Session 4aAA

Architectural Acoustics, Signal Processing in Acoustics, and Noise: Methods and Techniques Used for Simulation of Room Acoustics

Bruce C. Olson, Cochair
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Chair's Introduction—8:15

Invited Papers

8:20

4aAA1. Level of detail in room-acoustic simulation. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

The quality of present-day room acoustic simulations depends on the quality of the boundary conditions and of the underlying CAD room models. A "high-resolution" room model does not mean that it needs to have a visually perfect geometrical fine structure. To our experience, the required resolution of objects or surfaces does not need to be higher than about 1 m. In this presentation, an auralization engine is briefly introduced which uses a set of models of the same room but with a graduated level of detail (LOD). These different models can account for more physical correctness especially for very low-frequency specular reflections. Furthermore, a good estimate of scattering coefficients is essential. The relevance of the uncertainty of scattering coefficient data is discussed in a review on perception tests with varied surface scattering. Finally, guidelines for creation of CAD models are proposed.

8:40

4aAA2. Should we still rely on statistical calculations for the prediction of reverberation time? Ana M. Jaramillo (AFMG Services North America, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu) and Bruce Olson (Olson Sound Design, LLC, Brooklyn Park, MN)

Based on the conditions for the use of the most commonly used reverberation time equations, we have created room examples in EASE to compare how they correlate with ray tracing predictions and established a guideline on when we can rely on simple statistical predictions. The results show that statistical predictions are not always accurate, and the differences do not always go in the same direction, making it impossible to simply account for the under/over-estimation of the method.

9:00

4aAA3. Modelling the effects of spectators on speech intelligibility in a typical soccer stadium. Ross Hammond (School of Mathematics, Computing and Electronics, Univ. of Derby, Derby, United Kingdom), Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com), and Adam J. Hill (School of Mathematics, Computing and Electronics, Univ. of Derby, Derby, United Kingdom)

Public address system performance is frequently simulated using acoustic computer models to assess coverage and predict potential intelligibility. Simulations are most-often completed in unoccupied spaces as this provides worst-case scenario intelligibility due to the reduced absorption. When the typical 0.5 speech transmission index (STI) criterion cannot be achieved in voice alarm systems, due to design difficulties, justification must be made to allow contractual obligations to be met. An expected increase in STI with occupancy can be used as an explanation, though the associated increase in noise levels must also be considered. However, numerous approaches exist when modelling the people which can produce significant discrepancies. This work demonstrates typical changes in STI for different spectator conditions in a calibrated stadium computer model. This includes different audience modelling approaches, distribution, capacity, posture (standing/seated), and atmospheric conditions. The effects of ambient noise are also considered. The results can be used to approximate expected changes in STI caused by different spectator conditions.

9:20

4aAA4. Acoustics simulations to inform the designs of large worship and entertainment spaces to the client and contractor. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Computer based acoustical simulations can quickly communicate important information in visual and audible formats that have a strong and immediate impact on the non-acoustician decision makers and designers of a project. Two large spaces were modeled ($>$6000 m^3), one renovation, one new construction; simulations were used to help better understand the consequences of different design approaches for amplified sound and acoustical design, as well as handing value engineering response in a timely manner. Attention is given to the modeling's role in helping to sort out the paradigms of perception of the project team and then to inform the design options of the clients and end users.

9:40

4aAA5. Use and misuse of auralization. Wolfgang Ahnert (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

Auralization was developed as a tool in the 30s. The historic overview over this development starts by using scale models as a design tool which is used until now. Here, the needed components are explained, and the pros and cons will be discussed. With the use of computer simulation in the end of the 60s, the presentation of auralized files started around 1990 first considered as a toy. Today the use of auralization is widespread. This paper describes the development of the technical tools to present auralized signals available for binaural reproduction without and with head tracker and by using loudspeaker reproduction without and with crosstalk cancelation. Nowadays, Acoustic labs with Ambisonics reproduction or similar technologies are used. In this presentation, the advantages of auralization are named including all positive properties to demonstrate the achieved simulation results to different client groups. But also, the misuse of auralization is shown in detail by using found examples.

10:00–10:15 Break

10:15

4aAA6. New tools to auralize simulation results with EASE 5.0. Tobias Behrens (ADA Acoust. & Media Consultants GmbH, Arkonastr. 45-49, Berlin 13189, Germany, tbehrens@ada-amc.eu), Khaled Wazaefi (ADA Acoust. & Media Consultants GmbH, Berlin, Deutschland, Germany), and Wolfgang Ahnert (AFMG Ahnert Feistel Media Group, Berlin, Germany)

The new software package in EASE allows the production of binaural files to check the quality of the simulation results and to make these results audible for music or speech samples in real time. Also, tests with head trackers have been made. Since 5 years, the software allows generating B-format files of second order. To reproduce sound fields based on these 9 files, a sound lab has been built. This lab will be represented and explained. A post-processing software for EASE allows to reproduce not only the calculated simulation files but in comparison also measured files by using the microphone Ambeo VR. Additionally, VR glasses generate realistic 3D-visuals, in the same model as used for acoustic simulation. That way realized acoustic treatments in the room become visible and audio-visual representations are possible. Results for comparison between simulated rooms and measured real rooms will be discussed.

Contributed Papers

10:35

4aAA7. Room acoustic simulation as a means to affect a musical composition for a location specific performance. Edwin S. Skorski (Interior Architecture, Univ. of North Carolina - Greensboro, 102 Gatewood Studio Arts Bld., 527 Highland Ave., Greensboro, NC 48859, skors1es@cmich.edu) and Steven J. Landis (Music, Univ. of North Carolina - Greensboro, Greensboro, NC)

Computer model simulations of existing interior spaces are often generated to document and analyze room acoustic characteristics. In this case study, a large, multi-tiered public atrium is analyzed for its potential use as a performance space. Furthermore, the analysis is also used to transform an existing musical composition into a location specific performance piece. The computer simulation highlights acoustic characteristics believed to be good for musical performance as well as those considered defects. Taking into account the unique room acoustic qualities of this non-traditional performance space, an existing musical composition is rewritten resulting in a space-dependent arrangement. Musical variables transformed due to the analysis include tempo, pacing, register, as well as source and receiver positions. Of specific interest are the room characteristics typically considered acoustic defects which are purposefully exploited to strengthen the impact of the performed piece. These include non-optimal reverberation times, sound focusing, and echoes. Acoustic analysis of the room and recordings of the composition will be presented.

10:50

4aAA8. An analysis of ceiling geometry within active learning classrooms. Edwin S. Skorski (Interior Architecture, Univ. of North Carolina - Greensboro, 102 Gatewood Studio Arts Bld., 527 Highland Ave., Greensboro, NC 48859, skors1es@cmich.edu)

The architectural designs and furnishing of active learning classroom spaces are playing an increasingly important role in the facilitation of modern educational methods. Traditional static classroom spaces effectively support a lecture style of teaching where student participation is passive. Due to their rigid space plan, they are poor at encouraging interaction among students and teachers. Conversely, active learning spaces promote innovative teaching methods where quick room re-configuration allows for discussion groups of various sizes, the simultaneous use of a variety of teaching methods, and provides greater opportunity for the incorporation of technology into the classroom. From a room acoustic perspective, the increase in room arrangement flexibility leads to a complex acoustic environment where the spatial relationship between the source and the receiver is highly variable. This study uses digital modeling and computer simulation to analyze the effects of the ceiling geometry as it relates to the active learning classroom acoustic environment. Specifically, the study focuses on changes in speech intelligibility and reverberation time as the overhead plane is manipulated.

11:05

4aAA9. Evaluation of shape grammar-generated diffuser arrays. Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, 535 W. Michigan St., IT 371, Indianapolis, IN 46202, hsut@iu.edu) and Jonathan Dessi-Olive (Architecture, Georgia Inst. of Technol., Atlanta, GA)

This paper proposes a means of evaluating arrays of quadratic residue diffusers (QRDs) generated through a grammar-based generative design method. Design processes for architectural acoustics are often highly conventional: acoustical designers have preferred to use historical examples, known equations, and standard principles of performative success. This is particularly true for surface treatments using diffuser products that are aggregated in ways that perform sufficiently but are visually predictable and monolithic. In the first phase of this project, a shape grammar approach to design acoustic diffuser arrays was proposed as a means of addressing the issue of design homogeneity in architectural acoustics and to break current habits of uniform deployment of diffusion treatments in spaces. A set of shape rules were proposed that generate non-uniform and sometimes surprising arrays of QRDs. This paper aims to expand demonstrate phase two, which includes the following: (1) clarification and further development of shaper grammar rules, (2) proposal of initial methods to evaluate the acoustic performance of these arrays, and (3) calculation of quantitative metrics. Numerical simulations will show time and directivity responses for these shape grammar generated diffuser arrays. Furthermore, diffusion and scattering coefficients will be presented as well as other proposed evaluation metrics for these larger arrays.

11:20

4aAA10. Design of a multiple-slope sound energy decay system with string and block coupling methods. Xuhao Du (Dept. of Mech. Eng., Univ. of Western Australia, 35 Stirling Hwy., Perth, WA 6009, Australia, xuhao.du@uwa.edu.au), Jie Pan (Dept. of Mech. Eng., Univ. of Western Australia, Crawly, WA, Australia), and Andrew Guzzomi (Dept. of Mech. Eng., Univ. of Western Australia, Perth, WA, Australia)

Driven by the need of both speech intelligibility and music perception, multiple-slope sound decay in a room has been studied for decades. To step further, two different room coupling methods are proposed for achieving decay with quadruple, quintuple, or higher slope numbers. Starting from a triple-slope sound decay system, the understanding of energy flow in such a system is developed as well as the relationship between the decay turning points and the aperture size. Each decay slope represents the acoustic characteristic of the corresponding dominating room, and the dominating durations are determined by the couple condition. Based on the understanding of its energy flow, two different coupling methods, the string coupling and block coupling, are developed for achieving specific non-exponential sound decay with the specific number of slopes, slope values, and times of turning point. These are controlled by parameters including the room quantity, the reverberation of each room, and the coupling aperture sizes. Based on the above coupling methods, a few examples are simulated with the diffusion equation for verifying the energy flux and achieving a specific multiple-slope sound energy decay pattern inside the room.

THURSDAY MORNING, 16 MAY 2019 — CLEMENTS, 8:30 A.M. TO 10:00 A.M.

Session 4aAB

Animal Bioacoustics: Marine Mammal Bioacoustics

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Contributed Papers

8:30

4aAB1. Best available science? Are NOAA fisheries marine mammal exposure noise guidelines up to date? Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

NOAA fisheries employs a set of in-water noise exposure guidelines that establish regulatory thresholds for ocean actions that impact marine mammals. These are established based on two impact criteria: level A—a physiological impact including “Permanent Threshold Shift” (PTS) and level B—a behavioral impact or disruption. Recently, the level A exposure thresholds were reconciled to the frequency-dependent hearing sensitivities of five classes of marine mammals based on the work done more than a decade ago (Southall *et al*., 2007). Since that time, much more work has been published on behavioral impacts of various noise exposures, and consideration of more variables such as frequency-dependent noise propagation characteristics, cumulative, concurrent, and continuous exposures, and noise impacts on marine soundscapes has entered into the discussion—but have not been incorporated into the NOAA Fisheries guidelines. Some of these variables will be highlighted, suggesting that it may be time to reevaluate the thresholds for level B exposures.

8:45

4aAB2. Detection and classification beaked whale vocalization calls based on unsupervised machine learning algorithm. Kun Li and Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, Broussard Hall, Rm. 103, 240 Hebrard Blvd., Lafayette, LA 70503, kxl1737@louisiana.edu)

Currently, passive acoustic monitoring (PAM) becomes a more useful tool to record and analyze cetacean’s calls, such as beaked whales and sperm whales. To study the regional population variability of beaked whales in the northern Gulf of Mexico, the acoustic data were collected by the Littoral Acoustic Demonstration Center-Gulf Ecological Monitoring and Modeling (LADC-GEMM) consortium at three different sites in the vicinity of the 2010 oil spill site. One of the challenging goals is an identification of different species of beaked whales autonomously in large datasets. A multi-stage detector was developed to detect and classify the beaked whale species. The data processing results show that two major species of beaked whales (Cuvier’s and Gervais’) were acoustically observed in the study area. The results suggest that Cuvier’s beaked whales tend to be more active at the deeper sites (about 1500 m) and Gervais’ beaked whales prefer the shallower site which is about 1000-m deep. The results highlight the need

for high spatial resolution acoustic monitoring and support the ecological niche hypothesis. The results also show that the acoustic activity for two major species of beaked whale have distributed throughout the day, and beaked whales do not exhibit seasonal preference for the Mississippi Valley site. The important new insights into the population structure and habitat preferences of different species of beaked whales in the northern Gulf of Mexico were obtained.

9:00

4aAB3. Detection of dolphin burst-pulses off Cape Hatteras, North Carolina, correlated to oceanographic features. Stephen B. Lockhart, Mike Muglia, and Lindsay Dubbs (Univ. of North Carolina Coastal Studies Inst., 850 NC 345, Wanchese, NC 27981, sblockhart.z0@gmail.com)

To assess the ecological impact of extracting energy from the Gulf Stream, the University of North Carolina Coastal Studies Institute has deployed a mooring on the continental slope off Cape Hatteras at a depth of 230 m, equipped with an Acoustic Doppler Current Profiler, CTD, and a hydrophone. Analyzing 16 months of data, we automatically detected dolphin "quacks" or "barks", using two detectors. First, we used a pitch detector to automatically detect such signals over a specified range of pitch values. Next, we used a matched filter approach. All detections were reviewed manually to eliminate false alarms. For these signals, we found a strong correlation with temperature and salinity at the bottom; the vocalizations were detected when the water was relatively cooler and fresher. As the Gulf Stream meanders seaward of the mooring site, the temperature and salinity there both decrease. Since this cooler water is higher in nutrients, one explanation for the correlation is that the marine mammals are attracted to this more productive water. Alternatively, the meandering Gulf Stream may influence either (a) the *acoustic propagation* around the mooring and/or (b) the *acoustic noise* around the mooring. Evidence for each alternative will be presented.

9:15

4aAB4. Understanding detectability variations for density estimation of marine mammals. Thomas Guilment, Natalia Sidorovskaia, Kun Li (Dept. of Phys., Univ. of Louisiana at Lafayette, UL BOX 43680, Lafayette, LA 70504-3680, thomas.guilment1@louisiana.edu), and Christopher Tiemann (Phys., Univ. of Louisiana at Lafayette, Austin, Texas)

Passive acoustic monitoring (PAM) makes it possible to obtain reliable observations of marine mammals and to estimate the population density based on detected acoustic cues. PAM surveys offer higher accuracy density estimates than traditional visual surveys as long as the survey design is adequate and the probability of detection is reliably measured. The probability of detection depends on regional bathymetry, the season, the PAM system used, the detection algorithm, and the animal's acoustic apparatus. To improve the accuracy of the density estimation, this study focuses on understanding the relationship between the probability of detection of beaked whales and the detection algorithm used. The study utilizes the PAM data collected by fixed moored stations in the Gulf of Mexico in 2015 and 2017. The detectability function derived from experimental data is compared with the one obtained by modeling for two species of beaked whales (Cuvier and Gervais whales). The results will provide the guidance when and how modeling can be used to obtain reasonable estimates of the probability of the detection function. [Work supported by a grant from The Gulf of Mexico Research Initiative.]

9:30

4aAB5. Effects of click rate on bottlenose dolphin auditory brainstem response signal-to-noise ratio. James J. Finneran (SSC Pacific Code 71510, U.S. Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Univ. at Buffalo, Buffalo, NY)

Maximum Length Sequence (MLS) and Iterative Randomized Stimulation and Averaging (I-RSA) methods allow auditory brainstem response (ABR) measurements at high stimulus rates; however, it is not clear if high rates allow ABRs of a given signal-to-noise ratio (SNR) to be measured in less time than conventional averaging at lower rates. In the present study, ABR SNR was examined in six bottlenose dolphins using conventional averaging at rates of 25 and 100 Hz and the MLS/I-RSA approaches from 100 to 1250 Hz. Residual noise in the averaged ABR was estimated using root-mean-square values of the: waveform amplitude following the ABR, waveform amplitude after subtracting two subaverage ABRs, and amplitude variance at a single time point. For all approaches, residual noise decreased with the increasing measurement time. For a fixed recording time, SNR was highest at rates near 500 Hz, but optimal SNRs were only a few dB higher than that for conventional averaging at 100 Hz. Nonetheless, small improvements in SNR could result in significant time savings in reaching criterion SNR. The time savings allowed by the MLS and I-RSA methods will be discussed for both mean and individual data. [Work supported by U.S. Navy Living Marine Resources Program.]

9:45

4aAB6. Human auditory discrimination of bottlenose dolphin signature whistles masked by noise: Investigating perceptual strategies for anthropogenic noise pollution. Evan L. Morrison and Caroline M. DeLong (Dept. of Psych., Rochester Inst. of Technol., 18 Lomb Memorial Dr., Eastman 2309, Rochester, NY 14620, e.morrison@mail.rit.edu)

Anthropogenic masking noise in the world's oceans is known to impede many species ability to perceive acoustic signals, but little research has addressed how this noise pollution affects the detection of bioacoustic signals used for communication. Bottlenose dolphins use signature whistles which contain identification information. Past studies have shown that human participants can be used as models for dolphin hearing, but most previous research investigated echolocation. In experiment 1, human participants were tested on their ability to discriminate among signature whistles from three dolphins. Participants' performance was nearly errorless (M = 98.8%). In experiment 2, participants identified signature whistles masked by five different samples of boat noise, with different signals to noise ratios. Preliminary results suggest that participants perform worse in lower ratios of signal to noise, that some signature whistles are easier to identify in the presence of noise, and that some noises have more detrimental impacts on whistle recognition. The presence of boat noise may cause participants to use more auditory cues in order to identify whistles, although participants always relied most heavily on frequency contour and duration. This study may provide insight into the impacts of different types of boat noise on dolphin whistle perception.

