = 59 Hz. With the same drive level, a sweep tone starting from 70 Hz to 50 Hz exhibits at f_a = 57.5 Hz, a jump from a lower to a higher microphone response. [Goncalves *et al*., J. Vib. Sound **327**, 231 (2009).]

3:20

2pPA7. Non-electric sensing, amplification, and presentation of sound using fluidic laminar flow technology. Michael V. Scanlon (US Army Res. Lab., RDRL-SES-P, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

This briefing will describe and demonstrate fluidic-based sound reception, amplification, and pneumatic headphone presentation to the listener. Fluidics is non-electronic and uses laminar proportional amplifiers and fluid-flow technology to demonstrate a sensitive "microphone" due to its massless diaphragm (laminar jet of low-velocity air) and perfect impedance matching to the propagating medium (also air). Many years ago the U.S. Army developed a system called the Individual Soldier Operated Personal Acoustic Detection System (ISOPADS) that utilized this fluidic technology to enhance and extend the Soldier's listening range. This demonstration system will let people point a small parabolic dish and listen around the room using a pneumatic headset. It makes for a fun demo, especially when participants understand that they are listening to a non-electronic system that uses only air.

3:40

2pPA8. Generation of photoacoustic transients from optically induced thermal gradients. Clifford Frez, Binbin Wu, and Gerald Diebold (Dept. of Chemistry, Brown Univ., Box H, Providence, RI 02912, Gerald_Diebold@Brown.edu)

Irradiation of an absorbing surface in contact with a transparent fluid with a pulsed laser can result in the generation of extremely large thermal gradients. For example, when a laser with a pulse width of 10 ns and a fluence of 1 J/cm^2 irradiates a region with an absorption of 1 cm^1 having the thermal properties of liquid water, a thermal gradient on the order of 105 K/m at the interface is produced. Here, it is shown that the effect of such thermal gradients on photoacoustic waves from an infinite half space and from a uniformly irradiated sphere is the production of fast transients on the leading edges of the waves. The character of the transients is determined from an additional source term in the wave equation for pressure that obtains when heat conduction is taken into account. Experiments are reported showing the predicted transients on photoacoustic waves from absorbing layers in contact with transparent fluids irradiated with 10 ns laser pulses.

4:00

2pPA9. Investigation of capillary wave formation on water jets with internally propagating ultrasound. Nikhil M. Banda (ISVR, Faculty of Eng. and Environment, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, mnb1g10@soton.ac.uk), Offin Douglas, Birkin R. Peter (Dept. of Chemistry, Univ. of Southampton, Southampton, United Kingdom), and Leighton G. Timothy (ISVR, Faculty of Eng. and Environment, Univ. of Southampton, Southampton, United Kingdom)

The formation and control of capillary waves on jets are of importance in applications such as ink-jet printing, atomization of fuel, etc. External control of jet breakup processes is generally based on the ultrasonic atomization principle (used on jets of diameter in micrometer order) with transducers placed on nozzle tips. The present work investigates the formation of capillary waves and subsequent jet breakup on 10 mm and 15 mm diameter water jets with ultrasound propagating at either 121 kHz or 135 kHz. Experimental

observations of the jet breakup process with a high speed camera are reported. The input signal to the transducer was controlled to investigate the formation and growth of capillary waves, leading to the breakup of the jet. It was observed that once the waves are formed on jet surface, they grow in size leading to a necking zone. Once necking zone is formed, the capillary waves then just propagate along the jet (with the flow) with no further growth in their amplitude. Spraying of the jet was also observed at the same time. The measurement of capillary wavelength and jet breakup length are measured and presented in an attempt to understand the nature of the breakup process.

4:20

2pPA10. Comparison of experimental and theoretical mode studies on a square plate. Uwe J. Hansen (Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu)

Standard two-dimensional Chladni patterns on a square plate are demonstrated and compared with the results of Finite Element mode calculations

4:40–5:30 Demonstration Interaction

TUESDAY AFTERNOON, 6 MAY 2014 BALLROOM B, 1:25 P.M. TO 5:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Scientific Catalyst, Collaborator, and Gadfly: Honoring the Contributions of Tino (Constantine) Trahiotis to the Understanding of Binaural Auditory Processing

Leslie R. Bernstein, Cochair
Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, 263 Farmington Ave., Farmington, CT 06030

H. Steven Colburn, Cochair
Hearing Res. Ctr. and Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215

Richard M. Stern, Cochair
Electrical and Computer Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213

Chair's Introduction—1:25

Invited Papers

1:30

2pPP1. Constantine Trahiotis and hearing science: A half-century of contributions and collaborations. Leslie R. Bernstein (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, 263 Farmington Ave., Farmington, CT 06030, Les@neuron.uchc.edu)

Although best known for his contributions to human binaural processing, Constantine (Tino) Trahiotis has, throughout his career, been an avid student of virtually all aspects of hearing. His published work spans nearly 50 years and has been carried out with an impressive cadre of collaborators worldwide (all of whom he thinks of as family). That body of work attests to the diversity of Tino's interests and encompasses behavioral and lesion studies in animals, human psychoacoustics, mathematical modeling, methodology, signal-processing, and instrumentation. Tino's recall of the literature is legendary. As compared to the use of traditional methods to locate publications concerning a particular topic, many of us, especially the string of scientists he has mentored, know that it is often more efficient to just pick up the phone and ask Tino. This presentation traces the history of Tino's career, including his days as a graduate student with Don Elliott at Wayne State University, his time at Indiana University, and as a Professor at the University of Illinois and at the University of Connecticut Health Center.