Session 4aBAa

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Inverse Problems in Biomedical Ultrasound I

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Kang Kim, Cochair
Medicine, University of Pittsburgh, 950 Scaife Hall, 3550 Terrace Street, Pittsburgh, PA 15261

Chair's Introduction—7:55

Invited Papers

8:00

4aBAa1. Acoustic holography for calibration of ultrasound sources and *in situ* fields in therapeutic ultrasound. Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Leninskie Gory, Moscow State Univ., Moscow 119991, Russia and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA, oa.sapozhnikov@physics.msu.ru), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Sergey A. Tsysar, Dmitry A. Nikolaev (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Leninskie Gory, Moscow State University, Moscow 119991, Russia and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA)

Therapeutic ultrasound sources, which are typically piezoelectric transducers, are intended to deliver known acoustic pressures to targeted tissue sites. Each transducer vibrates in a unique way and radiates a corresponding 3D ultrasound field. Accordingly, transducer vibrations should be known accurately in order to characterize the pressures delivered to the patient. Acoustic holography is a technique that relies on hydrophone measurements to reconstruct a source hologram that characterizes transducer vibrations [Sapozhnikov *et al.*, *JASA*, **138**(3), 1515–1532 (2015)]. In this way, a hologram is a signature of each transducer that can be monitored over time for quality assurance. Using holography-defined source boundary conditions, numerical forward projection of the ultrasound field based on the nonlinear wave equation can be used to accurately predict *in situ* temperatures and pressures in heterogeneous media for treatment planning. As such, acoustic holography goes beyond simple hydrophone scans and is uniquely suited to meet clinical needs for quantifying therapeutic ultrasound fields. In this paper, several examples of acoustic holography implementation are presented, including the characterization of single-element and multi-element flat and spherically curved sources working in linear and nonlinear regimes and in continuous and pulsed modes. [Work supported by NIH 1R01EB025187, R01EB007643, and R21CA219793; RFBR 17-02-00261 and 17-54-33034.]

8:20

4aBAa2. Full wave 3D inverse scattering: 21st century technology for whole body imaging. James Wiskin, Bilal Malik, Rajni Natesan, Nasser Pirshafiey, Mark Lenox, and John Klock (R&D, QT Ultrasound, LLC, 3 Hamilton Landing, Ste. 160, Novato, CA 94949, james.wiskin@qtultrasound.com)

Quantitative high resolution (QHR) images of speed of sound and attenuation in human breast have been made using full wave inverse scattering in three-dimension (FWIS3D), where only soft tissue is present. The FWIS3D technology and method are reviewed. Recent QHR images in the presence of bone and gas have been obtained with FWIS3D and are shown. Transmission mode quantitative and refraction corrected reflection images of small piglet abdomen, thorax, and head are shown. QHR images of the human knee using the same technology are shown. Human Knee is difficult due to the predominant presence of bone. With low frequency FWIS3D, the meniscus, structure within the Femur-Tibia (F-T) space, ligaments, and the infrapatellar fat pad can be seen. The intra-condyle space in the Femur is visible. It was earlier established that 3D modelling was necessary for breast. It is shown to be even more important for F-T space and whole body imaging. Quantitative estimates of high speed early development bone are made, and imaging through neo-natal skull is performed. Clear correspondence with known structures even in the presence of gas is displayed. This reveals FWIS3D ultrasound tomography as a 21st century whole body imaging modality.

8:40

4aBAa3. Iterative image reconstruction algorithm for transcranial photoacoustic tomography applications. Joemini Poudel (Dept. of Biomedical Eng., Washington Univ. in St. Louis, 6648 Oakland Ave., Saint Louis, MO 63139, jpoudel@wustl.edu), Lihong Wang (Andrew and Peggy Cherng Dept. of Medical Eng., California Inst. of Technol., Pasadena, CA), and Mark Anastasio (Dept. of Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO)

Photoacoustic computed tomography (PACT) is an emerging computed imaging modality that exploits optical contrast and ultrasonic detection principles to form images of the absorbed optical energy density. The PACT reconstruction problem corresponds to recovering the total absorbed optical density within a tissue sample, from the acoustic waves recorded on a measurement aperture located outside the support of the tissue sample. A major challenge in transcranial PACT brain imaging is to compensate for aberrations in the measured photoacoustic data due to their propagation through the skull. To properly account for these effects, a wave equation-based iterative reconstruction algorithm that can model the heterogeneous elastic properties of the medium is employed. To accurately recover the absorbed optical energy density, complete knowledge of the spatial distribution of the elastic parameters of the medium is required. However, estimating the elastic properties of the medium prior to the experiment is practically infeasible. To circumvent this, we propose to jointly reconstruct the absorbed optical energy density and the spatial distribution of the elastic parameters of the medium from PACT data alone. Reconstructed images from both numerical phantoms and experimental data are employed to demonstrate the feasibility and effectiveness of the approach.

9:00

4aBAa4. Direct error in constitutive equation formulation for inverse problems in time harmonic elastography. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), Olalekan A. Babaniyi (Appl. Mathematics, UC Merced, Merced, CA), and Sanjay S. Yengul (Mech. Eng., Boston Univ., Brookline, MA)

Shear wave imaging techniques allow the evaluation of rigidity and viscosity of tissues locally within a material. From an inverse problem perspective, the approach is quite attractive insofar as it provides a densely sampled displacement field in the interior of the object from which to invert for material properties. We consider several challenges related to elastic wave inverse problems arising in acoustic radiation force imaging. First, we validate an axisymmetric viscoelasticity model suitable for some applications of acoustic radiation force imaging. Second, we consider reconstructing lateral displacement components from measured axial displacement components. Finally, we present a new variational formulation, the direct error in constitutive equation formulation, for inverse problems in time harmonic viscoelastic wave propagation with full-field data. The formulation relies on minimizing the error in the constitutive equation with a momentum equation constraint. Numerical results on model problems show that the formulation is capable of handling discontinuous and noisy strain fields and also converging with mesh refinement for continuous and discontinuous material property distributions. Applications to MRE and ARFI measured wave data are considered.

9:20

4aBAa5. Assessing FES-induced muscle fatigue using ultrasound to determine the inverse neuromuscular model for optimal FES input. Kang Kim (Medicine, Univ. of Pittsburgh, 950 Scaife Hall, 3550 Terrace St., Pittsburgh, PA 15261, kangkim@upmc.edu), Zhiyu Sheng, and Nitin Sharma (Mech. Eng. and Mater. Sci., Univ. of Pittsburgh, Pittsburgh, PA)

Functional electrical stimulation (FES) has been successful in activating paralyzed or paretic muscles to restore limb functions of individuals with impaired gait function. However, when activating the limb joint motion through externally stimulating muscle, rapid onset of muscle fatigue becomes a critical issue that results in injury. To overcome this challenge, an optimal FES input to the neuromuscular system needs to be determined and updated in real-time in order to maintain an effective, safe limb function. The inverse neuromuscular model between the desired limb joint motion and the FES input depends on time varying muscle contractility or fatigue level. In this study, ultrasound speckle tracking is proposed to assess muscle contractility and to establish a dynamic model. To demonstrate the feasibility, isometric knee extension experiments of healthy human participants were conducted with ultrasound imaging on the quadriceps muscle. The consistent decrease in peaks in strain and maximum knee joint torque during each contraction cycle suggest a potential correlation between the strain field and fatigue level of the target muscle. With further validation, ultrasound strain field can be used to solve for the dynamic neuromuscular model and further to determine the optimal FES input. Some technical challenges will also be discussed.

9:40

4aBAa6. Comparison of elastic modulus inverse estimation and the pulse wave velocity estimation for monitoring abdominal aortic aneurysms. Doran Mix, Luke Cybulski, Michael Stoner (Surgery, Univ. of Rochester Medical Ctr., Rochester, NY), and Michael S. Richards (Biomedical Eng., Rochester Inst. of Technol., 1 Lomb Memorial Dr., Rochester, NY 14623, msrbme@rit.edu)

The necessity of surgical intervention of abdominal aorta aneurysms is based on a risk-reduction paradigm primarily relying on trans-abdominal ultrasound (US) measurements of the maximum diameter of an AAA. However, the AAA diameter is only a rough estimate of rupture potential, and elastographic estimates of material property changes within aortic tissue may be a better predictor. This work compares an elastic imaging technique measuring aortic tissue stiffness in cross-section to a pulse wave velocity (PWV) estimate obtained from longitudinal images of the same geometry using a two-dimensional clinical US machine. The elastic imaging technique uses a linear elastic finite-element model to solve the elastic inverse problem and estimates the shear modulus. This technique uses a non-invasive pressure cuff to estimate the pressure in the aorta and normalizes the modulus values. The PWV technique uses geometric measurements and simplifies assumptions to create a direct relation between the wave speed and the modulus. Results of validation studies using aortic mimicking phantoms comparing modulus obtained from each of the techniques are presented. Initial clinical results will be also be presented.

10:00–10:15 Break

10:15

4aBAa7. Super-resolution ultrasound imaging for *in vivo* microvasculature assessment in acute kidney injury mouse model. Qiyang Chen (BioEng., Univ. of Pittsburgh, 3550 Terrace St., 624, Pittsburgh, PA 15261, chenqiyang0404@gmail.com), Brittney M. Rush (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Jaesok Yu (BioEng., Univ. of Pittsburgh, Pittsburgh, PA), Roderick Tan, and Kang Kim (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Acute kidney injury (AKI) is a disease with a high mortality rate and increasing incidence. It also generates a high risk of developing into chronic kidney disease (CKD). The deterioration from AKI to CKD is associated with rarefaction of microvasculature in renal cortex. However, there is lack of well-established diagnostic method that can evaluate the microvasculature changes noninvasively and conveniently with a high spatial resolution during the progression from AKI to CKD. Ultrasound super-solution imaging is an emerging technology that can achieve a high spatial resolution of the vasculature beyond the acoustic diffraction limit by localizing the center of the signals from microbubbles. In this study, deconvolution based super-resolution ultrasound imaging is used to noninvasively assess the microvasculature changes in mouse kidney after AKI. Ultrasound scans on mouse kidneys at 3 weeks and 6 weeks post ischemia-reperfusion injury are compared with control mouse kidneys. Obvious microvasculature reduction due to AKI is identified, which is evidenced by histology. The feasibility of ultrasound super-resolution imaging as a potential diagnostic method for progressive renal disease after AKI is demonstrated in the AKI mouse model.

10:30

4aBAa8. Efficient sub-diffraction passive cavitation imaging. Scott J. Schoen, Zhigen Zhao (Mech. Eng., Georgia Inst. of Technol., 901 Atlantic Dr. NW, Rm. 4125K, Atlanta, GA 30318, scottschoenjr@gatech.edu), and Costas Arvanitis (Mech. Eng. and Biomedical Eng., Georgia Inst. of Technol. and Emory Univ., Atlanta, GA)

Acoustic localization of microbubbles offers a unique method to assess vascular structure and function noninvasively. To this end, passive imaging of the acoustic cavitation with the angular spectrum method (AS-PCI) is appealing as it is inherently fast and frequency-selective and thus allows stable cavitation activity to be isolated from other scatters via the bubbles' harmonic emissions. However, diffraction imposes a physical limitation on the resolution of acoustic imaging systems, which is typically on the order of millimeters for PCI. To enable rapid visualization of vessel structures with diameters of few hundreds of microns, we present a technique based on the AS method for fast super-localization (SL) of multiple, spatially separated bubbles that is 100-fold more efficient than time domain techniques employed for resolution improvement. We demonstrate, via experiments and numerical simulations, that it is possible to super-localize multiple bubbles within a single image and resolve vessels with diameters 10 times smaller than the diffraction limit (300 μm vs. 3 mm, respectively). Furthermore, successive super-localization of hundreds of microbubbles with the proposed SL-AS-PCI method allowed visualization of three-dimensional vessel structures within a few seconds on ordinary hardware. SL-AS-PCI holds great promise for efficient diagnosis of diseases associated with abnormal vasculature.

10:45

4aBAa9. Orientation-dependent anisotropy of acoustic properties of tendons at micrometer scale. Takuya Ogawa (Graduate School of Sci. and Eng., Chiba Univ., Chiba, Japan), Bin Yang, Po Lam (Dept. of Opthalmology, Univ. Of Pittsburgh, Pittsburgh, PA), Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), Ian A. Sigal (Dept. of Opthalmology, Univ. Of Pittsburgh, Pittsburgh, PA), and Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org)

Tendons are bands of fibrous connective tissue connecting muscles to bones. They are composed of parallel arrays of collagen fibers closely packed together which makes them highly anisotropic. The anisotropy of the acoustic properties of tendons was investigated at an ultra-fine resolution (< 7 μm) using quantitative acoustic microscopy (QAM). Chicken tendons were fixed (formalin) while loaded longitudinally, then cryosectioned (16-μm) at several orientations (every 15 deg) from parallel (i.e., 0 deg) to perpendicular to the fibers (i.e., 90 deg). Two regions of two sections per angle were scanned using a QAM system operating at a center frequency of 250 MHz yielding a total of 28 QAM datasets which were processed to yield two-dimensional (2D) maps of the bulk modulus, mass density, acoustic impedance, and speed of sound for each scanned region. Acoustic parameters were averaged within each 2D map and mean and standard deviations computed at each angle. Results demonstrated a strong acoustical anisotropy. For instance acoustic impedance increased from 1.68 ± 0.08 to 1.87 ± 0.19 MRayl between 0 and 75 deg. Similarly, the speed of sound increased from 1686 ± 90 to 1958 ± 186 m/s between 0 and 75 deg. These results demonstrate the value of QAM to investigate the anisotropy of tissue microstructure and pave the way for using it to characterize other soft tissues with complex three-dimensional fiber orientations. [Work supported in part by NIH Grants EY023966 and EY028662.]

11:00

4aBAa10. Echo-mode aberration tomography: Sound speed imaging with a single linear array. Anthony Podkowa and Michael Oelze (Beckman Inst., 1009 W. Clark St. Apt. 205, Urbana, IL 61801, tpodkow2@illinois.edu)

Tomographic sound speed imaging has previously demonstrated the capability of producing images of comparable quality to that of X-ray CT and MRI. Traditionally, such reconstructions have only been achievable in transmission mode, either using diametrically opposed linear arrays or ring arrays. This is due to the conventional wisdom that forward scatter data are necessary for reconstruction in the general case, and consequentially, such setups are typically limited to easily externalized, soft tissues such as the female breast and thus are impractical for clinical usage. Recently, it has been demonstrated that in the presence of diffuse scatterers (Jaeger, 2015), pulse-echo reconstructions of slowness (inverse sound speed, proportional to refractive index) is feasible with a conventional single conventional linear array. By correlating data acquired with steered plane wave transmissions, depth dependent maps of phase lags can be generated and subsequently used to solve a multilinear inverse problem. The resulting images allow for baseband, speckle-free characterization of the underlying medium, which is complementary to the data acquired in traditional B-mode ultrasound. In this presentation, the fundamentals of echo-mode aberration tomography will be reviewed, completely with algorithmic formulation, beamformation considerations, and current challenges in practical reconstruction.

11:15

4aBAa11. Detection and imaging of prostate cancer using acoustic radiation force impulse imaging and quantitative ultrasound. Hong Chen (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), D. Cody Morris (Dept. of Biomedical Eng., Duke Univ., Durham, New York), Thomas Polascik (Dept. of Surgery, Duke Univ. Medical Ctr., Durham, NC), Wen-Chi Foo (Dept. of Pathol., Duke Univ. Medical Ctr., Durham, NC), Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York City, NY), Mark L. Palmeri, Kathryn Nightingale (Dept. of Biomedical Eng., Duke Univ., Durham, NC), and Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org)

Transrectal ultrasound (TRUS) imaging is routinely performed to guide core-needle biopsies for the definitive diagnosis of prostate cancer (PCa). Needles are rigidly attached to the transducer and randomly sample the prostate yielding a high rate of false negative determinations. This study investigated the combined use of acoustic radiation force impulse imaging (ARFI) and quantitative ultrasound (QUS) to detect cancerous lesions in the prostate during TRUS imaging. Three-dimensional (3D) RF data from 12 PCA patients scheduled for radical prostatectomy were acquired over the full gland volume during ARFI data acquisition and used to obtain effective

scatterer size (ESS), effective acoustic concentration (ESC), nakagami shape (μ), and nakagami scale (Ω) parameters. In each three-dimensional dataset, healthy and cancerous regions were obtained by manual segmentation using whole-mount histology slides. Linear discriminant and ROC methods were used to quantify the performance of ARFI displacements and QUS estimates at detecting PCa. Results for ARFI displacement and μ alone yielded an area under the ROC (AUC) curve of 0.84 and 0.69, respectively. The AUC value increased to 0.86 when μ and ARFI displacement were linearly combined. These results suggest that QUS and ARFI methods are sensitive to tissue properties affected by PCA. The proposed methods pave the way for novel real-time imaging of PCA during TRUS imaging. [Work supported in part by NIH Grants EB026233 and CA142824 and DOD PRCP Grant W81XWH-16-1-0653.]

11:30

4aBAa12. An Artificial Neural Network (ANN) approach to extract micro-architectural properties of cortical bone using ultrasound attenuation: A numerical study. Kaustav Mohanty, Omid Yousefian, Yasamin Karbalaeisadegh, Micah Ulrich, and Marie M. Muller (Dept. of Mech. and Aerosp. Eng., College of Eng., North Carolina State Univ., 3147 B, 911 Oval Dr., EB-3, Raleigh, NC 27606, kmohant@ncsu.edu)

The goal of this study is to estimate the porosity parameters including pore diameter, pore density, and porosity of cortical bone from ultrasound attenuation measurements using an artificial neural network (ANN). Two-dimensional (2D) finite-difference time-domain simulations are conducted to calculate the frequency-dependent attenuation in the range of 1–8 MHz in mono-disperse structures (constant pore size) with a pore diameter and density ranging from 20 to 120 μm and 3–16 pore/mm^2, respectively. Furthermore, poly-disperse structures (non-uniform pore distribution) are obtained from high resolution CT scans of human cortical bone and 2D numerical simulations are carried out in the same frequency range as for the mono-disperse cases. Then, a regression problem is formulated with the ultrasonic attenuation at different frequencies acting as the feature vectors and the output being set as the porosity parameters. Our dataset consists of 330 structures for the mono-disperse model and 668 structures for the poly-disperse model. ANN-based (3 hidden layers with 806 trainable weights) parameter prediction method achieves accuracies as high as 96% for pore size, 97% for porosity, and 78% for pore density for the poly-disperse model. This work demonstrates the potential of combining ultrasound methods to deep neural networks to quantify cortical bone parameters with high accuracies.

11:45

4aBAa13. Inferring elastic moduli of drops in acoustic fields. Jesse Batson, Rebekah Davis, and R. Glynn Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jbats@bu.edu)

Acoustically levitated drops serve as non-contact mini-laboratories from which one can infer material properties from the response of the drop to the acoustic radiation force. Oddly enough, the oscillatory problem is more well-developed than the static problem. Analysis of the static acoustic deformation of Newtonian liquid drops is well established, yielding the inference of the surface tension. But the static deformation of an elastic drop is less well studied. The present work aims to enable the inference of elastic moduli from static deformations of acoustically levitated drops. The drop will be modeled as an incompressible, linear elastic solid undergoing small axisymmetric deformations. The axisymmetric interior stress and displacement fields will be found using Love's strain potential. The traction boundary condition can be calculated using linear acoustic theory . The measured static deformation of experimentally levitated drops with known material properties (polymer and protein gels) will be compared to the predictions of the theory. Time permitting, a finite element computational model will also be employed for comparison.

THURSDAY MORNING, 16 MAY 2019 BREATHITT, 8:00 A.M. TO 11:00 A.M.

Session 4aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics I

Robert J. McGough, Cochair
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Hong Chen, Cochair
Washington University in St. Louis, 4511 Forest Park, St. Louis, MO 63108

Contributed Papers

8:00

4aBAb1. A parametric evaluation of shear wave speeds estimated with the time-of-flight approach in viscoelastic media. Luke M. Wiseman (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, wisemanl@msu.edu), Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

The time-of-flight approach estimates the shear elasticity in tissue mimicking elastography phantoms and in soft tissue. The time-of-flight approach is effective in elastic phantoms, but the time-of-flight approach tends to overestimate the shear elasticity in viscoelastic phantoms and in viscoelastic soft tissues. To characterize errors in estimated parameters for different values of the shear elasticity and the shear viscosity, three-dimensional (3D) shear wave simulations are evaluated for twelve different parameter combinations. The 3D acoustic radiation force is calculated for an L7-4 transducer using the fast nearfield method and the angular spectrum approach, and then, 3D shear wave propagation in a viscoelastic medium is simulated with Green's functions for a Kelvin-Voigt model. The time-of-flight method is then evaluated within a two-dimensional plane. The results show that the accuracy of the time-of-flight method depends on the values of the shear elasticity and the shear viscosity. In particular, the error in the estimated shear

elasticity increases as the shear viscosity increases, where the largest errors are observed when larger values of the shear viscosity are combined with smaller values of the shear elasticity. [Work supported in part by NIH Grants DK092255, EB023051, and EB012079.]

8:15

4aBAb2. Measured power law attenuation of shear waves in swine liver. Steven A. Grosz (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, groszste@msu.edu), Rebeca Pereira, Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

To characterize the attenuation and dispersion behavior of shear waves in excised swine liver samples, the complex shear modulus was measured with a Rheospectris C500+ from 10 Hz to 2000 Hz. The shear wave attenuation and shear wave speed were calculated from the complex modulus measurements. A power law fit was evaluated for the shear wave attenuation, and a power law fit with and without a constant offset was evaluated for the shear wave speed. The power law closely matches the measured shear wave attenuation over most of the frequency range evaluated, although some differences are observed in several of the samples below 400 Hz. The power law without the constant offset closely matches the measured shear wave speed above 200 Hz in all measurements, where the addition of the constant offset achieves a much closer fit in all measurements that contain discrepancies below 200 Hz. The results demonstrate that shear wave attenuation in swine liver follows a power law and that a power law with a constant offset is an effective model for the shear wave speed in swine liver. [Work supported in part by NIH Grants DK092255, EB023051, and EB012079.]

8:30

4aBAb3. Power law attenuation modeled as multiple relaxation. Sverre Holm (Informatics, Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316, Norway, sverre@ifi.uio.no)

Wave equations with non-integer order derivatives may model power law behavior in medical and sediment acoustics. As experiments only support a finite bandwidth, there is a limit to how much physical insight that can be gained from such models. Other ways to model a power law are with a fractional heat law, hierarchical ladder models for polymer chains, and the non-Newtonian rheology of grain shearing. Multiple relaxation processes may be motivated by a hierarchy of substructures at different scales. It is also inherent in soft glassy materials, such as cells, with disordering and metastability. Even the Biot model with contact squirt flow and shear drag (BICSQS) may be interpreted as a multiple relaxation model. A weighted sum of relaxation processes will approximate a power law over a limited band, and an even distribution of relaxation frequencies on a logarithmic frequency axis, and with equal relaxation strengths, will give a power law attenuation with unit power, y = 1. This can be generalized to other power laws if the contribution from each relaxation process varies in proportion to the relaxation frequency to the power of y - 1. This scale-invariant distribution may hint at some fractal medium properties.

8:45

4aBAb4. Approximate analytical time-domain Green's functions for space-fractional wave equations. Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), James F. Kelly (Dept. of Statistics and Probability, Michigan State Univ., East Lansing, MI), and Madison Carriere (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, carrie51@msu.edu)

Approximate and exact time-domain Green's functions are available for time-fractional wave equations that describe power law attenuation in soft tissue, where each expression contains a stable probability distribution function. Previous work has also demonstrated that the exact time-domain Green's functions for time-fractional and space-fractional wave equations that describe power law attenuation are similar. Approximate analytical time-domain Green's functions have recently been derived for the Chen-Holm and Treeby-Cox space-fractional wave equations, where the approximate time-domain Green's function for the Chen-Holm wave equation contains a symmetric stable probability distribution function and the approximate time-domain Green's function for the Treeby-Cox wave equation contains a maximally skewed stable probability distribution function. Comparisons between the exact numerical and approximate analytical expressions for these time-domain Green's functions are evaluated for published values of the power law exponent and attenuation constant for breast and for liver. The results for both breast and liver converge very close to the source, and similar performance is observed in time-domain Green's functions computed for linear with frequency attenuation. Despite minor differences in the arguments, the approximate analytical time-domain Green's functions derived for dispersive time-fractional and space-fractional wave equations are also quite similar. [Work supported in part by NIH Grants EB023051 and EB012079.]

9:00

4aBAb5. Validity of Independent Scattering Approximation (ISA) to measure ultrasonic attenuation in porous structures with mono-disperse random pore distribution. Omid Yousefian, Yasamin Karbalaeisadegh, and Marie M. Muller (North Carolina State Univ., 2704 Brigadoon Dr., Apt A, Raleigh, NC 27606, ykarbal@ncsu.edu)

The goal of this study is to assess the validity of the Independent Scattering Approximation (ISA) for predicting ultrasonic attenuation in structures mimicking simplified geometries of cortical bone. Finite Difference Time Domain (FDTD) methods were used to assess the ultrasound attenuation in porous media with a monodisperse distribution of pores, with pore diameters, density, and frequency in the range of $\phi = 40$–120μm, 3–16 pore/mm^2, and 1–8 MHz, respectively. The attenuation values obtained from the FDTD simulations were compared to attenuation values predicted by the ISA. The results indicate that the ISA reliably predicts the attenuation for $k\phi<1$ and $\phi \in [100,120]$ μm, with less than 15% error. The error increases up to 26% as ϕ decreases. The reason that ISA fails to predict accurate values for lower ϕ is investigated through the quantification of multiple scattering (MS). This is done by MS assessment in which the effect of multiple versus single scattering (SS) is compared by measuring the backscattered signals on a simulated linear array transducer. <!– Give a bit more details on how you compared the MS and SS –>The results revealed that MS is dominant at $\phi = 120$, but that SS is dominant for $\phi = 60$ μm. Assuming that the attenuation is a function of $k\phi$, the ISA is modified to test its applicability where single scattering is dominant. The results using the modified ISA showed that it can predict the attenuation in monodisperse porous structures for $k\phi < 1$ and $\phi \in [40$–$100]$ μm with less than 10% error.

9:15

4aBAb6. *In situ* calibration to account for transmission losses in backscatter coefficient estimation. Trong Nguyen (Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Urbana, IL 61801, tnnguyn2@illinois.edu), Alex Tam (Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Michael L. Oelze (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The backscatter coefficient (BSC) has demonstrated the ability to classify disease state and identify the response of cancer to therapy. However, estimating the BSC *in vivo* using a reference phantom technique does not account for transmission losses due to intervening layers, leading to increase in bias and variance of BSC-based estimates from one sample to the next. To account for transmission losses, an *in situ* calibration approach is proposed using a titanium sphere that is well-characterized ultrasonically, biocompatible, and embedded inside the sample. Ultrasound scattered from the sphere encounters the same transmission loss and attenuation as the investigated sample and can be used as a reference spectrum. To test the calibration procedures, phantoms were scanned with and without lossy layers on top, and BSCs were estimated using the *in situ* calibration approach and the reference phantom approach and compared. The differences of the BSCs, using the BSC from the reference phantom without a layer as baseline, were 0.16 ± 2.29 dB, -1.95 ± 2.99 dB and -10.90 ± 3.64 dB using the *in situ* calibration approach without the layer, with the layer, and using the reference phantom approach with the layer, respectively. The results indicate that an *in situ* calibration target can account for overlaying tissue losses thereby improving the robustness of BSC-based estimates.

9:30–9:45 Break

9:45

4aBAb7. Investigation into tendon histotripsy. Molly Smallcomb and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, molly.smallcomb@gmail.com)

Histotripsy uses cavitation bubble clouds or shock wave heating and millisecond boiling to fractionate soft tissues. While this modality has proven successful in debulking most soft tissues, highly collagenous tissues such as tendons have proven resistant to mechanical fractionation using histotripsy. In this study, *ex vivo* rat and bovine Achilles tendons were placed at the focus of a 1.5-MHz transducer and exposed to 1–20 ms pulses repeated at 1 Hz for 1 min over ranges of acoustic pressures up to p + = 88 MPa (peak positive), p = 20 MPa (peak negative). Simultaneous ultrasound imaging with the Verasonics® research ultrasound system and ATL L7-4 transducer monitored bubble activity, or hyperechogenicity, during the histotripsy exposure. Collected samples were stained with Hematoxylin and Eosin for histological analysis of tissue disruption. Preliminary results show hyperechogenicity within the tendon during the histotripsy exposure; however, thus far only thermal injury has been found histological. The threshold to detect hyperechogenicity in the tendon for 10-ms pulses were p + = 63 MPa, p = 19 MPa. Future work involves additional parameter testing to promote mechanical fractionation rather than thermal injury of tendons. [Work supported by Penn State College of Engineering Multidisciplinary Research Seed Grant]

10:00

4aBAb8. Toroidal intra-operative high intensity focused ultrasound transducer for treating liver metastases under ultrasound imaging guidance: Clinical results of Phase II study. David Melodelima (LabTAU - INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr), Aurelien Dupre, Yao Chen, David Perol, and Michel Rivoire (Ctr. Leon Berard, Lyon, France)

The aim of this study was to assess the feasibility, safety, and accuracy of HIFU ablation in patients with liver metastases in a prospective phase II trial. The transducer has a toroidal shape (diameter: 70 mm, radius of curvature: 70 mm) and was divided into 32 ring-shaped emitters operating at 2.5 MHz. Thirty-one patients were included. HIFU ablations were created to ablate metastases (up to 30 mm in diameter) with safety margins in all directions. The use of a toroidal transducer enables an ablation rate (10 cc min^{-1}) significantly higher than spherical transducers. Therefore, using electronic focusing of the beam, it was possible to treat all metastases with safety margin without the need to displace the device between HIFU exposures. The exposure time varied from 40 s to 370 s according to the diameter of the metastases to be treated. The dimensions of these HIFU ablations were a diameter of 48 $\pm$ 4.9 mm and a long axis of 45 $\pm$ 3.4 mm. No damage occurred to neighboring tissues. This study is the first clinical use of intra-operative HIFU in patients with liver metastases.

10:15

4aBAb9. Design of a histotripsy array for the treatment of intracerebral hemorrhage. Tyler Gerhardson, Jonathan R. Sukovich, Jonathan E. Lundt, Ning Lu (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48105, tgerhard@umich.edu), Aditya Pandey (Dept. of Neurosurgery, Univ. of Michigan, Ann Arbor, MI), Charles A. Cain, Zhen Xu, and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy is a focused ultrasound technique using short, high amplitude pulses to generate targeted cavitation. Recently, the feasibility to treat blood clots through human skulls with histotripsy has been shown *in vitro*. The purpose of this study was to evaluate acoustic parameters for developing an optimized array design for the treatment of intracerebral hemorrhage (ICH) with histotripsy. The main performance criteria were to achieve a large electronic focal steering range ($\geq$ 20 mm) and an ability to correct aberration through the skull. A hemispherical aperture of 150 mm radius was considered with modular elements to allow for arbitrary insertion of a catheter hydrophone to perform aberration correction. The attenuation at discrete frequencies from 250 kHz–2 MHz was measured through excised human skulls (n = 7) along with effects of incidence angle and aberration to optimize the operating frequency. Different piezoelectric materials were tested to optimize the peak-output, transduction efficiency, and durability at high PRF. One of the best performers, 17 mm square PZ36 material at 700 kHz, was found to produce at least 1.5 MPa free field at PRFs of 1 kHz. Finally, simulation showed that an optimally packed array configuration using 360 modules should be able to achieve an effective steering range of at least $\pm$20 mm through the skull.

10:30

4aBAb10. Focused ultrasound-mediated microbubble destruction for glioblastoma treatment. Lifei Zhu, Arash Nazeri, Michael Altman, Dinesh Thotala, Nima Sharifai (Washington Univ. in St. Louis, St. Louis, MO), and Hong Chen (Washington Univ. in St. Louis, 4511 Forest Park, St. Louis, MO 63108, hongchen@wustl.edu)

Glioblastoma is the most common primary brain tumor with a poor prognosis despite advances in various treatment modalities, such as radiation therapy (RT). This study compared the tumor growth inhibition effects of focused ultrasound-targeted microbubble destruction (UTMD) therapy with the RT using an orthotopic mouse glioma model. Mice were implanted with GL261 glioblastoma cells and divided into three groups: control group (no treatment); RT group (2 Gy/day, 5 days/week, 3 consecutive weeks); and UTMD group (FUS sonication in the presence of systemically injected microbubbles at the peak negative pressure of 1.5 MPa, frequency of 1.44 MHz, and 2 treatments/week for 3 consecutive weeks). Contrast-enhanced magnetic resonance imaging (MRI) was performed once every four days for measuring the tumor volume. Both UTMD and RT caused significant growth inhibition compared to the control group; however, there was no significant difference between the UTMD and RT groups. Terminal deoxynucleotidyl transferase dUTP nick end labeling (TUNEL) staining showed that the number of apoptotic tumor cells in both RT and UTMD groups were significantly higher than the control group without the difference between these two groups. This study suggests that UTMD suppressed glioblastoma tumor growth and this effect was comparable with that achieved by RT.

10:45

4aBAb11. *Ex vivo* thermal ablation monitoring using three-dimensional ultrasound echo decorrelation imaging. Elmira Ghahrahmani Z. (Biomedical Eng., Univ. of Cincinnati, 3960 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, ghahraea@mail.uc.edu), Peter D. Grimm (Elec. and Comput. Eng., Univ. of Cincinnati, Cincinnati, OH), Michael T. Cox, Kathryn J. Eary, E. G. Sunethra Dayavansha, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Echo decorrelation imaging is a method for quantitatively mapping transient heat-induced changes in pulse-echo ultrasound images. For clinical thermal ablation of liver cancer using radiofrequency or microwave ablation (RFA or MWA), real-time three-dimensional (3D) echo decorrelation imaging is necessary because the entire tumor, with typical diameter 2–5 cm, is ablated at once. We present a method for constructing 3D echo decorrelation maps during radiofrequency ablation (RFA) in *ex vivo* bovine liver using beamformed in-phase and quadrature (IQ) echo acquired from a Siemens Acuson SC2000 scanner and 4Z1c matrix array. To directly compare echo decorrelation images to the desired outcome of tissue ablation, 3D echo decorrelation images are compared to volumetric reconstructions of the thermal ablation zone, obtained from optical scans of regularly spaced tissue sections. Capability of echo decorrelation as a predictor of local ablation is assessed using receiver operator characteristic curve analysis. Similar to previous studies of two-dimensional echo decorrelation imaging, good correspondence is seen between 3D echo decorrelation images and ablated tissue histology.

Session 4aEA

Engineering Acoustics: General Topics in Engineering Acoustics: Sensors and Sources

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Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Contributed Papers

8:00

4aEA1. Developments in underwater thermophones. Nathanael K. Mayo, Christian Schumacher, John B. Blottman, and Thomas R. Howarth (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841-1708, nathanael.mayo@navy.mil)

Thermophones are acoustic projectors powered by resistive heating elements. The thermophone was invented shortly after Alexander Graham Bell's invention of the telephone, and since then, thermophone development has primarily focused around its use in air. Thermophones have historically been used as a precision source of sound for calibrating microphones and have been unable to break out of this limiting role largely due to device inefficiencies. The recent implementation of nanomaterials, which greatly improve the overall electric to acoustic conversion efficiencies, has brought about a resurgence of interest surrounding these devices; however, practical applications remain elusive. One of the most likely fits for this unique technology is as an underwater projector. Recent calibrated acoustic testing on various designs of encapsulated underwater thermophones reinforces our notional assumptions of the electro-thermo-acoustic transduction process as well as the general effect of various device parameters on acoustic performance. While efficiency is still a major concern, thermophones possess many other desirable features such as their low cost, wide bandwidth, and ability to produce low frequencies in a compact package.

8:15

4aEA2. Focal power efficiencies achieved with aluminum Fresnel and fraxicon low-profile lenses. Robert L. Lirette (Phys., Univ. of Mississippi, 2400 Anderson Rd., Apt. 4, Oxford, MS 38655, rlirette@go.olemiss.edu) and Joel Mobley (Phys., Univ. of MS, University, MS)

Fresnel and fraxicon phase plate lenses modify incident acoustic radiation by a series of phasing steps. A Fresnel lens is designed to approximate a spherical lens, and a fraxicon is designed to approximate an axicon generating a non-diffracting Bessel beam out to a depth of focus. For this study, two lenses were manufactured out of aluminum, a Fresnel with a focal point of 5 cm and a fraxicon with a DOF of 10 cm. Each lens was bonded to a 1.2-MHz piezoelectric driver with epoxy and the assemblies mounted in PVC housings for use under water. The field from each was measured by a hydrophone scanned in the transverse planes. Using an acoustic force balance, the electromechanical conversion efficiency was measured to be roughly 50% for each. The focal power fraction, defined as the ratio of the power within the -6 dB boundary of the focus to the total emitted power, was determined for both transducers. The results are compared to numerical simulations and previously published results.

8:30

4aEA3. Computational viscothermal acoustic study of micro-electro-mechanical systems (MEMS) perforated plates. Vahid Naderyan, Richard Raspet, Craig Hickey, and Mohammad Mohammadi (National Ctr. for Physical Acoust., Univ. of Mississippi, NCPA, 1 Coliseum Dr., University, MS 38677, vnaderya@go.olemiss.edu)

Micro-perforated plates (MPP) are widely used as sound absorption materials in many noise control applications. Acoustic properties of the MPPs have been theoretically and experimentally studied for many years. The results of these studies are often used in the studies of MEMS devices with perforated plates. However, there exist differences in the physical dimensions of MPPs and MEMS perforated plates. The typical MPP perforation radius is in the range of 1 mm to 1 cm. For these dimensions and audio frequencies, the shear wave-number is much larger than 1. The dimensionless shear wave-number, which is an unsteady Reynolds number, is a measure for the ratio between inertial and viscous effects. Hence for typical MMPs, the inertial effects are dominant. However, the typical hole radius in the MEMS perforated plates is below 20 μm corresponding to sub-unit shear wave-numbers. Therefore, in MEMS perforated plates, the viscous effects are the dominant part of the impedance. In addition, typical MPPs have low porosities on the order of 1%, whereas typical MEMS perforated plates have high porosities in the range of 25% to 75%. In this work, viscous and thermal losses and also the end effects of the MEMS perforated plates are studied using the finite element method.