1:50

2pPP2. Tino Trahiotis's impact on general binaural research progress. H. Steven Colburn (Hearing Res. Ctr. and Biomedical Eng. Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, colburn@bu.edu)

The evolution of thinking and understanding in binaural hearing research over the past six or seven decades will be reviewed. Both physiological and psychophysical developments, as well as their interactions and applications, will be described. The focus of this review will be the impactful role that Tino Trahiotis played across this wide spectrum of research areas. Tino's large impact came through his personal engagement with the ideas and the people involved in this research. His enthusiasm for the topic is contagious and provides fuel for digging deeper and making connections across diverse research topics. The progress has been exciting and the whole process has been great fun... consistent with the Trahiotis style. [Work supported by NIH/NIDCD DC00100.]

2:10

2pPP3. Time and intensity with Tino. Richard M. Stern (Elec. and Comput. Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rms@cs.cmu.edu)

Over a period of decades Constantine (Tino) Trahiotis has been a contemplative and creative scholar, an excellent teacher, a helpful collaborator, as well as a loyal friend. Tino, along with Bob Bilger and Erv Hafter, provided this author's introduction to the broader science of hearing beyond the particular perspective of my graduate training. At the same time, my attempts to interpret models of binaural interaction with Tino led to the review chapters on binaural modeling that the two of us jointly authored in the 1990s. Tino's demonstration of the unexpected dependence on bandwidth of the laterality of bands of noise presented with interaural temporal differences of large magnitude motivated me to think about the tradeoff between "straightness" and "centrality," which appears to be helpful in considering the lateralization of many binaural stimuli. Finally, Tino, along with Les Bernstein, insisted on a local implementation of my models in their laboratory which became one source (along with significant others) for the binaural modeling toolbox developed by Michael Akeroyd that is presently in wide circulation. This talk will review these facets of my life with Tino, and comment on my current perspectives on related issues. [Work supported by DARPA and Cisco Research.]

2:30

2pPP4. The temporal acuity for processing interaural cues. Steven van de Par and Darrin Reed (Acoust., Univ. of Oldenburg, Carl-von-Ossietzky-strasse 9-11, Oldenburg 26129, Germany, steven.van.de.par@uni-oldenburg.de)

It was during a three month visit of the first author to his Lab in 1998 that Tino pointed out an elegant study on detecting changes in Interaural Cross Correlation of stimuli that were roved in baseline Interaural Time Delay and Interaural Level Difference (Bernstein and Trahiotis, J. Acoust. Soc. Am. **102**, 1113). This study showed that, even though ICC can only change due to the presence of dynamically varying ITDs and ILDs, the ICC can be perceived as a separate cue in certain conditions. This articulate notion of three perceptually independent binaural cues, ITD and ILD, relating to perceived laterality, and ICC, related to perceived width, is applied in low bit-rate audio coding where these cues are used to encode the spatial image of stereo sound recordings. The entanglement of these cues on a signal basis raises the question of the time scale at which these three cues are perceptually evaluated. Some recent findings will be discussed that show that the ITD and ICC can be evaluated with a high temporal acuity of less than 10 (ms) in stimuli where these cues alternate periodically, and that these patterns of alternating binaural cues give rise to a modulation percept.

2:50

2pPP5. "The data are the data": Tino and the importance of empirical observation. Michael Akeroyd (MRC/CSO Inst. of Hearing Res. - Scottish Section, New Lister Bldg., Glasgow Royal Infirmary, Glasgow, Strathclyde G31 2ER, United Kingdom, maa@ihr.gla.ac.uk)

I was a postdoctoral scientist in Tino and Les' binaural psychophysics lab from 1999 to 2001. One of my abiding memories from my time then was the motto "the data are the data": in almost every discussion we had on science, Tino would illustrate a point or an idea by reference to a graph of data. Often the graphs were obscure and unjustly ignored by later scientists—but Tino, with his encyclopedic memory for data, would know just where to look for it in a issue of JASA or an mimeographed manuscript. It is a lesson I have always remembered: that science is fundamentally based on accurate measurements of phenomena, and, though the explanations and interpretations may change, the data are always the data, and they are not undone or falsified by the passage of time. This talk will illustrate the argument with data—after all, what else?—from Tino's, mine, and others' papers. [Work supported by the Medical Research Council and the Chief Scientist Office, Scotland.]

3:10–3:25 Break

3:25

2pPP6. Different ears. Marcel van der Heijden (Neurosci., Erasmus MC, P.O.Box 2040, Rotterdam 3000 CA, Netherlands, m.vanderheyden@erasmusmc.nl)

Most models of binaural processing assume identical inputs from the two ears to the binaural processing stage. From the high accuracy of binaural processing one may expect slight deviations from perfect symmetry of its inputs to affect performance. Such deviations may be systematic such as the tuning differences postulated by stereausis models. Alternatively, asymmetries may simply result from imperfections in cochlear frequency tuning or from "sloppy wiring" projecting to the binaural cells. I will analyze how different types of interaural asymmetry affect predicted performance in psychoacoustic tasks and discuss physiological evidence for imperfect symmetry of the monaural inputs.

3:45

2pPP7. The influence of pause, attack, and decay duration of the ongoing envelope on the extent of lateralization produced by interaural time differences of high-frequency stimuli. Mathias Dietz, Martin Klein-Hennig, and Volker Hohmann (Abteilung Medizinische Physik and Cluster of Excellence "Hearing4all," Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, mathias.dietz@uni-oldenburg.de)

Klein-Hennig *et al.* [J. Acoust. Soc. Am. **129**, 3856 (2011)] investigated the influence of the duration of specific modulation cycle segments within the ongoing envelope waveform on the sensitivity to interaural time differences (ITD). The ITD sensitivity, measured in a two alternative forced choice discrimination task, was reported to increase for increasing pause and decreasing attack segment duration. The study also revealed that "on" and decay durations have little to no influence on the threshold ITD. The current study employed a subset of nine envelope shapes from the previous study and measured the extent of lateralization produced by ongoing ITDs with an acoustic pointing task. Lateralization generally increased monotonically with ITD for the measured values of 0.2, 0.6, 1, and 2 ms. An additional condition measured combined lateralization of a 1 ms ITD and an opposing 5 dB interaural level difference. It was observed that the extent of lateralization increases with increasing pause duration or with decreasing attack duration in line with the threshold ITD data. However, the different influence of attack and decay flank on ITD sensitivity translates into significant differences in the extent of lateralization only for a subgroup of subjects.