8:45

4aEA4. Miniature fiber optic acoustic pressure sensors with air-backed graphene diaphragms. Qian Dong and Haijun Liu (Mech. Eng., Temple Univ., 1947 N. 12th St., Philadelphia, PA 191226018, haijun@temple.edu)

Graphene has been known to possess exceptional mechanical properties, including its extremely high Young's modulus and atomic-layer thickness. Although there are several reported fiber optic pressure sensors using a graphene film, a key question that is not well understood is how the suspended graphene film interacts with the backing air cavity and affects sensor performance. Based on our previously analytical model, we will show that sensor performance suffers due to the significantly reduced mechanical sensitivity by the backing cavity. To remedy this limitation, we will, through experimental and numerical methods, investigate two approaches to enhance the sensitivity of fiber optic acoustic pressure sensors using the graphene film. First, a graphene-silver composite diaphragm is used to enhance the optical sensitivity by increasing the reflectivity. Compared with a sensor

with pure graphene diaphragm, a graphene-silver composite can enhance the sensitivity by three-fold, while the mechanical sensitivity is largely unchanged. Second, a fiber optic sensor is developed with enlarged backing air volume through the gap between an optical fiber and a silica capillary tube. Experimental results show that the mechanical sensitivity is increased by 10xfrom the case where the gap side space is filled.

9:00

4aEA5. Detection of remotely piloted aircraft using bio-acoustic techniques. Jian Fang, Michael Driscoll, Russell Brinkworth, and Anthony Finn (Defence & Systems Inst., Univ. of South Australia, Bldg. W, Mawson Lakes, SA 5095, Australia, anthony.finn@unisa.edu.au)

This paper describes a biologically inspired approach for acoustically detecting and tracking small remotely piloted aircraft based on processing found in the insect visual system. Previous work has shown the insect visual system is excellent at enhancing and isolating signals in complex and noisy visual scenes. By constructing spectrograms of audio signals, we essentially converted audio data into images, which could then be processed in the same way as visual data sets. Traditional time-frequency analysis was used to characterise the signatures of remotely piloted aircraft observed by multiple sets of small microphone arrays located on the ground. A model based on multiple layers of non-linear dynamic adaptive components measured from responses of insect visual neurons was then applied to the observed spectrograms to enhance the related acoustic harmonics and suppress the unrelated noise. The result was crisp low-amplitude signal detection and classification of these difficult target sets. In contrast to traditional systems that operate uniformly across the entire spectrum—attempting to capture the world as faithfully as possible—the bio-inspired processing uses multiple time scales and operates independently on each time-frequency cell ("pixel"). This reveals unseen harmonics otherwise hidden by noise, thereby extending the maximum range at which even slow-moving, low amplitude targets are detected and tracked.

9:15

4aEA6. A bio-inspired sound source localization sensor with internal coupling. Qian Dong and Haijun Liu (Mech. Eng., Temple Univ., 1947 N. 12th St., Philadelphia, PA 191226018, haijun@temple.edu)

The mechanism of using internal coupling to enhance directional hearing has been found in various animals across multiple length scales, including crickets, lizards, frogs, birds, and alligators. For each eardrum, the acoustic stimuli impinge not only on the front side but also on the opposing side via the connecting cavity. The combination of these two stimuli renders a much higher directional sensitivity than the case with two uncoupled independent receivers. Inspired by this mechanism found in Nature, here we present a bio-inspired sound source localization sensor which consists of two pre-tensioned membranes on a three-dimensional printed housing. The vibration of the two membranes is detected by a low coherence fiber optic interferometric system. The experimental results from this prototype will be demonstrated to validate the feasibility of developing miniature bio-inspired devices for sound source localization.

9:30

4aEA7. Multi-channel broadband receive array for downward looking sonar applications. Bryan D. Todd, Jermaine L. Kennedy, and David E. Malphurs (Naval Surface Warfare Ctr. Div., 110 Vernon Ave., Panama City Beach, FL 32407, Bryan.D.Todd@navy.mil)

The recent proliferation in interest for increased detection and classification probabilities of submerged objects in maritime environments has established a need for broadband underwater acoustic receivers. A multi-channel sonar receive array composed of 16 piezoelectric ceramic elements was designed, fabricated, and developed using rapid prototyping techniques including a combination of three-dimensional printed materials, molding, and casting techniques to support various modalities of underwater sonar sensing applications. Four sets of the receive elements, consisting of four individual elements per set, were mounted using disparate methods. In-water acoustical property characterizations over a broad operating frequency range were examined and analyzed. The aforementioned results fostered identification of key array design characteristics for super-critical grazing angle downward looking sonar (DLS) systems.

9:45–10:00 Break

10:00

4aEA8. Design of a conformal acoustic parametric array. Matthew Malone and Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, mmalo2@unh.newhaven.edu)

The acoustic parametric array exploits the nonlinearity of air to create an audible sound beam that can propagate long distances. Transmitted signals are modulated around a nominal 40 kHz carrier, creating sum and difference components as the signal propagates through air. Since attenuation is proportional to frequency squared only the low-frequency difference component remains at long distances. Current commercially available parametric arrays arrange ultrasonic transducers in a planar array to create a beam of sound that is audible at distances up to 100 m. Our goal is to create a concave parametric array to determine if the added geometrical focusing allows for tighter spatial control of the audible signal. Ultrasonic transducers were mounted on a flexible three-dimensional printed structure to create an array with a variable curvature. Simulation and experimental results are presented comparing our conformal array to two commercially available planar arrays.

10:15

4aEA9. Analysis of a passive radio frequency excited acoustic transducer. Charles Thompson (ECE, UMASS, 1 University Ave., Lowell, MA 01854, charles_thompson@uml.edu), Johnetta Jallah (Lowell HS, Lowell, MA), Grace Remillard, and Kavitha Chandra (ECE, UMASS, Lowell, MA)

In this paper, the acoustic sensitivity of passive transducers excited at radio frequency is examined. This wireless battery-free sensing platform derives its power from an externally applied electromagnetic field generated by a radio transmitter. The audio signal is encoded in the backscattered electromagnetic field. Electro-Mechano-Acoustical analogies are developed and presented. Power generation, sound transduction, and radio frequency backscatter transmission of the audio signal are examined.

10:30

4aEA10. Acoustic radiation characteristics improvement according to the shape change of flat-plate display exciter speaker. Hyung Woo Park (IT, Soongsil Univ., Seoul, South Korea), SungTae Lee, and Kwanho Park (LG Display, 245 LG-ro, Paju-si, Paju, South Korea, owenlee@lgdisplay.com)

For human, sound and video are evolving as the information transmission method. People are quick and easy to understand when sound and video are transmitted at the same time. In previous studies, we introduced a study to increase the quality of sound by equalizing the position of sound and sound in a flat panel display such as OLED (organic light emitting diode) TV. In that, we have implemented a sound field by separating two or more channels by vibrating the display panel of an OLED display. In this study, we investigated the effect of the variation of the shape of the pole piece on the radiation characteristics of the exciter speaker. In dynamic speakers, a pole piece guides the voice coil. However, in the exciter, the Pole Pieces excite the diaphragm directly. Initially, due to the limitations of exciter manufacturing technology, two exciters were placed separately. The first improvement was arranged on the same axis, and the sound was implemented by twin structure. However, because of the use of two pole pieces, the subtle phase difference of the two influenced the radiation pattern. In this study, the elliptical pole piece was introduced to improve the radiation characteristics in the transverse axis direction. The use of a single pole piece not only improves acoustical characteristics in the longer radius direction; furthermore, it is confirmed that the sound quality is improved by reducing manufacturing and driving errors even in the radial direction.

10:45

4aEA11. On the downward direction of the flat panel display speaker. Hyung Woo Park (Commun. Eng., Soongsil Univ., 520 Computing Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul 06978, South Korea, pphw@ssu.ac.kr), SungTae Lee, Kwanho Park (Commun. Eng., Soongsil Univ., Paju, South Korea), and Myungjin Bae (Commun. Engeering, Soongsil Univ., Seoul, South Korea)

Information display devices are being advancement by ICT development. Particularly, besides the image quality and appearance design of the information display device, the development of the accompanying elements such as the sound quality is also progressed. Conventional information display devices such as LCD and LED TV have focused on the pixel configuration and color implementation and developed to such an extent that the human eye can not follow it, and the image quality is as good as the actual view in a proper field of view. However, in the case of sound, it has occurred at various alternative positions with the limitation that it cannot penetrate the hard screen. In the case of a flat-screen TV, however, the position of the sound was properly configured by reproducing the sound directly from the left and right sides of the screen. However, focusing on the image quality and design elements, it was hidden above. With several experimental factors, we were able to reproduce a lot of sound from the bottom speaker. This is disadvantageous in that it cannot hear the reproduced sound directly, hears mainly the reflected waves of the space below the space where the information display device is located, and hears different sounds depending on the characteristics of the reflection surface. In this study, we introduce a technique to make a sound with a direct screen that complements these shortcomings.

THURSDAY MORNING, 16 MAY 2019 — SEGELL, 8:50 A.M. TO 12:00 NOON

Session 4aNS

Noise and Education in Acoustics: Increasing Noise Awareness in Society

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Chair's Introduction—8:50

Invited Papers

8:55

4aNS1. The international Noise Awareness Day in Germany. Brigitte Schulte-Fortkamp (TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

The International Noise Awareness Day in Germany was organized first time in 1998 and will have on 24 April 2019 his 22nd yearly event. The sensitization in relation to the problem of noise along with the spreading of knowledge about causes and consequences of noise (both socially and health wise) is elementary constituents of the "Tag gegen Lärm" (Noise Awareness Day). Through its continuity over the past 22 years and its public acceptance, the "Tag gegen Lärm" has become an institution that has a permanent place in Germany's calendar. The "Noise Awareness Day" is aiming everyone who is interested in noise, its causes, consequences, and countering, including people affected by noise, subject-specific interested groups, and people with political responsibilities (citizens, economy, and politics). The "Noise Awareness Day" happens every year in April, always scheduled coordinated with the "International Noise Awareness Day" organized by the Center for Hearing and Communication (CHC) USA. Current work and activities will be presented.

9:15

4aNS2. International Noise Awareness Day and the Acoustical Society of America: Past, present, and future. L. Keeta Jones (Acoust. Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11787, kjones@acousticalsociety.org)

Prior to 2018, the Acoustical Society of America (ASA) did little to support the Center for Hearing and Communication's International Noise Awareness Day (INAD). Task Force 1 (TF1) members of the five-year ASA Strategic Plan determined that ASA must make a greater effort to support the INAD campaign to help raise public awareness of noise to meet our own strategic goals. For 2018, TF1 members organized and promoted activities that would not only increase noise awareness but would also encourage the public to interact with the ASA. These activities included taking sound level measurements using a mobile app, reading a Proceedings of the Meetings on Acoustics paper, watching a movie, and taking part in a live YouTube discussion with expert panelists. The success of these activities is measured in increased downloads, website traffic, followers, and subscribers. To continue ASA's involvement with future INADs once the strategic plan ends, organizing was moved to the Technical Committee on Noise. This presentation will end with a summary of ASA INAD 2019, set to take place on Wednesday, 24 April 2019, as well as an update on future INAD plans.

9:35

4aNS3. Soundprint and the ASA's International Noise Awareness Day Campaign—Results, what worked, and going forward. Gregory S. Farber (SoundPrint, P.O. Box 74, New York City, NY 10150, greg@soundprint.co)

SoundPrint, a crowdsourcing app that objectively measures noise levels of venues and described as "Yelp for Noise," partnered with the Acoustical Society of America (ASA) for International Noise Awareness Day in 2018 and 2019 to raise noise pollution and hearing health awareness. SoundPrint served as the technological tool by which the public used to "via their sound level submissions" to show the public that noise is an important public health issue. The 2018 INAD campaign's initial success led to a significant bump in the number of crowdsourced submissions to SoundPrint's database spanning numerous countries, states, and cities (figures will be presented). The campaign was supported with specific marketing content on noise pollution and hearing health that raised visibility for both INAD and ASA. The presentation will discuss data results associated with the campaign, marketing methods used to raise awareness, and subsequent steps other organizations can employ to further ASA's cause going forward.

9:55

4aNS4. Centers for Disease Control and prevention efforts to increase awareness and prevention of noise-induced hearing loss. Yulia Carroll and John Eichwald (National Ctr. for Environ. Health, Ctr. for Disease Control and Prevention (CDC), 4770 Buford Hwy., CDC Chamblee - Bldg. 102 Rm. 2128, Atlanta, GA 30341, YCarroll@cdc.gov)

For 45 years, the Centers for Disease Control and Prevention (CDC) has researched noise induced hearing loss (NIHL) in the workplace and disseminated its research to prevent occupational hearing loss. Additionally, CDC has made research and educational materials available on hearing loss in children. In 2015, CDC received inquiries from the public and medical community about NIHL in non-workplace settings. In response, CDC began efforts to raise public awareness of NIHL and awareness about how to prevent its health effects. A CDC intra-agency working group collaborated with the World Health Organization, the National Institute on Deafness and Other Communication Disorders and the Dangerous Decibels® program for the promotion of the materials including (1) MMWR Vital Signs: Noise-Induced Hearing Loss Among Adults—United States 2011–2012; (2) CDC Public Health Grand Rounds: Promoting Hearing Health Across the Lifespan; (3) World Hearing Day educational materials; and (4) MMWR: Use of Personal Hearing Protection Devices at Loud Athletic or Entertainment Events Among Adults — United States, 2018. Additionally, CDC scientists and communicators continue to leverage internal and external channels for developing materials and spreading the word about the prevention of NIHL at work, at home and in communities.

10:15–10:30 Break

10:30

4aNS5. Total hearing health: An approach for raising noise awareness in society. Christa L. Themann (Hearing Loss Prevention Team, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., MS C-27, Cincinnati, OH 45226, clt6@cdc.gov)

Noise-induced hearing loss is one of the most common work-related illnesses in the United States, and an estimated 24% of hearing difficulty among the working population is attributable to workplace exposures. However, the harmful effects of noise are evident even among non-workers. Regardless of occupation, nearly everyone will encounter hazardous noise at some point during their lifetime. NIOSH promotes Total Hearing Health, which broadens the scope of hearing loss prevention interventions to encompass all risks to hearing, both at and away from work. This presentation will discuss tools for applying the Total Hearing Health approach to increasing noise awareness. These tools include (a) Apps and devices which measure noise exposure and provide information on hearing loss risk; (b) Promotional ideas to raise awareness of hearing health at worksites, classrooms, health fairs, sporting events, and other venues; and (c) Wikipedia, blogs, and social media tools for expanding the reach of hearing loss prevention messages. Increasing noise awareness requires engaging the public in an authentic and meaningful way. Examples of how NIOSH have implemented Total Hearing Health to accomplish this goal and recommendations for incorporating Total Hearing Health in your own work will be provided.

10:50

4aNS6. National occupational research agenda for hearing loss prevention. William J. Murphy (Hearing Loss Prevention Team, Ctr. for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov) and Amanda S. Azman (Pittsburgh Mining Res. Div., Ctr. for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Pittsburgh, PA)

The National Institute for Occupational Safety and Health (NIOSH) has a mandate to conduct research on occupational safety and health. The research portfolio is organized by industrial sectors and cross-sectors for illnesses and injuries that are found in all sectors. The Hearing Loss Prevention research cross sector council comprises representatives from government, labor organizations, academia, and industry representatives. The HLP council held several meetings throughout 2018 to determine research needs for occupational hearing loss prevention in the United States. Five topic areas were determined. (1) Provide input for policies and guidelines that will inform best practices for hearing loss-prevention efforts. (2) Develop effective, evidence-based education designed to improve hearing conservation program outcomes for exposed workers and management. (3) Develop, commercialize, and widely implement noise control solutions on jobsites in key industries. (4) Develop audiological tests for hearing loss prevention. (5) Improve occupational hearing loss surveillance. These topic areas will be discussed in detail to help motivate other researchers to join further our knowledge to prevent occupational hearing loss.

Contributed Papers

11:10

4aNS7. Variation of sound events that stand out in one's memory across the ages: Comparison between 2008 and 2016. Takeshi Akita and Lei Qu (Dept. of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho, Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp)

To contribute to create good soundscape, sound events that stand out in one's memory are surveyed. In the present research, the data of written questionnaire that is carried out in the coursework of acoustics is analyzed. Especially, results of 2016 are compared with that of 2008. In the questionnaire, students are instructed to write down sound events that are easily recalled at the moment, after they remember the sound events that they heard in the period from the time of awakening to the coursework. Additionally, students evaluate each sound event whether they have good or bad impression. As the result, average of recalled sound events was 10.1 per person in 2016 while it was 8.2 in 2008. Sound events are classified into three large categories at each year, and the composition of the classification is the same between 2016 and 2008. They are labeled as Sound produced by a person or people, Artificially produced sound, and Nature sound. In 2016 data, the number of artificially produced sound that has no good nor bad evaluation increases significantly in comparison to 2008. It is suggested that popularization of information technology and smartphone produces more electronic sound in the urban soundscape.

11:25

4aNS8. Public health impacts from subway noise: Case study Hong Kong. Stephany Y. Xu (Harvard Univ., Extension School, 51 Brattle St., Cambridge, MA 02138, syx440@g.harvard.com), Changyong Jiang, and Lixi Huang (Lab for AeroDynam. and Acoust., Dept. of Mech. Eng. and Zhejiang Inst. of Res. and Innovation, The Univ. of Hong Kong, Hong Kong, Hong Kong)

In cities, subway noise is often cited as a major contributor to noise pollution that impacts millions of people every day. Previous studies on this topic have shown that peak subway noise levels in some cities can be as high as 110 dB, which greatly exceeds the 70 dB level set by the World Health Organization (WHO) and EPA for safe environmental noise levels. This work aims to characterize the subway noise in Hong Kong, analyze potential source features, and make technical recommendations for consideration by government and metro companies. First, the overall noise data on all nine subway lines in the city are presented and compared with published data of other subway lines around the world. Spectra of the loudest segments are analyzed to show the effects of tunnel modes, track curvature, and other features that may play a significant role in noise radiation and reverberation. A detailed correlation study is conducted for the short-time noise level and vehicle speed. A new train speed profile that optimizes noise exposure reduction is proposed for consideration of a future auto-drive system. Finally, a study of vibroacoustic exposure by passengers is also conducted to examine the impacts beyond the audible frequency range.

11:40–12:00 Panel Discussion

THURSDAY MORNING, 16 MAY 2019 — JONES, 8:05 A.M. TO 12:00 NOON

Session 4aPA

Physical Acoustics and Signal Processing in Acoustics: Infrasound I

Roger M. Waxler, Cochair
NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Philip Blom, Cochair
Los Alamos National Laboratory, P.O. Box 1663, Los Alamos, NM 87545

Invited Papers

8:05

4aPA1. Estimating multiple bearings-of-arrival from tornadic storms using the complex Wishart distribution. William G. Frazier, Carrick L. Talmadge, Claus Hetzer, and Roger M. Waxler (NCPA, Univ. of Mississippi, 145 Hill Dr., P.O. Box 1848, University, MS 38677, wgfrazier@gmail.com)

Several array signal processing methods can be used to estimate bearings-of-arrival (BOA) in the presence of multiple infrasound sources, and their effectiveness depends upon several factors including array geometry, relative signal and noise power spectra, and noise cross-spectra. One of the most effective and computationally efficient methods is Multiple Signal Classification (MUSIC). However, MUSIC's performance not only degrades with the decreasing signal-to-noise ratio, as all methods, but also degrades as the noise model deviates from the assumption of uncorrelated, equal noise power on all channels. Uncorrelated, but *unequal*, noise power levels are a common situation with infrasound arrays, and the degradation of MUSIC's performance has been observed when estimating 2–10 Hz acoustic emissions from tornadic storms. This presentation examines the performance of formal maximum-likelihood estimation of multiple BOAs using the complex Wishart distribution as a model for the array's cross-spectral density matrix estimates. Estimation and computational performance comparisons with MUSIC are also reported.

8:25

4aPA2. Improved infrasound array processing with robust estimators. Jordan W. Bishop, David Fee, and Curt Szuberla (Wilson Alaska Tech. Ctr., Geophysical Inst., Univ. of Alaska Fairbanks, 2156 Koyukuk Dr., P.O. Box 757320, Fairbanks, AK 99709, jwbishop2@alaska.edu)

Accurate infrasound source and path characterization rely on high-quality array processing parameter estimates. Physical and statistical assumptions underlying conventional array processing techniques sometimes fail in practice due to propagation effects or station degradation. Unlike conventional least squares regression, robust regression estimators are relatively insensitive to data that deviate from the assumed planar model. We compare two such estimators, M-estimators, and least-trimmed squares (LTS), to conventional array processing methods (frequency-wavenumber beamforming, progressive multi-channel cross correlation, L1 regression, and ordinary least squares) using synthetic and real infrasound data. Synthetic testing suggests that robust estimators are resistant to timing errors and noise contamination. We also present case studies from both International Monitoring System and the Alaska Volcano Observatory infrasound data that demonstrate how these techniques have produced accurate array processing results despite an element polarity reversal, timing error due to the loss of GPS lock, and a deviation from the plane wave assumption. We also evaluate the effectiveness of these techniques to arrays with differing geometries and number of elements, and note that the examination of LTS residuals enables outlying inter-element differential times to be flagged automatically, providing a data quality tool.

8:45

4aPA3. Using acoustic waveform inversion to quantify volcanic emissions. Alexandra M. Iezzi and David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, 2156 Koyukuk Dr., UAF - Geophysical Inst., Fairbanks, AK 99775, amiezzi@alaska.edu)

Volcanic eruptions produce immense sound, particularly in the infrasound band. Acoustic waveform inversion shows promise for improved eruption characterization by providing robust estimates of erupted volume and mass. Previous inversion studies have generally assumed a simple volumetric acoustic source (monopole) that radiates sound equally in all directions. However, more complex and complete source reconstructions are possible with a combination of equivalent sources (multipole). Recent work has made progress using Finite-Difference Time-Domain modeling over high-resolution topography to obtain the full three-dimensional Green's functions. The source-time function can then be inverted for and converted to a volume and mass flow rate. We review the acoustic waveform inversion as it has been applied to volcanic eruptions and discuss current limitations and how they can be mitigated. In most cases, the simple (monopole) source mechanism is a good approximation for discrete volcanic explosions, but a small directionality (dipole) component may remain. Furthermore, the neglecting effects of topography can lead to the overestimation of both the monopole and dipole strengths. Volcano infrasound source mechanisms are also not well constrained due to infrasound sensors usually being deployed on the surface. The methods discussed here can be extended to anthropogenic explosions and monitoring efforts, potentially in near-real time.

9:05

4aPA4. Probabilistic inversion for submerged or buried source depth and strength from infrasound observations. Gil Averbuch (Dept. of GeoSci. and Eng., Delft Univ. of Technol., Graswinckelstraat 64, Delft 2613 PX, The Netherlands, g.averbuch@tudelft.nl), Roger M. Waxler (NCPA, Univ. of Mississippi, Oxford, MS), and Läslo G. Evers (R&D Dept. of Seismology and Acoust., Royal Netherlands Meteorological Inst., De Bilt, The Netherlands)

In seismology, the depth of a near surface source is hard to estimate in the absence of local stations. However, long-range infrasound propagation from an underwater or underground source is very sensitive to variations in its depth. This characteristic is employed in an infrasound based inversion for the sources depth and effective-acoustic-strength (EAS). A synthetic dataset, generated by the Fast-Field-Program (FFP), is used to investigate the accuracy of a Bayesian inversion scheme under the variations of the number of stations, source depth, and signal-to-noise ratio (SNR). SNR has proved to have the most dominant influence on the inversion precision. Results from a single station inversions with $SNR = 5$ had a standard deviation (SD) of ± 20m in depth and 10% in EAS. For $SNR = 1$, SD values increased to ± 40 m in depth and 40% in EAS. Similar results were obtained from five and ten stations inversions. This is the first attempt to extract the absolute source depth and EAS from long-range infrasound signals. Results show that infrasound may be used to accurately obtain underwater and underground source parameters.

9:25

4aPA5. Modelling ocean ambient noise with finite ocean depth and comparisons with observations. Marine De Carlo, Alexis Le Pichon (CEA/DAM/DIF, Arpajon F-91297, France, marine.decarlo@cea.fr), and Fabrice Ardhuin (Laboratoire d'Océanographie Physique et Spatiale (LOPS), Univ. Brest, CNRS, Ifremer, IRD, Brest, France)

The global International Monitoring System (IMS) network continuously detects coherent ambient infrasound noise between 0.1 and 0.5 Hz. This noise, referred to as microbaroms, is generated by the second order non-linear interaction of ocean waves, mostly during severe storms. A global and multi-year analyze of microbaroms highlights the strong influence of middle atmospheric conditions on the propagation. Various source models have been developed. Brekhovskikh *et al.* (1973) and Ardhuin and Herbers (2013) considered a source directivity effect in infinite depth ocean with radiative pressure depending on the wave elevation angle. Waxler and Gilbert (2006) and Waxler (2007) investigated the radiation of infrasound by ocean waves in finite depth ocean by monopolar sources. In this study, the combined effects of non-monopolar source and bathymetry on the radiation are addressed. Beyond theoretical issues, source modelisation and propagation through a realistic atmosphere are carried out. Comparing the predicted signals with the observed ones at all IMS stations shows good agreements for both directional and amplitude information. Building a global reference database of oceanic noise sources opens new perspectives for providing additional integrated constraints on middle atmosphere dynamics and disturbances.

9:45

4aPA6. Data-driven interpretable models of wave dynamics for infrasound monitoring. Christophe Millet (CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr) and Francois Lott (LMD, ENS, Paris, France)

Accurate and efficient models are essential to understand and predict acoustic signals propagating in the atmosphere. In many practical problems such as source localization and yield estimation, multiple models for describing the atmospheric fluctuations and infrasound are available, with varying approximation qualities. In standard practice, inferences are exercised as if the selected models had generated the observations. This approach ignores the model uncertainty, leading to biased inferences and to estimates that may be extremely sensitive to tunable parameters. This work explores a hierarchical Bayesian framework for producing interpretable models for both atmospheric gravity wave (GW) dynamics and acoustic propagation, from ground-based infrasound measurements. It is shown that there are only a few important terms that govern the GW dynamics and the interactions with infrasound. The resulting GW models can either be incorporated into global climate models to better describe the effects of GWs on the global circulation or used together with infrasound propagation models for improving inference accuracy and efficiency. This perspective, combining the resulting infrasound-driven models with sparse sensing and machine learning to monitor the atmosphere, is explored using recurring events such as the ammunition destruction explosions at Hukkakero, in northern Finland.

10:05–10:20 Break

10:20

4aPA7. Preliminary analyses of seismo-acoustic wave propagation in outdoor field-scale analog volcanic explosions. Tracianne B. Neilsen (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Robin S. Matoza, Sean Maher (Univ. of California, Santa Barbara, Santa Barbara, CA), Margaret G. McKay (Brigham Young Univ., Provo, UT), Richard Sanderson (Univ. of California, Santa Barbara, Santa Barbara, CA), Greg A. Valentine, Ingo Sonder, and Andrew G. Harp (Univ. at Buffalo, Buffalo, NY)

Shallow and subaerial volcanic processes radiate infrasound directly into the atmosphere; sampling these infrasound complement seismic data and aids with physical quantification of explosive eruption mechanisms. More advanced quantitative models of the infrasonic source and associated seismo-acoustic wave conversion and coupling have the potential to improve volcano monitoring capability. Field-scale outdoor experiments under relatively controlled conditions provide the opportunity to test, refine seismo-acoustic wave propagation and source inversion strategies, and provide a critical bridge between laboratory-scale experiments, numerical simulations, and full-scale volcano field data. We present preliminary investigations of data collected during an NSF-sponsored workshop at the University at Buffalo in July 2018. Sets of buried explosives were detonated sequentially. The explosions were recorded at 30–330 m on colocated broadband seismometers buried at 1 m, infrasound sensors, and microphones. Analyses of waveform signatures, including cross-correlation and coherence analyses, provide insights into coupling between seismic and acoustic signals over different frequency bands as a function of distance. Comparisons of the seismo-acoustic coupling for a variety of blast strengths and detonation sequences provide insights into how seismo-acoustic coupling scales with amplitude and source depth. The use of both microphones and infrasound sensors highlights the potential benefit of wideband volcano-acoustic recordings.

10:40

4aPA8. The sub-microbarom notch in acoustic wind-filter response. Thomas B. Gabrielson (Penn State Univ., P.O. Box 30, State College, PA 16804, tbg3@psu.edu)

The measurement of the frequency response of infrasound elements with spatial-averaging wind filters is often done by comparison with a reference sensor and with ambient noise as the excitation. Frequently, a notch appears in the response just below the microbarom band—a notch that is not explained by the acoustics of the wind filter. In fact, this notch is diagnostic of the spatial averaging of wind-associated turbulence. The frequency region of the notch is bounded above and below by regions in which excellent determinations of response can be made (1) below the notch under moderate- to high-wind conditions where the scale of the turbulence exceeds the scale of the wind filter rendering the wind filter ineffective and (2) in the microbarom region where the acoustic component is strong and coherent across the entire wind-filter aperture. Furthermore, the phase of the response is not affected in the region of the notch. Consequently, the true acoustic response can be estimated in the notch region in several ways. It would, however, be a mistake to ignore the information about the effectiveness of the wind filter that, in effect, creates the notch.

Contributed Papers

11:00

4aPA9. Acoustic resonances of atmospheric oscillations. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu), Nikolay A. Zabotin, and Liudmila Zabotina (Elec., Comput. and Energy Eng. Dept., Univ. of Colorado, Boulder, CO)

Atmosphere responds in a resonant way to broad-band excitation by earthquakes, volcano eruptions, and convective storms. Energetic oscillations, known as acoustic resonances, occur at frequencies of 3.5–4.5 MHz and involve infrasound propagation between lower thermosphere and either the ocean or the solid earth. Several approaches have been proposed in the literature to determine the conditions of the acoustic resonances occurrence, predict their frequencies, and relate the frequencies to thermal structure of the atmosphere. This paper presents an asymptotic theory of atmospheric resonances. Contributions to the resonance condition of the Berry phase of infrasonic waves as well as phase shifts at turning points and at reflection from the ground surface are discussed. Unlike low and middle latitudes, acoustic resonances are predicted to be a seasonal phenomenon in polar regions. Excitation of atmospheric resonances by plane-wave vertical displacements of the ground surface and by finite sources is considered. Asymptotic predictions are compared to results of numerical simulations. Infrasound tunneling between turning points via evanescent waves is shown to play a critical role in ionospheric manifestations of the acoustic resonances. [Work supported in part by NSF.]

11:15

4aPA10. Initial characterization of infrasound sources using the phase and amplitude gradient estimator method for acoustic intensity. Francisco J. Irarrazabal, Mylan R. Cook, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, firarrazabal@gmail.com), Pauline Nelson (Dept. of Phys., Brigham Young University-Idaho, Rexburg, ID), Daniel J. Novakovich, and Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The phase and amplitude gradient estimator (PAGE) method for vector acoustic intensity [Thomas *et al*., *J. Acoust. Soc. Am.* **137**, 3366–3376 (2015)] has been used previously to improve source characterization over broad frequency ranges. This paper describes initial applications of the PAGE method to the infrasound region for outdoor sources using multi-microphone probes. Measurement challenges include wind noise, which reduces signal-to-noise ratio and coherence, low-frequency phase and amplitude mismatch between microphones, and determining an appropriate microphone spacing to maximize bandwidth. Analysis challenges include phase unwrapping above the spatial Nyquist frequency, source statistical stationarity, and balancing frequency resolution with averaging across finite record lengths. This paper discusses how these challenges are being addressed for specific sources of infrasound, namely, wind turbines and large rocket motors. [Work supported by NSF.]

11:30

4aPA11. The transition of gravity wave or mesoscale flow to boundary layer turbulence and implications for infrasound propagation conditions. Jelle D. Assink (Seismology and Acoust., Royal Netherlands Meteorological Inst. (KNMI), P.O. Box 201, De Bilt 3730 AE, The Netherlands, assink@knmi.nl) and Gregory W. Lyons (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Knowledge of the atmospheric boundary layer (ABL) is important for weather and climate forecasting as well as infrasound detection. In spite of many advances, the ABL is still not represented realistically in weather models, which leads to uncertainties in weather predictions. At the same time, the influence of the ABL on infrasound detectability is not well understood, which potentially limits the capability of the technique. In this work, a network of anemometers, microbarometers, and ceilometers is used to characterize gravity wave activity and turbulence in the ABL. The sensors are co-located at the Cabauw Experimental Site for Atmospheric Research (CESAR). The sensor network allows for the estimation of velocity and pressure spectra, as well as boundary-layer structure and mixing height. The goal of this work is to study the transition from gravity wave or mesoscale flow to boundary layer turbulence and to consider implications for infrasound propagation conditions. Here, an analysis of spectra under convective and stable conditions is presented with respect to both gravity wave and boundary-layer turbulence theory. Observed distinctions in the velocity and pressure spectral transitions will be discussed, including large-scale turbulence and the mesoscale spectral gap.

11:45

4aPA12. The effects of spatial and temporal frequency of meteorological data sampling on accurate prediction of infrasound propagation. Ross E. Alter (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, ross.e.alter@usace.army.mil), Michelle E. Swearingen (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), Mihan H. McKenna (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS), Christopher Simpson (GeoTech. and Structures Lab., U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS), and Brian G. Quinn (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Accurately characterizing the environment is essential for robust predictions of infrasound propagation. In particular, meteorological data often exhibit large spatiotemporal variations; thus, accurate characterization can be challenging given the limitations of current measurement techniques. Though numerical weather models can provide realistic representations of the meteorological environment, the spatial and temporal sampling frequencies needed for accurate prediction of infrasound propagation are currently unknown. To address this issue, various simulations were conducted with the Weather Research and Forecasting (WRF) meteorological model, and the output was incorporated into a wide-angle parabolic equation method (PE) model to predict local infrasound propagation between 1 and 20 Hz. The sensitivity of infrasound propagation to spatial (1–15 km) and temporal ($\leq$1 min) sampling frequency within the WRF model was then calculated. Furthermore, the results of these simulations were compared to calibrated field measurements of meteorological and infrasound data at ranges up to 15 km to identify which sampling frequencies and locations along the propagation path provided the most accurate propagation results. Next steps for applying these results to future infrasound simulations will also be discussed, particularly regarding the optimization of horizontal and vertical meteorological sampling. Distribution Statement A: Approved for public release; Distribution is unlimited.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Spatial Hearing, Complex Acoustic Scenes, and Clinical Devices (Poster Session)

Z. Ellen Peng, Cochair
Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53705

Axel Ahrens, Cochair
Department of Electrical Engineering, Hearing Systems group, Technical University of Denmark, Ørsteds Plads, Building 352, Kgs. Lyngby 2800, Denmark

All posters will be on display from 8:00 a.m. to 12:00 noon. To give contributors in this session 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

4aPP1. Synergy of spectral and spatial segregation cues in simulated cocktail party listening. Brianna Rodriguez (Dept. of Commun. Sci. and Disord., Univ. of South Florida-Tampa, Tampa, FL 33620, bcrodriguez@mail.usf.edu), Jungmee Lee (Commun. Sci. and Disord., Univ. of South Florida - Tampa, Madison, Wisconsin), and Robert Lutfi (Commun. Sci. and Disord., Univ. of South Florida - Tampa, Tampa, FL)

An approach is borrowed from Measurement Theory [Krantz *et al.*, *Foundations of Measurement* (1971), Vol. 1] to evaluate the interaction of spectral and spatial cues in the segregation of talkers in simulated cocktail-party listening. The goal is to determine whether mathematical transformations exist whereby the combined effect of cues can be additively related to their individual effects. On each trial, the listener judged whether an interleaved sequence of 4 vowel triplets (heard over headphones) was spoken by the same $BBB_{BBB...}$ or different $ABA_{ABA...}$ talkers. The talkers had nominally different fundamental frequencies and spoke from nominally different locations (simulated using Kemar HRTFs). Natural variation in these cues was simulated by adding a small, random perturbation to the nominal values independently for each vowel on each trial. Psychometric functions (PFs) relating *d'* performance to the difference in nominal values were obtained for the cues presented individually and in combination. The results revealed a synergistic interaction of cues wherein the PFs for cues presented in combination exceeded the simple vector sum of the PFs for the cues presented individually. The results are discussed in terms of their implications for possible emergent properties of cues affecting performance in simulated cocktail-party listening. [Work supported by NIDCD R01-DC001262].

4aPP2. Context-dependent trading of binaural spatial cues in virtual reality. Travis M. Moore and G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, tmoore16@samford.edu)

A classic paradigm used to quantify the perceptual weighting of binaural spatial cues requires a listener to adjust the value of one cue, while the complementary cue is held constant. Adjustments are made until the auditory percept appears centered in the head, and the values of both cues are recorded as a *trading ratio*, most commonly in μs ITD per dB ILD. Interestingly, the existing work has shown that TRs differ according to the cue being adjusted. The current study investigated whether cue-specific adaptation-which might arise due to the continuous, alternating presentation of signals during adjustment tasks-could account for this poorly understood phenomenon. Three experiments measured TRs via adjustment and via lateralization of single targets in virtual reality (VR). Targets were 500 Hz pure tones preceded by silence or by adapting trains that held one of the cues constant. VR removed visual anchors and provided an intuitive response technique during lateralization. The pattern of results suggests that adaptation can account for cue-dependent TRs. An adaptation-based theory states that the ITD contributes most to the TR during adjustment, and adjusting the ILD results in a TR reflects contributions from both the ITD and ILD. [Work supported by NIH R01 DC016643.]