4:05

2pPP8. Sound source localization: Clicks and click trains. William Yost, Xuan Zhong, and Anbar Najam (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287, william.yost@asu.edu)

Tino Trahiotis and Les Bernstein have provided valuable information about human listeners' ability to process interaural time differences in the envelopes of high-frequency carrier signals. These data have enriched models of binaural processing. In this paper, we explore sound source localization of click (100 microsecond transients) stimuli in the azimuth plane in the free field. We are especially interested in the sound source localization of click trains as they provide stimuli with robust envelope properties. We measured sound source localization accuracy for tones, single clicks, and click trains–unfiltered and filtered in low- and high-frequency regions. The filtering was performed to implicate the role of interaural time and level differences in sound source localization. These data involving clicks will be compared to recent findings from our laboratory involving sound source localization of broadband and filtered noise bursts as compared to sound source localization of umodulated and amplitude modulated tonal carriers. These data suggest that stimulus bandwidth is a major factor determining sound source localization accuracy, and that amplitude modulation plays a small role, at best, in determining sound source localization accuracy. The data involving click trains will help expand this database. [Research supported by the AFOSR.]

4:25

2pPP9. A neural measure of interaural correlation based on variance in spike count. David McAlpine, Simon Jones, and Torsten Marquardt (Ear Inst., Univ. College London, 332 Gray, London WC1X 8EE, United Kingdom, d.mcalpine@ucl.ac.uk)

Interaural Correlation (IAC) is related to variance in Interaural Time Difference (ITD) and Interaural Level Difference (ILD). While normalized IAC can account for behavioral performance in discrimination tasks, so can models directly employing this variance as a cue. Attempts at identifying a neural correlate of IAC discrimination have focused on changes in mean spike count, typically at the peaks of ITD tuning curves. We propose that IAC discrimination relies on variance in spike rates on the slope of neural tuning curves (for ITD). We developed a physiologically based hemispheric-balance model of IAC, where fluctuations in the ratio of activity between left- and right-brain serve as the detection cue to a reduction in IAC from unity, a ratio that is stimulus power invariant. Adjusting model parameters, we find that two orders of magnitude less activity is required in the variance-based, compared with mean spike-rate based, model, in order to achieve the same performance. This adjustment also revealed a necessary neural time integration of 10 ms, which is comparable with physiological estimates. The model was tested by recording neural responses from the midbrain of anesthetized guinea pigs to noise stimuli of various IAC.

Contributed Paper

4:45

2pPP10. Combination of two consecutive monaural "looks" as a spatial hearing cue. Xuan Zhong and William Yost (Speech and Hearing Sci., Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85281, xuan.zhong@asu.edu)

Interaural time, level, and spectral differences are the major cues being used for sound source localization in the horizontal plane. Past studies have shown that human subjects could be trained to use monaural spectral cues to localize the sources of sound in the azimuth plane, but performance is poor. The current study investigates whether the combination of two monaural signals at both ears, one after another in time, could benefit sound source localization accuracy. The purpose is to study the possibility that the human subjects can compare a monaural signal at one ear to the short term memory of a prior monaural signal arriving at the other ear. Subjects were asked to judge the position of a loudspeaker that presented a 250-ms, 40 dBA noise burst with a roving spectral contour in a quarter field. The rms sound source localization error was measured in three conditions: (1) a single monaural signal; (2) two consecutive monaural "looks" at identical signals, with an interval of 3 s and (3) normal binaural hearing. It was found that the "two looks" localization performance was better than that in the case of monaural presentation, but is still inferior to that of the binaural presentation. Similar experiments were carried out over headphones. Lateralization performance was compared to the sound-field localization data. Contributions of level and spectral differences will be discussed. [Research supported by the AFOSR.]

Session 2pSA

Structural Acoustics and Vibration, Physical Acoustics, Engineering Acoustics, and Noise: Acoustic Metamaterials II

Christina J. Naify, Cochair
Acoust., Naval Res. Lab., 4555 Overlook Ave. SW, Bldg. 2, 138G, Washington, DC 20375

Michael R. Haberman, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

1:00

2pSA1. Harnessing geometric and material nonlinearities to design tunable phononic crystals. Katia Bertoldi, Pai Wang, Sicong Shan, and Sahab Babaee (Harvard Univ., 29 Oxford St., Cambridge, MA 02138, bertoldi@seas.harvard.edu)

We investigate numerically and experimentally the effects of geometric and material nonlinearities introduced by deformation on the linear dynamic response of two-dimensional phononic crystals. Our results not only show that deformation can be effectively used to tune the band gaps and the directionality of the propagating waves, but also reveal how geometric and material nonlinearities contribute to the tunable response of phononic crystals. Our study provides a better understanding of the tunable response of phononic crystals and opens avenues for the design of systems with optimized properties and enhanced tunability.