4aPP3. Perceptual weighting of elevation localization cues across frequency. Axel Ahrens (Facebook Reality Labs, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, aahr@elektro.dtu.dk) and Owen Brimijoin (Facebook Reality Labs, Redmond, WA)

Spectral cues are thought to be of particular importance in the perception of the elevation of a sound source. While some work has been done on demonstrating the importance of individual frequency bands, the relative importance of bands across a wide range of frequencies has not been firmly established. To estimate this, we built a broadband signal consisting of seven 1-ERB-wide noise bands that could each be assigned to a different elevation. The frequency range was either from 1 to 16 kHz with 3-ERB-wide spectral gaps or a higher-resolution range from 3 to 12 kHz with 1-ERB-wide spectral gaps. On each trial, each frequency band was independently convolved with a randomly chosen personalized head-related transfer function from one of seven elevations (±60 deg, 15 deg steps). In a 1-interval, 2-alternative forced choice task, listeners were asked to judge whether the sound was perceived above or below a reference stimulus presented on the horizontal plane. Two azimuth angles at -15 deg and -45 deg were considered. Perceptual weights for each frequency band were then calculated using a regression analysis method. Results showed that listeners tended to weight the 6.5 kHz band the highest for both azimuth directions and frequency resolution conditions.

4aPP4. The effect of reverberation on listening effort. Yi Shen, Yuan He, Kimberly G. Skinner, and Donghyeon Yun (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

The current study investigates whether long reverberation increases listening effort during speech recognition. Listening effort during word recognition in multi-talker babble noise was assessed with or without high levels of reverberation. A dual-task paradigm was adopted, in which the primary task was word recognition in noise at individually selected signal-to-noise

ratio (SNR) that yielded an average performance level of 50% correct, and the secondary task was a visual-tracking task with individually adjusted difficulty level to yield an average performance level of 85% correct. In each 30-s trial, seven monosyllabic words were presented sequentially at a rate of four seconds per word. Young normal-hearing listeners were instructed to verbally repeat each word while performing the secondary task. In the reverberant condition, a cascade of all-pass filters was used to achieve a reverberation time of 1 s without altering the original spectrum of the speech. For the primary task in isolation, the reverberant condition required 10 dB or more in SNR to achieve the 50% target performance level. When the listeners performed the two tasks simultaneously, no consistent adverse effect of reverberation was found on the performance of the primary or secondary task compared to the no-reverberation condition.

4aPP5. Perceptual squelch of room effect in listening to speech. Aimee Shore (Phys. and Astronomy, Michigan State Univ., 567 Wilson Rd., Biomedical Physical Sci. Bldg., MSU, East Lansing, MI 48824, shoreaim@msu.edu), Brad Rakerd (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

Room effect squelch—the auditory system's ability to suppress reverberation and coloration—has historically been entirely attributed to binaural listening. An alternative hypothesis is that a listener's own head-related transfer functions (HRTFs) are necessary for maximum squelch. Two perceptual experiments were conducted to investigate the role of individualized HRTFs. The first experiment used binaural synthesis over headphones to deliver speech stimuli to listeners. Head-related impulse responses were measured in a test room and convolved with anechoic female speech. Headphone presentation of convolved stimuli was diotic or binaural, and listeners rated the amount of perceived room effect in each stimulus. Regression analyses indicated that listeners perceived less room effect in binaural listening mode, but ratings were similar for individualized and nonindividualized HRTF conditions. Because it was thought that headphone presentation did not adequately convey HRTFs, a second experiment was conducted using loudspeakers. Transaural synthesis was used to present individualized and nonindividualized speech stimuli to listeners. Analyses indicated that listener ratings of perceived room effect were often, but not always, lower when listening to own-ear conditions. We conclude that there is limited support for the hypothesis that listeners experience maximum squelch when listening with their own ears.

4aPP6. Listening while balancing: Dual-task costs in speech vs. noise maskers. Karen S. Helfer, Richard L. Freyman (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu), Richard Van Emmerik, Jacob Banks (Kinesiology, Univ. of Massachusetts Amherst, Amherst, MA), Michael Clauss, and Lincoln Dunn (Commun. Disord., Univ. of Massachusetts Amherst, Amherst, MA)

Many middle-aged adults report that listening is effortful in adverse communication situations. One means of quantifying listening effort is by measuring dual-task costs. The present study examined the influence of early aging on dual-task costs using a technique which required participants (younger and middle-aged adults) to complete a postural control task while listening to speech. For the postural control task, participants stood on a force platform and had to maintain their center of pressure within a prescribed area (denoted using real-time visual feedback). Two speech perception tasks were used, each presented with two types of maskers (same-sex two-talker speech masker and steady-state speech-shaped noise): repeating back low-predictability sentences, and listening to Connected Speech Test passages and then answering content questions based on each passage. This presentation will describe data analyses designed to uncover how listener age group and masker type influenced listening effort as measured by dual-task costs. [Work supported by NIDCD 012057.]

4aPP7. Informational masking of speech analogues by intelligible and non-intelligible but acoustically similar interferers. Robert J. Summers and Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Psych., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk)

Informational masking of target speech is generally greater when the interfering speech is intelligible than when it is not (e.g., speech from an unfamiliar language), but the relative contributions of acoustic-phonetic and linguistic interference are often difficult to assess owing to acoustic differences between interferers (e.g., different talkers). This study used three-formant analogues (F1 + F2 + F3) of natural sentences as targets and interferers. Target formants were presented monaurally (F0 = 120.3 Hz) either alone or accompanied with the contralateral ear by interfering formants from another sentence (F0 = 151.5 Hz); a target-to-masker ratio (TMR) between ears of 0, 6, or 12 dB was used. Interferers were either intelligible or rendered non-intelligible by delaying F2 and advancing F3 by 150 ms relative to F1, a manipulation designed to minimize spectro-temporal differences between corresponding interferers. Target-sentence intelligibility (keywords correct) was 67% when presented alone but fell considerably when a non-intelligible interferer was present (49%) and significantly further when the interferer was intelligible (41%). The changes in TMR produced neither a significant main effect nor an interaction with interferer type. The results suggest that although linguistic factors contribute to informational masking, interference with acoustic-phonetic processing of the target can explain much of the impact on intelligibility. [Work supported by ESRC.]

4aPP8. Is auditory distance perception in rooms binaural? Luna Prud'homme and Mathieu Lavandier (Laboratoire Génie Civil et Bâtiment, Univ Lyon, ENTPE, 3 rue Maurice Audin, Vaulx-en-Velin 69120, France, luna.prudhomme@entpe.fr)

The goal of this study was to determine whether auditory distance perception is binaural or monaural. Listeners performed an experiment in which they judged the distance of a sound source using headphones. Individualized and non-individualized binaural room impulse responses were measured to simulate sound sources placed between 1 and 4 m in front of the listener. The listening test was performed in the same room used for the measurements, and listeners were facing visual anchors. Different conditions tested the influence of controlling the sound level, individualizing the stimuli, and the amount of binaural information present in these stimuli. Results showed that binaural information does not seem to be necessary for auditory distance perception in rooms for naive listeners. However, its absence can alter externalization of sounds, which could prevent listeners from judging distance via headphones when it creates a mismatch between auditory and visual information. The variation of the sound level was a preponderant cue used by the listeners. Its absence or artificial variation greatly altered distance judgments for naive listeners.

4aPP9. Investigating the role of temporal fine structure in everyday hearing. Agudemu Borjigan and Hari M. Bharadwaj (Biomedical Eng., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, aagudemu@purdue.edu)

Human listeners can derive substantial masking release when there are discrepancies in pitch or spatial location between the target and masking sounds. While the temporal fine-structure (TFS) in low-frequency sounds can convey information about both pitch and location, a nuanced debate exists in the literature about the role of these TFS cues in masking release. The long-term goal of the present study is to leverage individual differences to understand the role of TFS in everyday hearing. As a first step, we sought to measure individual TFS sensitivity using monaural frequency modulation (FM) and binaural interaural time difference (ITD) detection tasks.

Preliminary data show large individual differences in these measures. Moreover, individual differences in ITD sensitivity were correlated with monaural FM sensitivity suggesting that monaural TFS coding can be a primary bottleneck determining binaural sensitivity. Alternately, both FM and ITD sensitivity variations could be reflecting common non-sensory factors (e.g., attention). To disambiguate between these hypotheses, we designed two passive EEG metrics of TFS coding. Follow-up experiments will compare individual differences in these perceptual and EEG measures to each other, and to speech-in-noise perception in complex environments.

4aPP10. A method for conversational signal-to-noise ratio estimation in real-world sound scenarios. Naim Mansour (Hearing Systems, Dept. of Health Technol., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kongens Lyngby 2800, Denmark, naiman@elektro.dtu.dk), Márton Marschall, Tobias May (Hearing Systems, Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), Adam Westermann (Widex A/S, Lynge, Denmark), and Torsten Dau (Hearing Systems, Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

The analysis of conversational signal-to-noise ratios (SNRs) measured in real-world scenarios can provide vital insight into people's communicative strategies and difficulties and guide development of hearing devices. However, measuring SNRs accurately and realistically is challenging in typical recording conditions, where only a mixture of sound sources is captured. This study introduces a novel method for realistic *in situ* SNR estimation, where the speech signal of a person in natural conversation is captured by a cheek-mounted microphone, adjusted for free-field conditions, and convolved with a measured impulse response to estimate the clean speech component at the receiver. A microphone near the receiver computes the noise-only component by applying a voice activity detector. Obtained SNR values are analyzed using *in situ* recordings of a real-world workspace meeting. It is shown that the temporal resolution is increased, and fluctuations in the speech level are more accurately tracked compared to a typical spectral-subtraction-derived method. The application of the proposed SNR estimation method may be valuable for compensation procedures in hearing instruments that take conversational dynamics into account.

4aPP11. Disentangling the contribution of head shadow, loudness summation, and binaural unmasking to spatial release from masking in children. Z. Ellen Peng and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53711, zpeng49@wisc.edu)

Segregating target speech from noise is crucial for children's ability to communicate effectively in everyday environments. Past research clearly shows that when target sources are spatially separated from maskers, compared with target-masker being co-located, children as young as 2–3 years old demonstrate improved speech understanding. This effect is known as spatial release from masking (SRM). Generally, studies have used free-field or dichotic listening; hence, the contributions of head shadow, loudness summation, and binaural unmasking to SRM are unknown in children. This study aimed to quantify these factors in virtual auditory space. By varying the target-masker spatial configurations (co-located versus separated) and ear conditions (monaural versus binaural), speech understanding benefit was defined as improvement in the signal-to-noise ratio to achieve an accuracy of 50%. Results from 29 children with normal hearing (6–15 years old) show that head shadow cues are dominant in providing benefit, followed by binaural unmasking. Loudness summation, through increased intensity by listening with both ears, provided little to no benefit. No age effects were found. Results also suggest a re-balancing between cues depending on listening strategies adopted by children. For example, children who relied more on binaural unmasking received less benefit for speech understanding from head shadow. [Work supported by NIH-NIDCD.]

4aPP12. Sound source localization in two-dimensions: Rotating sources and listeners. William Yost (ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu) and M. Torben Pastore (ASU, Troy, New York)

In 1940, Wallach published the last of three articles on localizing sound sources in two dimensions: azimuth and elevation. He proposed: "Two sets of sensory data enter into the perceptual process of localization (1) the changing binaural cues and (2) the data representing the changing position of the head." Wallach explained how head motion resolves cone-of-confusion errors to support his proposal. A group of experiments in which listeners and sound sources rotated on the azimuth plane demonstrated how head motion contributes to localization in terms of both azimuth and elevation by inducing illusory perceived sound source locations. The results generally supported his proposal regarding head motion. As Wallach was not aware of aspects of current knowledge of sound source localization, such as the role of head-related-transfer function, HRTF, and the use of the resulting spectral cues, some of his conclusions turn out to be incorrect. We conducted a series of experiments similar to Wallach's to more fully examine the roles of head and sound source rotation in localizing sound sources in two-dimensional auditory space. Some of these results will be described in this presentation. [Work supported by NIDCD and Facebook Reality Labs.]

4aPP13. Audio and visual distance perception of familiar and unfamiliar objects using Wave Field Synthesis and a stereoscopic display. Sarah Richie and Jonas Braasch (Graduate Program for Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, richis@rpi.edu)

Object distance perception can be influenced both by auditory and visual cues. This work seeks to examine the influence of both perceptual domains for familiar and unfamiliar auditory and visual stimuli. For example, an alarm clock is a familiar object and a generic vibrating sphere is an unfamiliar object because the distance cannot be estimated from known dimensions. A Wave Field Synthesis (WFS) system and a stereoscopic large screen display using shutter glasses was used to create the virtual objects. Utilizing WFS allowed for sources to be placed virtually behind and in front of the speaker array. Cues were presented audio only, visual only or audio and visual simultaneously. Participants were asked for the estimated depth of the object while randomizing the above scenarios. This work expands upon a previous study [*J. Acoust. Soc. Am.* **137**, 2374] that suggested that the visual cues tend to dominate perception even when auditory cues are available. One goal of the new study is to investigate if finding holds true and if the listener is presented with more salient cues that also allow for head movements. In the previous study, the virtual environment was based on static Head-Related Transfer Functions (HRTFs).

4aPP14. Hearing impairment and reverberation preference: Results from a virtual sound space. Andrew Burleson, Kendra L. Marks, and Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, andrewburleson@u.northwestern.edu)

Reverberation is regarded as a positive component of music perception and may lead to feelings of envelopment in well-designed auditoria. While relative reverberation time preferences are clear for young, normal-hearing (YNH) listeners, previous work indicates that older, hearing-impaired listeners (OHI) show less distinct preferences for reverberation time in music. OHI listeners have degraded temporal and spatial processing abilities that impact both reverberation perception and binaural processing of auditory stimuli. Previous work has been limited to earphone presentation, precluding an individualized head-related transfer function. This experiment employed these individualized auditory cues by evaluating reverberation preference in a virtual sound room for OHI and YNH listeners. Three symphonic excerpts, spatialized to simulate orchestral performance, were presented with a range of reverberation times. Listeners selected a preferred reverberation time in a series of paired comparisons. Thresholds for interaural coherence correlation (ICC)—a binaural processing measure—were obtained. Preliminary results indicate that YNH listeners have better ICC thresholds than OHI. Concordant with previous work, YNH listeners show relative reverberation time preference at roughly 2.5 s. OHI listeners show a different preference pattern than YNH. Results to date indicate that naturalistic listening cues may play an important role in music perception for OHI listeners. [Work supported by NIH.]

4aPP15. Impact of spatial variability and masker fringe on detectability of brief signal. Michelle H. Wang, Robert H. Gilkey (Psych., Wright State Univ., 3640 Colonel Glenn Hwy., Dayton, OH 45435, wang.202@wright.edu), and Brian Simpson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Seemingly, there should be a close relationship between spatial release from masking and sound localization, but this is not always the case. For example, in binaural detection, randomizing the spatial parameters of the target or masker from trial to trial has little impact on threshold. In contrast, Simpson (Ph.D. dissertation, 2011) found that left/right localization judgments for a 60-ms target masked by a simultaneous 60-ms noise were considerably less accurate (equivalent to 10-dB reduction in SNR) when the location of the masker varied randomly from trial to trial than when the masker location was fixed. However, when a forward masker fringe was added, so that the noise was turned on 500 ms before the target, the impact of location variability was very small (about 1 dB, comparable to the detection literature). To determine if the presence of masker fringe could have limited the impact of spatial variability in previous detection experiments, the current study examines the effect of masker fringe and both target and masker spatial variability on detectability in conditions comparable to those of Simpson. The results will be compared to previous findings and models reported in the binaural detection and sound localization literature.

4aPP16. The effect of musical training on ecological cocktail party listening. Anneliese K. Schulz, Elissa Hoffman, and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, 901 S. Sixth St., Champaign, IL 61820, akschul2@illinois.edu)

A multitude of studies have investigated the phenomenon that experience, such as musical training, has an impact on listener performance in challenging auditory environments. Many studies examining speech-in-speech listening (i.e., the cocktail party problem) simulate an unnatural scenario where the target talker and maskers are all facing the listener. We analyzed participants' performance in a more realistic situation with a target talker facing the listener and co-located maskers with head orientations facing away from the listener (45 or 60 deg relative to the listener). We aimed to determine if musical training provided an advantage to our participants under these ecological conditions. Stimuli were presented over a loudspeaker to listeners in a sound treated booth. Preliminary data indicate that highly trained musicians ($N = 6$) perform better than nonmusicians ($N = 25$) in our task. Musical training may improve auditory functioning in challenging ecological listening situations. Data collection for listeners with extensive musical training is ongoing.

4aPP17. Efficacy of iPad "spatial release" application. Allison Holtz, Kelli Clark, and Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, aholtz3@students.towson.edu)

Spatial Release from Masking (SRM) is the ability to obtain better speech recognition thresholds when the maskers are spatially separated from the target. Here, we present SRM data collected using three techniques: over headphones using a virtual speaker array, using Spatial Release iPad application (https://bgc.ucr.edu/games/spatialrelease/), and loudspeaker presentation in a sound-attenuated room. For all three techniques, Coordinate Response Measure (CRM) sentences were used as the stimuli, and "Charlie" was the call sign. A progressive tracking procedure was used to estimate the Speech Recognition Thresholds (SRTs) for listeners with varying hearing thresholds. The target sentence was always presentedg at 0 deg azimuth angle whereas the maskers were colocated (0 deg) with the target or symmetrically spatially separated by ± 15 deg, ± 30 deg, or ± 45 de. Initial data analysis revealed similar SRTs for the iPad and headphone conditions and slightly poorer thresholds for the loudspeaker array condition. This was true for all spatial separations between the target and the maskers. The individual effects of age and hearing loss on spatial release from masking will be discussed. These data will aid clinicians to rapidly characterize difficulties perceived by individuals in everyday listening scenarios and to evaluate patient progress with hearing aid adjustments and aural rehabilitation over time.

4aPP18. A deep learning based segregation algorithm to increase speech intelligibility for hearing-impaired listeners in reverberant-noisy conditions. Yan Zhao, DeLiang Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH), Eric Johnson, and Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu)

Recently, deep learning based speech segregation has been shown to improve human speech intelligibility in noisy environments. However, one important factor not yet considered is room reverberation, which characterizes typical daily environments. The combination of reverberation and background noise can severely degrade speech intelligibility for hearing-impaired (HI) listeners. In the current study, a deep learning based time-frequency masking algorithm was proposed to address both room reverberation and background noise. Specifically, a deep neural network was trained to estimate the ideal ratio mask, where anechoic-clean speech was considered as the desired signal. Intelligibility testing was conducted under reverberant-noisy conditions with reverberation time $T60 = 0.6$ s, plus speech-shaped noise or babble noise at various signal-to-noise ratios. The experiments demonstrated that substantial speech intelligibility improvements were obtained for HI listeners. The algorithm was also somewhat beneficial for normal-hearing (NH) listeners. In addition, sentence intelligibility scores for HI listeners with algorithm processing approached or matched those of young-adult NH listeners without processing. The current study represents a step toward deploying deep learning algorithms to help the speech understanding of HI listeners in everyday conditions. [Work supported by NIH.]