1:20

2pSA2. Damping and nonlinearity in elastic metamaterials: Treatment and effects. Romik Khajehtourian, Michael J. Frazier, Clémence Bacquet, and Mahmoud I. Hussein (Aerosp. Eng. Sci., Univ. of Colorado Boulder, ECAE 194, UCB 429, Boulder, CO 80309, mih@colorado.edu)

Locally resonant acoustic/elastic metamaterials have been the focus of extensive research efforts in recent years due to their attractive dynamical characteristics, such as the possibility of exhibiting subwavelength bandgaps. In this work, we present rigorous formulations for the treatment of damping (e.g., viscous/viscoelastic) and nonlinearity (e.g., geometric/material) in the analysis of elastic wave propagation in elastic metamaterials. In the damping case, we use a generalized form of Bloch's theorem to obtain the dispersion and dissipation factors for freely propagating elastic waves. In the nonlinear case, we combine the standard transfer matrix with an exact formulation we have recently developed for finite-strain elastic waves in a homogeneous medium to obtain the band structure of a 1D elastic metamaterial. Our analysis sheds light on the effects of damping and nonlinearity on the dispersive characteristics in the presence of local resonance.

1:40

2pSA3. Dynamics of geometrically reconfigurable one dimensional and two dimensional magneto-elastic metamaterials. Massimo Ruzzene and Marshall Schaeffer (Georgia Inst. of Technol., 270 Ferst Dr., Atlanta, GA 30332, ruzzene@gatech.edu)

Periodic structures are presented as metamaterials that exhibit multistability due to the nonlinearities of magneto-elastic interactions and structure geometry. The multistability of these structures affords them the ability to adapt their properties though geometric reconfiguration, bringing about changes in stiffness and Poisson's ratio, and introducing anisotropy. These changes in structural properties cause drastic changes in wave propagation, which is of interest for wave control. The dynamic transformation of one-dimensional (1D) and two-dimensional (2D) lattices between stable states are studied through nonlinear numerical simulations. The analysis is conducted using a lumped mass system of magnetic particles. The structures studied include hexagonal, re-entrant, and kagome lattices. Changes in plane wave propagation properties are predicted by applying Bloch theorem to lattice unit cells with linearized interactions. Results from Bloch analysis are then verified through direct numerical simulations. The propagation of plane waves in these lattices before and after topological changes is compared, and large differences are evident.

2:00

2pSA4. Microscale granular metamaterials. Nicholas Boechler (Dept. of Mech. Eng., Univ. of Washington, Mech. Eng. Bldg., Box 352600, Seattle, WA 98195, boechler@uw.edu)

Locally resonant metamaterials and granular media are both known to drastically affect acoustic wave propagation. However, there are thus far few examples of such materials which have microscale elements and are designed to control acoustic waves with megahertz frequencies or greater. In this talk, I will discuss our recent work at the intersection of these two types of materials, in which we explored the interaction of megahertz-gigahertz frequency surface acoustic waves with a self-assembled metamaterial composed of microspheres

adhered to an elastic substrate. I will present our theoretical model and describe our photoacoustic experiments, in which we used transient-grating spectroscopy to measure the acoustic dispersion of the system. Finally, I will also discuss several potential applications of these novel materials such as signal processing and biosensing devices.

Contributed Papers

2:20

2pSA5. Nonlinear behavior of heterogeneous materials containing snapping acoustic metamaterial inclusions. Stephanie G. Konarski, Kyle S. Spratt, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

This work studies the forced dynamical behavior of a heterogeneous material containing metamaterial inclusions undergoing large deformations. The inclusions exhibit non-monotonic stress-strain behavior, modeled with an expansion to third order in volume strain, where the coefficients of the expansion depend on the metamaterial structure. The resulting constitutive behavior of interest displays regimes of both positive and negative stiffness and the inclusion therefore exhibits hysteretic snapping when forced by an acoustic pressure. Two cases are explored using a generalized Rayleigh-Plesset analysis to model the large-deformation dynamics of the metamaterial inclusion following an approach similar to Emelianov *et al.* [J. Acoust. Soc. Am., **115**, 581 (2004)]. The first case focuses on the forced dynamics of a single inclusion embedded in a weakly compressible elastic medium. The second case broadens the model to analyze the behavior of a heterogeneous material comprised of a low volume fraction of non-interacting metamaterial inclusions embedded in a weakly compressible material. Finally, estimates of the effective bulk modulus and loss factor of the heterogeneous medium are presented for instances of the forcing pressure inducing either large or small inclusion deformation. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and the Office of Naval Research.]

2:35

2pSA6. Large-amplitude stress waves in nonlinear periodic structures. Pai Wang, Filippo Casadei, and Katia Bertoldi (SEAS, Harvard Univ., 29 Oxford St., Cambridge, MA, pai@seas.harvard.edu)

The ultimate goal of this research is to investigate the propagation of large-amplitude stress waves in nonlinear periodic structures. Sources of non-linearity are associated with large-strain kinematics, material non-linearity, and bifurcation paths. In this study, we use a numerical approach to investigate the propagation on large strain waves in periodic lattice structures of finite size. Insights on the dispersion properties of such systems, and their functional dependence on the strain levels, are obtained by post-processing the time-history results obtained through time-domain explicit simulations. In particular, we highlight the effects of nonlinear amplitude parameters on the bandgaps and wave directionality of the considered systems.

2:50

2pSA7. Control of the dynamic properties of nano cantilevers with structural imperfections. Marcus Rutner, Dimitri Donskoy, and Mark Conticchio (Civil, Environ., and Ocean Eng., Stevens Inst. of Technol., Castle Point on Hudson, Hoboken, NJ 07030, mrutner@stevens.edu)

Acoustic metamaterials can be made out of micro/nano size structures employing various structural elements such as cantilever oscillators [JASA, **132**(4), 2866–2872]. Natural frequencies and eigenmodes depend on stiffness and mass distribution which should be reflected in macro as well as micro (nano) structure analysis. This study comprises three parallel approaches to model nano cantilever beams, i.e., the analytical method, the finite element analysis, and the molecular/atomic dynamics analysis, to identify how material imperfections influence the dynamic response of the nanocantilever. The study explores to what extent and under what circumstances macrostructure mechanics differs from nanostructure molecular/atomistic mechanics and how the built-in imperfection can be used to control structural dynamic properties at various scales.