4aPP19. A deep learning algorithm to increase intelligibility for hearing-impaired listeners in the presence of a competing talker and reverberation. Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH), Masood Delfarah (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH), Eric Johnson (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, johnson.7289@buckeyemail.osu.edu), and DeLiang Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH)

For deep learning based speech segregation to have translational significance as a noise-reduction tool, it must perform in a wide variety of acoustic environments. In the current study, performance was examined when target speech was subjected to interference from a single talker and room reverberation. Conditions were compared in which an algorithm was trained to remove both reverberation and interfering speech, or only interfering speech. A recurrent neural network (RNN) incorporating bidirectional long short-term memory (BLSTM) was trained to estimate the ideal ratio mask (IRM) corresponding to target speech. Substantial intelligibility improvements were found for hearing-impaired (HI) and normal-hearing (NH) listeners across a range of target-to-interferer ratios (TIRs). HI listeners performed better with reverberation removed, whereas NH listeners demonstrated no preference. Algorithm benefit averaged 56% points for the HI listeners at the least-favorable TIR, allowing these listeners to numerically exceed the performance of young NH listeners without processing. The current study highlights the difficulty associated with perceiving speech in reverberant-noisy environments, and it extends the range of environments in which deep learning based speech segregation can be effectively applied. This increasingly wide array of environments includes not only a variety of background noises and interfering speech but also room reverberation. [Work supported by NIH.]

4aPP20. Integration of auditory and tactile stimuli in the perception of building noise and vibration. Ben Loshin and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, loshib@rpi.edu)

Auralization is used in architectural and environmental planning to build a visceral understanding of a design. However, current auralization techniques are limited to the auditory sensory modality, while real environments require the integration of complex stimuli across multiple modalities. This study explores the human perception of coupled sound and vibration sources encountered in real spaces through the creation of immersive virtual representations of those spaces. Acoustic and vibration responses are combined with common building noises and simulated on a calibrated motion platform, incorporating vertical whole-body vibration with binaural audio. Test participants are asked to make judgments of relative loudness and annoyance of building sounds simulated in combination with different vibrational contents. Test results are compared with those published in other studies on the psychophysics of audio-tactile summation, and the implications of the results are discussed with respect to the perception of building noise.

4aPP21. Effects of age and hearing loss on spatial release from speech-on-speech masking, performance in envelope-based psychophysical tasks, and EEG envelope-following responses. Chhayakanta Patro (Psych., Univ. of Minnesota, 311 Harvard St. SE, Apt. 805, Monneapolis, MN 55414, cpatro@umn.edu), Alix Klang (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Heather A. Kreft (Psych., Univ. of Minnesota, Minneapolis, MN), and Magdalena Wojtczak (Psych., Univ. of Minnesota, New Brighton, MN)

Behavioral measures of amplitude modulation (AM) detection and envelope interaural-phase-difference (eIPD) detection reflect listeners' ability to process temporal information. Robust encoding of temporal envelopes is necessary for understanding speech in a complex acoustic environment and for spatial segregation of a target speech from interfering background. It has been suggested that a large variability in performance in psychophysical tasks involving temporal envelope processing and in spatial release from masking for speech intelligibility may arise from cochlear synaptopathy. However, many studies have not found significant correlations between these measures and the amount of self-reported noise exposure in young listeners with audiometrically normal hearing. Similarly, electroencephalographic envelope-following responses did not significantly correlate with noise exposure or with behavioral performance reliant on envelope processing young normal-hearing population. In this study, behavioral measures in psychophysical tasks (AM and eIPD detection) and speech intelligibility in two-talker babble were measured for listeners with normal and near-normal hearing across a wide age range (20 to 69 years). Correlational analyses were performed using behavioral measures and envelope-following responses collected from the same listeners. Results will be discussed in terms of sensitivity of these measures to effects of aging and high-frequency hearing loss. [Work supported by NIH Grant R01 DC015987.]

4aPP22. Reverberation detection threshold estimates in normal-hearing listeners. Pavel Zahorik (Dept. of Otolaryngol. and Communicative Disord., Heuser Hearing Inst. and Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu) and James Shehorn (Heuser Hearing Inst., Louisville, Arizona)

The study of human sensitivity to a single acoustic reflection (echo) has a long and rich history. The influence of time delay, level, direction, and source material are well documented. Unfortunately, real world listening seldom involves only a single reflection. Multiple reflections and reverberation are instead the norm. It is therefore surprising that the detection threshold for acoustical room effects (early reflections plus reverberation) has not been extensively studied, if at all. This study represents an initial step to fill this gap in knowledge. Using virtual auditory space techniques to simulate room acoustic sounds fields over headphones, the detection threshold for reflected/reverberant sound energy was measured for three sound field conditions: a small office-sized room (broadband T60 = 0.5 s), a concert hall (broadband T60 = 1.5 s), and a reference condition with a single echo at 40 degrees to the right of midline. The source signal was a 220 Hz complex tone, 250 ms in duration. Thresholds for the single-echo reference condition and the small room condition were found to be comparable, whereas the concert hall produced thresholds that were at least 20 dB lower. Temporal integration and binaural effects are considered as potential explanations for these results.

4aPP23. Binaural modeling from an evolving habitat perspective. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Functional binaural models have been used since the mid-20th century to simulate laboratory experiments. The goal of this chapter is to extend the capabilities of a cross-correlation model so it can demonstrate human listening in complex scenarios found in nature and human-built environments. A ray-tracing model is introduced that simulates a number of environments for this study. This chapter discusses how the auditory system is used to read and understand the environment and how tasks that require binaural hearing may have evolved over the course of human history. As use cases, sound localization in a forest is examined, as well as the binaural analysis of spatially diffuse and rectangular rooms. The model is also used to simulate binaural hearing during a walk through a simulated office-suite environment. [Work supported by NSF BCS-1539276 and CISL.]

4aPP24. Relationship between localization acuity and spatial release from masking. Nirmal Kumar Srinivasan and Jess Wince (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu)

Spatial Release from Masking (SRM) is defined as the ability to obtain better Speech Recognition Thresholds (SRTs) when the masking sounds are spatially separated from the target sound. Localization refers our ability identify the direction of the sound and localization acuity is measured as the difference in locations between the actual and perceived locations. SRM and localization share many common cues on how the task is performed. Here, we present data from young normal hearing on SRM task using Coordinate Response Measure (CRM) sentences and localization acuity using three different noise Gaussian white noise bursts: low pass (1/3 octave wide centered at 500 Hz), high pass (1/3 octave wide centered at 3150 Hz), and broadband (200–5000 Hz). Thirteen loudspeakers (Orb Mod 1), separated by 15 deg in the frontal plane were used to present the stimuli. Initial analyses of the results indicated that, as expected, all the listeners obtained substantial spatial release from masking consistent with the literature, filtering the broadband noise had little effect on localization acuity for listeners with normal hearing. Finally, the relationship between SRM and localization acuity will be discussed.

4aPP25. Comparison of localization assessments for hearing protection. Eugene Brandewie (Dept. of Psych., GN Adv. Sci., 75 E. River Rd., Minneapolis, MN 55455, ebrandewie@gnresound.com) and Andrew Dittberner (GN Adv. Sci., Glenview, Illinois)

Sound localization is a critical component of the assessment of situational awareness for hearing protection devices (HPDs). A new standard is being developed for the assessment of sound localization in the horizontal-plane concerning head-worn devices. One of the methodologies offered by this standard is a quick and easy-to-use paradigm for rapid prototyping of hearing devices. This study compared the results of this new assessment methodology with a more traditional assessment of horizontal-plane sound localization. Four devices (including open-ear) were tested on normal-hearing individuals using both paradigms. A comparison of localization errors, front-back-reversals, hardware for data collection set-ups, and expected data collection time is presented. Results suggest comparable differences between the two methodologies.

4aPP26. Difference limens for noise bandwidth discrimination in listeners with normal and impaired hearing. Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Frequency-lowering (FL) in hearing aids is often used to move inaudible high-frequency energy from sibilant fricatives to spectral regions where hearing better. Clinically, FL settings that maximize the spectral separation

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between these sounds, which can be modeled as bands of noise, are assumed to maximize discrimination between them and other speech contrasts. The purpose of this study was to quantify the minimum spectral differences for normal-hearing and hearing-impaired listeners to discriminate between bands of frozen noise. Noise bands corresponded to the average frequency and bandwidth of sibilant fricatives after undergoing FL using settings appropriate for mild-to-moderate, moderately-severe, and severely-profound hearing losses. Noise bands differed on the low- and/or the high-frequency edges. Discrimination in normal-hearing listeners was constant across presentation level and the three frequency ranges. Discrimination was also better for high-frequency edge differences. Neural excitation patterns generated from an auditory nerve model account for these findings. Neural excitation patterns generated for the three severities of hearing loss indicate that hearing-impaired listeners will rely heavily on spectral differences on the low-frequency edge and indicate that sibilant fricatives processed with FL will not be able to be discriminated solely on the basis of high-frequency edge differences. [Grant supported by Sonova USA, Inc.]

4aPP27. Can listeners reliably identify their preferred amplification profiles for speech listening? Donghyeon Yun, Yi Shen, and Zhuohuang Zhang (Speech and hearing Sci., Indiana Univ. Bloomington, 1603 E. 3rd St. 216, Bloomington, IN 47401, dongyun@iu.edu)

Personal hearing devices, such as hearing aids, may be fine-tuned for individual users' preferences by allowing them to self-adjust the amplification profiles. The purpose of the current study was to compare two self-adjustment methods in terms of their test-retest reliability. Both methods estimated preferred amplification profiles in six octave-frequency bands using the method of adjustment. In one method (method A), listeners adjusted the gain in one of six frequency bands using a programmable knob on a given trial; while in the other method (method B), listeners adjusted the gains in all six bands simultaneously according to a linear model using the same programmable knob. Ten normal-hearing listeners participated in the study. The experiment was completed in two test sessions, at least one week apart. During each session, the preferred amplification profile was estimated using both methods. Running speech in quiet or in speech-shaped noise was used as the test stimuli. At the beginning of each method, the initial amplification profile was generated randomly with the gains drawn from a uniform distribution spanning between -25 and 25 dB. The test-retest reliability for method B was better than method A. For method B, the test-retest reliability was better at lower signal-to-noise ratios.

4aPP28. Perception of musical instruments and music genres in cochlear implant recipients. Ying Hsiao, Valeriy Shafiro, Chad Walker, Jasper Oh, Megan Hebb, Kelly Brown, Stanley Sheft (Dept. of Commun. Disord. and Sci., Rush Univ., 600 S. Paulina St., Chicago, IL 60612, ying_y_hsiao@rush.edu), Kara Vasil, and Aaron C. Moberly (Dept. of Otolaryngol.-Head & Neck Surgery, Ohio State Univ. Wexner Medical Ctr., Columbus, OH)

We examined perception of musical and nonmusical stimuli in 17 experienced postlingually deafened cochlear implant (CI) recipients and 10 normal-hearing (NH) listeners using real-world music excerpts derived from the Appreciation of Music in Cochlear Implantees (AMICI) test. Following stimulus presentation, participants selected one most appropriate option among nine instrument or five genre options. Compared to NH, CI listeners demonstrated reduced instrument (99.1% vs. 68.7%) and genre (96% vs. 55.7%) identification performance. For CI listeners, the least accurately identified instruments were flute (17.6%) and saxophone (37.2%), while the drums were most accurately identified (98%). The flute was most often confused with strings (76.5% error) and the saxophone was most confused with brass instruments (23.5% error). The least accurately identified genres were Latin (41.2%) and Rock "n" Roll (41.2%), while Classical was most accurately identified (82.5%). Latin was most often confused with Rock "n" Roll (26.2% error), and Rock "n" Roll was most often confused with Country (28.7% error). For the CI recipients, instrument and genre identification strongly and significantly correlated with recognition of environmental sounds, sentences in noise, and frequency pattern discrimination. These results indicate considerable deficits in music perception in CI recipients and indicate the need for further rehabilitation. Correlations with speech further suggest a potential for cross-domain improvements.

4aPP29. 3D printed pinna embedded in circumaural hearing devices for spectral cue preservation. Carlos Acosta Carrasco (W.M. Keck Ctr. for 3D Innovation, The Univ. of Texas at El Paso, 500 W University Ave., El Paso, TX 79968, cfacostacarrasco@miners.utep.edu), Vidya Krull, Andrew Dittberner (GN Adv. Sci., GN Hearing, Glenview, IL), and Ryan Wicker (W.M. Keck Ctr. for 3D Innovation, The Univ. of Texas at El Paso, El Paso, TX)

Innovations in additive manufacturing [three-dimensional (3D) printing] have allowed for the fabrication of objects as complex as the human ear. The visible part of the human outer ear (pinna) serves as a funnel and a natural filter for incoming sound. Spectral cues generated by the pinna help with auditory localization and externalization. In an attempt to preserve spectral cues when using circumaural hearing devices, the present work explored the use of 3D printing to fabricate individualized pinna within custom-designed and fabricated hearing devices. Through 3D scanning, a computer aided design (CAD) model of a pinna from an anthropometric mannequin was generated to replicate human pinnae. Multiple 3D printing technologies were used to fabricate the CAD model, investigating different material options, dimensional accuracies, and overall printing costs. The fabricated pinnae were subjected to acoustic testing to assess spectral cue preservation by comparing mannequin head-related transfer functions obtained with the printed pinnae to those with the original. A sample circumaural hearing device with the pinna embedded within it was then designed, fabricated, and subjected to the same acoustic testing for comparison. Results from testing will be described within the context of providing individualized circumaural hearing devices that assist with spectral cue preservation.

4aPP30. Exploring the relationship between sound localization and individual use of spectral and temporal cues among hearing-impaired listeners. Gregory M. Ellis and Pamela Souza (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL 60201, gregory.ellis@northwestern.edu)

When locating a sound source, listeners are expected to rely more on spectral cues for elevation and temporal cues for azimuth. This work investigates whether sound localization ability can be predicted by individually-measured use of spectral and temporal cues. Participants were older adults with sensorineural hearing loss. Sound sources for the localization task were created in a virtual room, using a mix of virtual and physical loudspeakers in the front hemifield of the listener. Sources were distributed evenly between $\pm$90 deg azimuth and $\pm$20 deg elevation. The signal was a 1 s broadband 4-Hz amplitude-modulated noise. To assess the relationship between localization ability and use of spectral and temporal cues, listeners performed cue weighting and cue discrimination tasks. In the cue weighting task, listeners identified synthetic speech sounds that varied spectro-temporally. The cue discrimination task measured the smallest detectable difference in either spectral or temporal information among the same set of ambiguous speech sounds. Together, these tasks form a cue profile which identifies whether the listener relies to a greater extent on temporal or spectral cues. Older hearing-impaired listeners varied in their ability to localize sounds. Localization results will be discussed in the context of cue profiles and audiometry. [Work supported by NIH.]

4aPP31. Benefits from different types of acoustic beamforming in bilateral cochlear-implant listeners. David Yun (Hearing and Speech Sci., Univ. of Maryland, College Park, 0100 Lefrak Hall, College Park, College Park, MD 20742, davidyunjr@gmail.com), Todd R. Jennings, Christine Mason, Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci., Boston Univ., Boston, MA), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD)

Acoustic beamforming improves speech reception in "noise" at the cost of removing spatial cues. Recently, novel approaches have been proposed for enhancing speech reception via beamforming while preserving sound localization by reintroducing natural binaural cues to the beamformer output. It is, however, unclear whether such a hybrid approach will improve

performance for bilateral cochlear-implant (BICI) listeners because they receive little benefit from natural spatial cues. Here, effects of beamforming on masked speech reception thresholds (SRTs) were determined in BICI and normal-hearing (NH) listeners for a standard single-channel beamformer and a new triple-beamformer comprising a diotic central beam, -40 deg left-ear-only beam, and +40 deg right-ear-only beam [Jennings and Kidd, *JASA* **143**, 1743]. Speech targets and maskers were co-located (0 deg) or maskers were symmetrically separated at ±30 deg or ±90 deg. Natural binaural listening, single-channel beamformer, and triple-beamformer conditions were tested. Both beamformers were better than natural cues. For BICI listeners, single-channel beamforming improved SRTs by 5 (±30 deg) and 15 dB (±90 deg). SRTs for the triple-beam were best for NH but were 5 dB worse than the single-channel beam for BICI listeners. These preliminary findings suggest beamforming provides significant spatial release from masking for BICI listeners but assessing overall benefit, including localization, awaits further study.

4aPP32. Selective deficits in a clinical population with self-reported hearing difficulties but normal audiometric thresholds. James Shehorn (Heuser Hearing Inst., 117 E Kentucky St., Louisville, KY 40203, jshehorn@thehearinginstitute.org), Olaf Strelcyk (Sonova U.S. Corporate Services, Cincinnati, OH), and Pavel Zahorik (Otolaryngol. Head and Neck Surgery and Communicative Disord., Univ. of Louisville, Louisville, KY)

Audiologists often encounter patients who report hearing difficulties despite having normal audiometric thresholds. Many of these patients are told that they have normal hearing, although it is possible that the current typical audiometric test battery is not sensitive to these hearing difficulties. A test battery including immittance testing, binaural listening tasks, cognitive testing and subjective questionnaires of speech understanding, spatial hearing, and annual noise exposure was administered to 26 patients (age range: 18–53 years) who had sought out audiologic assessment at a community clinic but had normal hearing sensitivity. Despite normal hearing sensitivity, the participants in the study exhibited deficits in several binaural listening tasks, weakened middle-ear muscle reflexes, and rated their speech understanding and spatial hearing as being significantly worse than a normative population with normal hearing sensitivity. This patient group did not perform significantly worse on any of the cognitive measures nor did they report significantly more noise exposure than a normative young adult population; however, nearly two-thirds of participants were classified as “high risk” for noise exposure. These findings confirm self-reported hearing difficulties reported by this audiometrically “normal” population and suggest that several of the measures used in this study should be considered for standard audiologic evaluation.