Session 2pSC

Speech Communication: Determinants of Speech Perception: Session in Honor of Joanne L. Miller

Rachel M. Theodore, Cochair
Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269

Robert E. Remez, Cochair
Dept. of Psychol., Barnard College, Columbia Univ., 3009 Broadway, New York, NY 10027

Chair's Introduction—1:00

Invited Papers

1:05

2pSC1. Talker-specific influences on phonetic category structure. Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

A major goal of research in spoken language processing has been to describe how listeners achieve stable perception given the marked variability in mapping between the acoustic signal and linguistic representation. Toward this end, the research of Joanne L. Miller and colleagues has shown that speech sound categories, like other cognitive/perceptual categories, have rich internal structure, with category membership represented in a graded fashion. Moreover, category structure robustly shifts as a function of variation in the speech signal including variation associated with phonetic context, speaking rate, and dialect. I will discuss evidence indicating that internal structure also reflects talker-specific phonetic variation. These experiments concern talker differences in voice-onset-time (VOT), an acoustic parameter that marks the voicing distinction in stop consonants. I will begin with findings from production experiments that explicate talker differences in VOT with the goal of generating predictions for how listeners might accommodate such differences. I will then present findings from perception experiments indicating that listeners comprehensively adjust the mapping between the acoustic signal and phonetic category to reflect a talker's characteristic VOT distribution. Collectively, these findings demonstrate that listeners accommodate variability in the speech stream by dynamically adjusting internal category structure in light of systematic acoustic variation.

1:25

2pSC2. Measuring visual contributions in phonetic categorization. Lawrence Brancazio (Psych., Southern Connecticut State Univ., 501 Crescent St., New Haven, CT 06410, brancaziol1@southernct.edu)

Although most of Joanne Miller's work has explored the mapping from the acoustic signal to phonetic categories, she has also investigated the contributions of visual phonetic information in this process (Green and Miller, 1985). Audiovisual integration is commonly assessed using the McGurk effect, an illusion occurring with audiovisually conflicting stimuli. We (Brancazio and Miller, 2005) suggested that the McGurk effect may underestimate visual contributions in speech perception, in part based on our finding that visual speaking rate influences phonetic judgments when the McGurk effect does not occur (with stimuli that typically produce the effect). Rather, audiovisual integration of incongruent stimuli might result in percepts that fall between phonetic categories—including the one consistent with the acoustic signal—and then are mapped onto one of these categories. Thus, variability in the incidence of the McGurk effect might reflect variability in the process of mapping onto phonetic categories more so than variability in audiovisual integration. In recent work, I have sought to disentangle some factors that might influence the magnitude of the McGurk effect. I will describe findings from recent studies involving variations on the standard McGurk paradigm, and discuss the implications for developing finer-grained methods of assessing audiovisual integration.

1:45

2pSC3. Neural sensitivity to phonetic category structure. Emily Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Storrs Mansfield, CT 06269, emilybmyers@gmail.com)

Much of the recent work on the neural bases of speech perception has concentrated on the phenomenon of categorical perception, and in particular the observation that listeners appear to ignore or be insensitive to acoustic variation within the phonetic category. This has led to a search for phoneme-level processes in the brain, while neglecting a crucial aspect of the speech category, namely, that phonetic categories contain rich internal structure. Pioneering research on phonetic category structure from Joanne L. Miller and her colleagues shows us that listeners are not only sensitive to variation within the phonetic category, but that this sensitivity is modulated by speech rate, context, and talker identity. Recent work from our group using fMRI shows that the neural systems underlying phonetic processing are likewise sensitive to phonetic category structure, with neural responses in the temporal lobes that reflect to the "goodness

of fit" of a token to its phonetic category. In this paper I discuss evidence for the neural encoding of phonetic category structure, and in particular the sensitivity of this encoding to context and to experience.

2:05

2pSC4. Some neuromyths concerning the effectiveness of cochlear implants: Inferring function from dysfunction. David B. Pisoni (Psychol. and Brain Sci., Indiana Univ., 1101 E 10th St., Bloomington, IN 47405, pisoni@indiana.edu) and David B. Pisoni (Dept. of Otolaryngol. - HNS, Indiana Univ. School of Medicine, Bloomington, IN)

Cochlear implants provide profoundly deaf infants and young children with access to critical temporal and spectral patterns of speech needed for language development. Despite the enormous benefits of cochlear implants, many controversial questions remain. I will consider ten pressing issues that are the focus of current efforts in the field: (1) individual variability in speech and language outcomes, (2) neuroplasticity and early implantation, (3) learning and linguistic experience, (4) linguistic vs indexical channels in speech perception, (5) bilateral vs. unilateral implantation, (6) Contribution of other neural and cognitive systems, (7) early predictors of outcomes and risk factors, (8) workload and mental effort, (9) speech in noise, and (10) acoustic simulations in normal-hearing listeners. These issues raise important theoretical questions about basic processes in speech perception and spoken language processing. Cochlear implant research may be viewed as a "model system" enabling inferences about normal function from the study of dysfunction providing additional converging support for the proposal that the ear does not function in isolation from the rest of the brain; it is an inseparable component of a complex adaptive self-organizing system that evolved to support the perception and production of spoken language. [Work supported by NIH Grants: R01 DC-XYZ and DC-ABC to Indiana University.]

2:25

2pSC5. Effects of early auditory deprivation on auditory-visual development. Derek Houston (Otolaryngol. - Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr., RR044, Indianapolis, IN 46202, dmhousto@indiana.edu)

Auditory perception does not develop in isolation. In typically developing infants, the auditory system develops integrally with other sensory and motor systems. This integrality is disrupted in deaf infants. Even those who gain access to sound through cochlear implantation undergo a period of auditory deprivation where the other sensory systems develop independently from audition before access to sound begins. In this presentation, I will report findings from several studies of infants and toddlers with cochlear implants where we have found that they perform more poorly on auditory-visual integration and association tasks (e.g., novel word learning) than normal-hearing peers. Moreover, longer periods of auditory deprivation correlate with poorer performance on auditory-visual association tasks. I will discuss the implications of these findings on sensitive period of language acquisition. I will also present preliminary results from a new study that is further exploring multi-modal integration and learning in hearing-impaired infants. We are using head-mounted cameras, eye trackers, and microphones to analyze multi-modal communicative interactions between infants with hearing loss and their parents during free play sessions. By investigating the dynamics between multi-modal input and interactive behavior, we hope to gain important insights into how impaired sensory integration affects communicative interactions and multi-modal learning.

2:45

2pSC6. Understanding the fine structure of speech: Contributions of Joanne L. Miller. Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195, pkkuhl@u.washington.edu)

As a graduate student in the 1960s, Joanne Miller was extremely well organized, exceptionally attentive to details, and very goal directed. When Joanne directed that level of attention toward the fine structure of speech and to theories explaining how human perceivers (both adult and infant) deciphered it, many important discoveries were made. In this talk, I'll review some of the data and theoretical arguments that Joanne Miller has put forward, and show how they advanced the field.

3:05–3:30 Break

Contributed Papers

3:30

2pSC7. Imitation of a talker improves perception of the talker's speech. James W. Dias, Theresa C. Cook, Dominique C. Simmons, Josh J. Dorsi, and Lawrence D. Rosenblum (Psych., Univ. of California, Riverside, 900 University Ave., Riverside, CA 92521, jdias001@ucr.edu)

Human perceivers have a tendency to imitate the idiolect (talker-specific articulatory style) of a perceived talker. This phonetic convergence manifests between perceivers during live conversations and when perceivers shadow (say aloud) the speech spoken by a pre-recorded talker. The following investigation explores the potential facilitative effects of phonetic convergence on subsequent perception of the imitated talker's speech. To test this, the strength of phonetic convergence was manipulated by varying the delay of perceivers' shadowing responses. When perceivers are required to delay their shadowing responses, phonetic convergence has been found to reduce [Goldinger, Psychol. Rev. **105**(2), 215–279 (1998)]. If phonetic convergence can facilitate later perception of speech spoken by the shadowed talker, then perceivers who immediately shadow a talker should better identify the speech spoken by that talker, compared to perceivers who delayed their shadowing responses. Results suggest immediate—but not delayed—shadowing facilitates later identification of shadowed words spoken by the shadowed talker, compared to a non-shadowed talker. However, the facilitation effect does not transfer to utterances of novel (non-shadowed) words. The results may suggest a link between the mechanisms of speech perception and speech production based on idiolectic information available within lexical episodes.

3:45

2pSC8. Facilitating perception of speech in babble through conceptual relationships. Sara Guediche, Megan Reilly, and Sheila E. Blumstein (Dept. of Cognit., Linguistic, and Psychol. Sci., Brown Univ., 190 Thayer St., Providence, RI 02906, Sara_Guediche@brown.edu)

Speech perception is influenced by many different sources of information. Here we examine whether auditorily presented conceptual/semantic information facilitates the perception of degraded speech. To this end, acoustically clear sentences preceded sentences presented in speech babble. These sentence pairs were either conceptually related, conceptually

unrelated, or the same. The conceptually related and unrelated sentence pairs did not share any content words. Behavioral results show the highest accuracy for the same condition, followed by the conceptually related condition. The worst performance was in the unrelated condition. We then examined the neural substrates of this effect using fMRI. Preliminary results show that a direct contrast between related and unrelated sentences recruits a semantic/conceptual network including fronto-parietal and subcortical areas. The same sentence pair condition showed relatively greater activation in the superior temporal gyrus compared to the other two conditions. These findings suggest that the type of congruency (acoustic-phonetic vs. conceptual) enhances the activation of different functional networks. They also suggest that conceptually related sentences override a reliance on temporal lobe structures typically activated in resolving the perception of degraded speech.

Invited Papers

4:00

2pSC9. Dances of the tongue: Temporal cues and temporal context in the production and perception of vowels. Winifred Strange (Retired, P.O. Box 1226, Anna Maria, FL 34216, strangepin@aol.com)

One focus of the research program of Joanne Miller and her colleagues has been on the influence of temporal context (including speaking rate) on the perception of consonant contrasts differentiated primarily by temporal cues (e.g., Voice Onset Time as a major cue for voicing contrasts; formant transition durations for manner contrasts). This talk will summarize some of my own findings on the role of temporal context (speaking style and rate) in the perception of vowels by native and non-native speakers of American English (AE). In AE Consonant-Vowel-Consonant syllables, duration varies systematically, but redundantly, among vowels contrasting in vowel "height," cued primarily by first formant (F1) target frequency. Additional patterns of F1 temporal trajectories (timing of F1 maximum frequency and temporal symmetry/asymmetry of F1 onset and offset trajectories) differentiate so-called tense (long) and lax (short) AE vowels. Results of this research support the conception of speech production as a rhythmic activity with nested levels of timing control. Temporal patterns in syllabic- and multisyllabic-length segments of speech simultaneously provide information for phonetic identity and super-segmental structure within an exquisitely choreographed "dance" of the articulators. Differences in perception by native and non-native listeners support the conclusion that these are language-specific, learned patterns.

4:20

2pSC10. The role of systematic variation in speech perception. Lynne C. Nygaard (Dept. of Psych., Emory Univ., Atlanta, GA 30322, lnygaar@emory.edu)

A signature problem in our understanding of spoken language processing is the highly variable nature of the speech signal. The realization of linguistic form changes profoundly from utterance to utterance, individual speaker to individual speaker, and group of speakers to group of speakers. On the one hand, this high degree of variation is a problem for accounts of speech perception and spoken language comprehension. Listeners must maintain robust perceptual constancy in the face of the enormous variability in the instantiation of linguistic form. On the other hand, listeners are sensitive to the fine-grained structure of linguistic segments that signals differences among talkers and speaking styles. Variation is informative providing important cues to attributes of the individual speaker and social context. Empirical and theoretical work will be presented that attempts to reconcile the stability of speech perception with the informative nature of systematic variation. This work suggests that listeners both dynamically adapt to systematic changes in linguistic category structure and encode linguistically relevant variation in representations of spoken language. The findings suggest considerable behavioral and representational plasticity in speech perception and spoken language processing and highlight the importance of lawful variation for spoken communication.

4:40

2pSC11. Talker contingency in spoken communication. Robert E. Remez (Dept. of Psych., Program in Neurosci. & Behavior, Barnard College, Columbia Univ., 3009 Broadway, New York, NY 10027, remez@columbia.edu)

Those who speak the same language share its words, yet each spoken expression is unique, nonetheless. Research on the shared linguistic attributes perceived from highly varied physical acoustics has classically invoked the perturbations that drive expression from canonical phonemic form: coarticulation, articulatory rate variation, and anatomical scale variation. More recently, studies confirm that affect, dialect, idiolect, and idiosyncrasy also shape phonetic expression. Is the perception of this acoustic-phonetic variation prothetic, in which graded variation in physical acoustics creates similarly graded phonetic impressions? Some investigations of American English voicing encourage this view, though descriptive and perceptual studies also show that subphonemic phonetic expression can vary discontinuously in production and in perception. Overall, such studies of talker contingency in production and perception constrain claims of grammatical governance of articulation, and define new challenges of perceptual explanation.

5:00–5:30 Panel Discussion

Session 2pSP

Signal Processing in Acoustics: Session in Honor of William M. Carey II

Edmund Sullivan, Chair
Prometheus Inc., 21 Arnold Ave., Newport, RI 02840

Contributed Papers

1:30

2pSP1. Spatial coherence and radiated power for sound propagating in complex ocean environments. Timothy Duda (Woods Hole Oceanogr.Inst., AOPE Dept. MS 11,Woods Hole, MA 02543, tduda@whoi.edu)

Underwater sound propagation in areas of complex bathymetry, variable water masses (fronts), or variable stratification can result in detailed patterns of signal phase and amplitude. Here, we examine a few of the countless scenarios with numerical simulation. Synthetic horizontal arrays can be laid down within the acoustic fields produced with time-varying three-dimensional acoustic simulations. The power received at the phones of the arrays and the spatial structures of amplitude and phase can be used to create an estimate of array-exploitable power transmitted from a source at a known location to positions throughout the environment. Maps of this parameter and its components (horizontal coherence length and incoherent power arriving at the arrays) are presented for multiple simulated environments, including offshore of Southern California and offshore of the eastern seaboard of the United States.

1:45

2pSP2. Arrays and signal processing during the "Nantucket Sound Experiment": A review of work in honor of William M. Carey. Jason D. Holmes (Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02375, jholmes@bbn.com)

Bill Carey was an engineer at heart and he loved finding practical solutions to important sonar problems. The development of a small hydrophone array towed by an unmanned vehicle built off of practical experience in one area of Bill's expertise (towed array signal processing and coherence length) and served as a practical tool to investigate another area he was interested in (acoustics of bottom interaction). This paper discusses what Bill referred to as the "Nantucket Sound Experiment" in which the small towed array was deployed. The development of the array, its use to measure sediment properties by forming a long synthetic aperture, and the relationship between synthetic array performance and coherence are discussed. Particular attention is paid to Bill's insight into the mechanisms that influenced array performance for various aspects of the experiment.

Invited Papers

2:00

2pSP3. The "Carey Number" continued—Mechanisms governing the horizontal array coherence length in shallow water acoustics. James F. Lynch, Timothy F. Duda, Ying-Tsong Lin, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 209 Bigelow Lab, MS #11, Woods Hole, MA, jlynch@whoi.edu)

The use of simplified "feature models" (geometric idealizations of specific, isolated ocean features) for coastal oceanographic features can allow one to calculate acoustically useful quantities approximately and even generate analytic forms for them. Feature models for coastal fronts, eddies, internal tides, linear and nonlinear internal waves, and spice are presented, and the scattering of sound from these objects is calculated. This allows one to estimate the useful quantity Lcoh, the horizontal coherence length that represents a physical limit for array signal processing. Continuing work on calculations of Lcoh, including source/receiver motion, will be presented. [Work sponsored by the Office of Naval Research.]

2:15

2pSP4. Bill Carey as a scientist, motivator, sponsor, and colleague. Thomas G. Muir, D. P. Knobles (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu), and Clark Penrod (Appl. Res. Labs., Univ. of Texas at Austin, Austin, Vermont)

Bill Carey's long association with the Applied Research Laboratories at the University of Texas at Austin (ARL:UT), gave us the pleasure and good fortune to be associated with him in a wide range of pursuits, over many years. His insight and fortitude as a scientist and motivator led a number of us into productive scientific research that we may not have otherwise undertaken. As a sponsor at the Defense Advanced Research Projects Agency (DARPA), he posed visceral questions with unique viewpoints, sometimes in forceful terms, which led us to undertake difficult and productive projects. Some of these projects are linked to the series of shallow water sea trials called the Area Characterization Tests, which proposed and tested hypotheses, while collecting invaluable data to delineate the acoustics of significant scenarios in littoral ocean environments. We were privileged to participate in these experiments as well as analyze and model the results. As a colleague, while at other institutions, Bill provided the motivation and encouragement for us to undertake many projects, including a seismo-acoustic experiment with the SACLANT Undersea Research Center, which demonstrated the dispersive beam-forming concept for geophone arrays on the sea floor and for the detection and study of Sholte type interface waves. We pay tribute to Bill Carey in this talk by illustrating some of the discoveries made in a number of at-sea experiments and the theoretical and modeling work they spawned. [Work supported by ARL:UT Austin.]

2:30–2:45 Break

2:45

2pSP5. William M. Carey and the nonlinear frequency dependence of low frequency attenuation in sandy sediments. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), Richard B. Evans (Retired, North Stonington, CT), and Jason Holmes (Sensing and Control Systems, Raytheon BBN Technologies, Cambridge, MA)

The early "conventional wisdom" was that low frequency sediment attenuation could be predicted with a downward extrapolation with attenuation presumed directly proportional to frequency. This goes back to work at Woods Hole reported in a 1968 Bryn Mawr doctoral thesis by Bennett. However, the attenuation at such low frequencies is difficult to measure directly. In subsequent years, a number of investigators, including Carey and colleagues, using a variety of experimental techniques (all indirect) different from that of Bennett, found frequency dependences with exponents substantially larger than unity. A survey paper by Holmes, Carey, Dediu, and Siegmann in 2007 concluded that the appropriate exponent was approximately 1.8. Carey, however, felt that a substantial part of the community still clung to the belief that the dependence was linear, and led an effort to garner support from basic theory. Analytical models, including that of Biot and others based on a rigorous examination of water-sand-pebble interaction at low frequencies, predicted that the attenuation should be quadratic in frequency. Carey conjectured that the discrepancy was because the inference technique ignored the possibility of shear waves in the sediment. That this explained the discrepancy was subsequently confirmed with computations by Collis and theoretical analysis by Pierce.

Contributed Paper

3:00

2pSP6. Comparison of shallow water mode transport theory with acoustic transmissions during Shallow Water Experiment 2006. Kaustubha Raghukumar, John A. Colosi (Oceanogr., Naval Postgrad. School, 315B Spanagel Hall, Monterey, CA 93943, kraghuku@nps.edu), Ying-Tsong Lin, Timothy F. Duda, Arthur Newhall (Appl. Ocean Phys. & Eng. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA), Kyle M. Becker (Appl. Res. Lab., Penn State Univ., State College, PA), and Paul Hines (Defence R&D Canada – Atlantic, Dartmouth, NS, Canada)

Coupled-mode transport theory appears to have put on solid theoretical ground acoustical scattering by internal waves in both deep and shallow water, over a range of low, medium, and high frequencies [Raghukumar and Colosi (2014) and Colosi and Morozov (2009)]. Here, full-field theoretical calculations of the acoustic field moments are compared against experimental data gathered during the Shallow Water Experiment 2006. Transport theory at low frequency is validated using data gathered by the WHOI Shark array at 175 Hz with a source towed by R/V Sharp at several different speeds over distances of 1.5–5.5 km. At high frequencies, comparisons are made at 1.2 kHz using data received by a WHOI bottom-mounted single hydrophone unit with a source towed by CFAV Quest over distances of 0–20 km. Acoustic observables include the mean and variance of intensity. The effect of a range dependent stochastic internal wave field is examined in the context of data-model comparison, along with the effect of random surface waves. In addition, further insights into mode coupling are presented using the shallow water hybrid transport theory.

Invited Paper

3:15

2pSP7. Bill Carey and passive synthetic aperture. Edmund Sullivan (Prometheus Inc., 21 Arnold Ave., Newport, RI 02840, ejsul@fastmail.fm)

Although it was long a controversial subject in the acoustics community, passive synthetic aperture remained of great interest to Bill Carey. Over the years, there were several "proofs" that it couldn't be done in any practical way. These proofs were technically correct in terms of the models upon which they were based, but it eventually became clear that these models were quite constraining in that they had no relation to actual practice and further, were hamstrung in that they were centered on the concept of beamforming. It is shown how passive synthetic aperture was placed on a firm theoretical basis by avoiding the focus on a synthetic "beam pattern" and treating the bearing and range estimation as pure estimation problems, where the approach is based on a joint estimation of bearing and source frequency, or in the case of the wavefront curvature problem a joint estimation of bearing, source frequency and range. A history of Bill's contributions to the area is outlined and explanations of the shortcomings of the so-called "proofs" are discussed. Several examples of experimentally verified results are outlined and several examples are given.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C.J. Struck, Chair ASC S3
CJS Labs, 57 States Street, San Francisco, CA 94114-1401

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, *6* May 2014,

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 6 MAY 2014 PROVIDENCE 1, 3:45 P.M. TO 5:00 P.M.

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

,

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

D.S. Houser, Vice Chair ASC S3/SC 1
National Marine Mammal Foundation, 2240 Shelter Island Drive, Suite 200, San Diego, CA 92106

Accredited Standards Committee S3/SC 1 **on Animal Bioacoustics.** Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 6 May 2014.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

TUESDAY EVENING, 6 MAY 2014 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings.

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

Committees meeting on Tuesday are as follows:

4:30 p.m.	Engineering Acoustics	550AB
7:30 p.m.	Acoustical Oceanography	552AB
7:30 p.m.	Architectural Acoustics	555AAB
7:30 p.m.	Physical Acoustics	551AB
7:30 p.m.	Psychological and Physiological Acoustics	554AB
8:00 p.m.	Structural Acoustics and Vibration	553AB