4aPP33. Portable psychoacoustics with passive and active noise-attenuating headphones. Esteban S. Lelo de Larrea Mancera (Hearing & Speech Sci., Univ. of Maryland, 900 University Ave., Psych. Bldg. Rm. 3209, Riverside, California 92521, elelo001@ucr.edu), Trevor Stavropoulos (Psych., Univ. of California Riverside, Riverside, CA), Frederick J. Gallun (Dept. of Veterans Affairs, VA RR&D NCRAR, Portland, OR), David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), Eric C. Hoover (Hearing & Speech Sci., Univ. of Maryland, Tampa, Florida), and Aaron Seitz (Psych., Univ. of California Riverside, Riverside, CA)

This presentation will describe data collected using a freely available procedure for psychoacoustical testing that harnesses commercially available tablet computer technology to address the current gap between (1) modern auditory neuroscience and psychophysics and (2) current clinically available tests of hearing. Portable Automated Rapid Testing (PART) measures running on an iPad were used to evaluate the detection of (a) tones in noise; (b) spectral, temporal, and spectro-temporal modulation; (c) monaural and binaural frequency modulation; and (d) brief temporal gaps inserted between brief tone pulses. Listeners also performed a spatial release from speech-on-speech masking task. Data from 151 UCR undergraduates were collected using both passive and active noise-attenuating headphones in a quiet environment and in the presence of recorded cafeteria noise. Across these and several other manipulations of equipment and threshold-estimation techniques, performance approximated that reported in the literature. These data provide a distribution of thresholds that can now be used as a normative baseline against which auditory dysfunction can be identified in future work.

4aPP34. Musical emotion perception in bimodal patients: Relationship between bimodal benefit and neural representation of temporal fine structure using Rhodes piano stimuli. Kristen D’Onofrio (Hearing and Speech Sci., Vanderbilt Univ., 1215, Nashville, TN 37215, kristen.l.donofrio@vanderbilt.edu), Spencer Smith (Univ. of Texas at Austin, Austin, TX), David Kessler, Grace Williams, and René Gifford (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN)

Combining electric and acoustic hearing across ears allows significant “bimodal hearing” benefit for speech recognition, sound quality, and music perception. The degree of bimodal benefit for speech recognition and musical emotion perception is significantly correlated with neural representation of F0 envelope using the frequency following response (FFR) for a 170-ms /da/ stimulus (D’Onofrio *et al.*, in prep). The purpose of the current study is to examine the relationship between bimodal benefit for musical emotion perception and neural representation of F0 using Rhodes piano stimuli at the following fundamental frequencies: 98 Hz (G2), 262 Hz (C4), and 440 Hz (A4). Our hypotheses are (1) the correlation between bimodal benefit and neural representation of F0 and temporal fine structure will be strengthened via use of a “music” stimulus, compared to the /da/ “speech” stimulus, and (2) bimodal benefit for speech recognition will be better explained via FFR for speech stimuli. Stimuli were presented at 90 dB SPL to the non-implanted ear of bimodal listeners using magnetically shielded insert earphones. Implications regarding the clinical utility of FFR will be discussed, with particular attention given to its use as an objective measure of expected bimodal benefit for speech recognition and musical emotion perception.

4aPP35. Spatial release from masking and sound localization using real-time sensorineural hearing loss and cochlear implant simulation. Hannah M. Wright, Wesley Bulla, and Eric W. Tarr (Audio Eng., Belmont, 1900 Belmont Boulevard, Nashville, TN 37212, hannah.wrightmusic@gmail.com)

Simulations of sensorineural hearing loss (SNHL) and bilateral cochlear implantation (BCI) have been modeled successfully under static and non-real time conditions. This study performed two experiments testing the validity of a novel real-time SNHL/BCI simulation application for iOS using an in-ear binaural-recording headphone apparatus. The first experiment measured spatial release from masking (SRM) with normal hearing (NH), against headset apparatus simulations of NH, SNHL, and BCI using HINT sentences, speech shaped noise and forward masking. A one-sample *t-test* revealed significant differences between NH and simulated SNHL and BCI conditions showing reduced benefit from SRM. The second experiment employed noise bursts across nine frontal-plane loudspeakers and measured localization accuracy under NH and six frequency band BCI simulation conditions. Repeated measures two-way ANOVA and Cronbach’s Alpha suggested significantly reduced localization ability with BCI simulation. While further testing is needed, results here provide promising evidence that real-time binaural recording with low-latency processing and in-ear playback may be used to simulate SNHL and the BCI percept in NH listeners. The limitations and potential of this technology to expand the subject pool and expedite innovative testing are discussed.

4aPP36. Predicting speech-cue weighting in older people with impaired hearing. Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Richard Wright (Dept. of Linguist, Univ. of Washington, Seattle, WA), and Pamela Souza (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Previously published data (Souza *et al.*, 2015; 2018) revealed individualized patterns of cue-weighting in the identification of synthetic speech (bah, dah, lah, wah). The speech tokens were constructed such that temporal modulation (TM; envelope rise-time) and spectrotemporal modulation (STM; formant transitions) were systematically varied. Here, a

discrimination task with the speech stimuli and a modulation detection task with noise tokens were related to cue weighting for thirty people aged 61–90 years with impaired hearing. Discrimination of STM in the speech stimuli was correlated with both STM sensitivity and cue weighting, while TM discrimination and the audiogram were not. The direct link between STM sensitivity and cue weighting was not supported, however. Results suggest that there was no substitute for actually measuring the ability to discriminate the speech stimuli on the basis of spectrotemporal information. The discrimination task may provide a more time-efficient measure of cue weighting than the cue identification task. These results have important implications for the rehabilitation of impaired hearing, which is generally guided only by the audiogram, which was the worst predictor of cue weighting tested. [Work supported by NIH R01 DC006014 (PI: Souza) and NIH R01 DC015051 (PI: Gallun).]

4aPP37. Effects of training on sensitivities to spatial changes in auditory scenes. Natalie J. Ball (Cognit. Psych., Univ. at Buffalo, 373H Park Hall, University at Buffalo, Buffalo, NY 14260, njball@buffalo.edu), Matthew Wisniewski (Kansas State Univ., Manhattan, KS), Brian Simpson (U.S. Air Force Res. Lab., Wright-Patterson AFB, OH), and Eduardo Mercado (Cognit. Psych., Univ. at Buffalo, Buffalo, NY)

Previous work has shown that change deafness can occur with changes in the spatial location of objects within auditory scenes. Whether performance can be improved with training has yet to be directly tested. In the present study, the impact of training was examined in a "flicker"-like paradigm, where an initial scene comprising environmental sounds presented on the horizontal plane alternated with the presentation of a comparison scene, which was either the same or contained a change in the location of one or more sounds. Trained participants were trained on a set of sounds on day 1, while control subjects completed an unrelated visual task. On day 2, participants were tested using the same paradigm, with trained subjects hearing either the sounds they were trained on or new sounds, and controls being exposed to the paradigm for the first time. Overall, trained participants performed better (93% accuracy) than untrained participants (79%). Both groups had lower reaction times on correct-response change trials than on correct-response no-change trials. Trained subjects performed no better on trained sounds than new sounds. These data indicate that training can improve task performance, but improvements may be not be limited to sounds experienced during training.

4aPP38. Further analysis of behavioral measures of cochlear gain and gain reduction in listeners with normal hearing or minimal cochlear hearing loss. Elizabeth A. Strickland, Miranda Skaggs, Nicole Mielnicki, William Salloom, Hayley Morris, and Alexis Holt (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, estrick@purdue.edu)

This is a continuation of a study examining the relationship between cochlear hearing loss and psychoacoustic measures thought to be related to a cochlear function. In the listeners tested, audiometric thresholds for long tones ranged from well within the clinically normal range to just outside this range. Where thresholds were elevated, other clinical tests were consistent with a cochlear origin. Because the medial olivocochlear reflex decreases cochlear gain in response to sound, when possible, measures were made with short stimuli. Signal frequencies were from 1 to 8 kHz. One point on the lower leg of the input/output function was measured by finding threshold masker level for a masker almost one octave below the signal frequency needed to mask a signal at 5 dB SL. Gain reduction was estimated by presenting a pink broadband noise (BBN) precursor before the signal and masker and measuring the change in signal threshold as a function of the precursor level. Previous studies with listeners with normal hearing have shown that gain reduction begins at a low precursor level and grows compressively as the precursor level is increased. The current study is designed to determine whether this pattern changes when cochlear gain is permanently reduced. [Work supported by NIH(NIDCD)R01 DC008327 (EAS), the Purdue Office of the Executive Vice President for Research, and the Purdue Graduate School (WBS).]

4aPP39. Effects of self-reported hearing difficulty on intensity discrimination judgments. Gwen O. Saccocia and Joseph C. Toscano (Dept. of Psychol. and Brain Sci., Villanova Univ., 800 E Lancaster Ave., Villanova, PA 19085, gsaccoci@villanova.edu)

Listeners may report difficulty understanding speech, particularly in background noise, despite having normal audiograms that do not suggest a sensorineural hearing loss. One potential cause of this hearing difficulty is auditory neuropathy (AN), a disruption in the function of the auditory nerve. AN may specifically affect auditory nerve fibers that code intensity differences at higher sound levels, resulting in particular difficulty with speech recognition but preserved audiometric thresholds. This study aims to detect AN by measuring intensity discrimination thresholds at different frequencies and sound levels, comparing performance to self-reported measures of hearing difficulty. Listeners performed an intensity discrimination task where they heard pairs of tones and judged whether the first or second tone was louder. Psychophysical functions were computed to measure listeners' discrimination thresholds and point of subjective equality (PSE, i.e., point at which the two tones are judged to have equal intensity). Results showed that listeners who report greater speech-in-noise difficulty had shifted PSEs specifically at higher sound levels (60–70 dB SPL, the range used in conversational speech), such that they were more likely to perceive the second tone as louder than it was. The results suggest that an intensity discrimination task may be a useful test for AN.

4aPP40. Experiment on the relationship between the dynamic range of hearing and psychophysical tuning curves. Marc Brennan (Special Education and Commun. Disord., Univ. of Nebraska-Lincoln, 4075 East Campus Loop South, Lincoln, NE 68583, Marc.Brennan@unl.edu)

In this study, we hypothesized that improving the dynamic range of hearing for adults with hearing loss would improve spectral resolution by providing better audibility across frequency and—due to the higher sensation level—improved phase locking. However, limited outer hair cell (OHC) function and spread of excitation likely curtail the benefit of improved audibility on spectral resolution. In this experiment, the relationship of the dynamic range of hearing to measures of psychophysical tuning curves was quantified for 13 adults with hearing loss. Twenty-one adults with normal hearing served as controls. A dynamic range of hearing was manipulated by systematically adjusting hearing-aid gain and compression ratios. To better understand the mechanisms that support the encoding of spectral resolution with amplification, the behavioral data were modeled using a cochlear excitation model. For most listeners, the psychophysical tuning curves were similar regardless of changes in the dynamic range of hearing. For a subset of participants, tuning on the low frequency side improved as the dynamic range of hearing was increased—which could be accounted for in the model by residual OHC function. The model better accounted for unaided than aided measures of tuning.

4aPP41. A software controlled audiometric setup to predict types of hearing loss from types of response delays. Amitava Biswas, Edward L. Goshorn, and Jennifer Goshorn (Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, SHS Res. Lab., Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Diagnosis of certain types of non-organic hearing loss is often confused with organic hearing loss. A software controlled audiometric setup has been designed by the authors for this purpose. Its salient features are (1) partially randomized step sizes, (2) partially uncorrelated increase and decrease in the stimulus intensity after each response or no response of the subject, (3) recording of corresponding response delays of the subject, and (4) providing audio-visual feedback. Its sensitivity, specificity, and applications in clinics and the industry will be discussed.

4aPP42. Effects of auditory-nerve loss on tone detection in roving-level noise. Kenneth S. Henry, Kassidy N. Amburgey (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave., Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu), and Kristina S. Abrams (Neurosci., Univ. of Rochester, Rochester, NY)

Auditory-nerve (AN) loss has emerged as a significant public health concern because it occurs steadily with age and potentially following noise-induced temporary threshold shifts. AN loss without hair-cell damage remains undetectable with an audiogram yet is commonly assumed to degrade auditory perception under real-world, noisy conditions. Here, we tested whether AN loss impacts behavioral tone-in-noise (TIN) detection in the budgerigar, an avian species with sensitivity similar to humans on many simple and complex listening tasks. AN damage was induced with kainic acid and confirmed using auditory evoked potentials and otoacoustic emissions. TIN thresholds were quantified in 1/3-octave noise as a function of frequency and sound level using operant conditioning and two-down, one-up, adaptive tracking procedures. Kainic acid reduced gross AN potentials by 40%–70% across animals without impacting otoacoustic emissions. TIN thresholds in control animals decreased with increasing frequency and showed minimal elevation (<1 dB) when sound level was roved ± 10 dB across trials. TIN thresholds in kainic-acid exposed animals were as sensitive as in the control group and showed similar preservation with roving sound level. These results suggest a minimal impact of AN loss on behavioral TIN detection, even under conditions requiring rapid adaptation to changing sound level.

4aPP43. Evaluating new hearing aid technologies in laboratory simulations of listening scenarios. Peggy B. Nelson, Elizabeth Anderson, Trevor T. Perry, Kristi Oeding, and Andrew Byrne (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu)

It can be important for clinical researchers to be able to evaluate the performance of sensory aids using both objective and subjective methods. New technology (such as self-fit hearing aids) can be evaluated in a laboratory setting in calibrated listening scenarios that reflect daily listening situations. In the Center for Applied and Translational Sensory Science (CATSS) multisensory laboratory, we have developed simulations of challenging conversational scenarios so that users of sensory aids can make judgments of sensory aid performance in realistic but controlled conditions. Listeners with hearing loss make ratings of intelligibility, sound quality, and preference in scenarios such as small group conversations and entertainment listening. At the same time, measures of hearing-aid gain and speech intelligibility are obtained. These ratings are compared to outcome measures such as the Speech, Spatial, and Qualities of Hearing Scale (SSQ; Gatehouse and Noble, 2004) and Social Participation restrictions questionnaire (SPaRQ; Heffernan *et al*., 2018) to determine relationships between intelligibility, preference, benefit, and hearing aid gain. Results will help refine methods for evaluating the performance of emerging technologies for hearing loss. [Work supported by NIDCD R01 DC 13267.]

4aPP44. Factors influencing auditory localization with deep insertion hearing aids or earplugs. Douglas Brungart (Walter Reed NMMC, 4401 Holly Ridge Rd., Rockville, MD 20853, dsbrungart@gmail.com), Nathaniel Spencer (AFRL/711th HPW, Wright-Patterson AFB, OH), Nina Pryor (AFRL/711th HPW, WPAFB, OH), Eric R. Thompson (AFRL/711th HPW, Wright-Patterson AFB, OH), Nandini Iyer (AFRL/711th HPW, Wpafb, OH), Griffin D. Romigh, and Brian Simpson (AFRL/711th HPW, Wright-Patterson AFB, OH)

Virtual localization experiments have demonstrated that Head Related Transfer Functions measured a few millimeters inside a blocked ear canal can produce localization performance approaching what is measured in the free field. This suggests that an earplug inserted entirely inside the ear canal should be able to preserve normal localization performance so long as the stimulus is loud enough to overcome any insertion loss in the device at all frequencies. In this study, localization performance of normal-hearing listeners was measured with the Lyric extended wear hearing aid, both in active mode (where it acted like an electronic pass-through earplug) and in passive mode (where it acted like a passive hearing protector). In an active mode, localization accuracy approached the open-ear condition. However, under the passive condition, localization was much worse than with the open ear even at high stimulus levels where the full spectrum should have been audible. This result suggests there may be fundamental limitations on localization accuracy with passive hearing protection that are unrelated to the directionality of the HRTF. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of the Army/Air Force, Department of Defense, or U.S. Government.]

4aPP45. The effects of age on narrowband and broadband measures of spectral processing in listeners with hearing loss. Kristi Oeding and Evelyn E. Davies-Venn (SLHS, Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55445, venn@umn.edu)

The propensity for degraded auditory perception increases with age. Several studies have shown that while age-related hearing impairment explains a high percentage of the often-reported degradation in auditory perception, there still remain some effects that can only be attributed to the aging process. Even though some classic studies have shown that spectral processing may be immune to age-related degradation, some recent work with broadband measures of spectral processing appears to challenge this notion. This study evaluated the effect of age on narrowband and broadband spectral processing abilities for individuals with mild-to-moderate hearing loss. We controlled for the amount of hearing loss and measured auditory filter bandwidths using notched-noise masking and spectral modulation detection using rippled noise in the same cohort of listeners. Results to date suggest that broadband spectral processing, which uses stimuli that share ecological validity with speech, may be more sensitive to age-related changes in spectral processing compared to narrowband spectral processing.

4aPP46. Executive functions predict improvements in pure-tone thresholds for children with normal hearing and children with hearing loss. Ryan W. McCreery, Lori Leibold (Audiol., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ryan.mccreery@boystown.org), Emily Buss (Univ. of North Carolina, Chapel Hill, NC), and Elizabeth Walker (Univ. of Iowa, Iowa City, IA)

Detection of pure tones is known to improve in early childhood. The ability to suppress self-generated noise during detection tasks has been identified as a contributing factor to higher audiometric thresholds in children. Data from children with hearing loss indicate that children with thresholds that are above the level of self-generated noise do not show the same improvements across age as peers who can hear their self-generated noise. In an effort to examine the specific cognitive mechanisms that affect this process, audiometric thresholds and executive functions skills were measured in a group of 213 1st and 3rd grade children (84 children with normal hearing, 129 children with hearing loss). The relationship between selective attention and vigilance and thresholds were assessed for both groups. Children with stronger selective attention and vigilance skills had lower pure-tone thresholds than peers with poorer skills in these domains. The previous finding that children only showed an improvement in thresholds when they could hear their self-generated noise was also replicated. Audiometric test equipment that can monitor the sound level in the ear canal may help to increase the accuracy of clinical pure-tone detection tasks in children.

4aPP47. Just-noticeable differences of fundamental frequency change in Mandarin-speaking children with cochlear implants. Wanting Huang, Lena Wong (Div. of Speech and Hearing Sci., The Univ. of Hong Kong, Rm. 730, Meng Wah Complex, Hong Kong 999077, Hong Kong, wtwong88@connect.hku.hk), and Fei Chen (Dept. of Elec. and Electron. Eng., Southern Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Fundamental frequency (F0) provides the primary acoustic cue for lexical tone perception in tonal languages but is poorly presented in cochlear implants (CIs). Currently, there is still a lack of understanding on the sensitivity to F0 information in CI users speaking tonal languages. In the present study, just-noticeable differences (JNDs) of F0 contour and F0 level change in Mandarin-speaking kindergarten-aged children with CIs were measured and compared with those in age-matched normal-hearing (NH) peers. Statistical analysis showed that both JND of F0 contour change (JND-C) and JND of F0 level change (JND-L) were significantly larger in CI group than in NH group. Furthermore, within-group comparison of JND-C and JND-L found that JND-C was significantly smaller than JND-L among children with CIs; however, opposite pattern was observed among children with normal hearing. The contrary sensitivity to F0 contour and F0 level change between children with CIs and children with normal hearing suggest discrepant mechanisms of F0 processing in these two groups as a result of hearing experience.

4aPP48. Assessment of a feasible virtual acoustics method for testing hearing aids using the Hearing-Aid Speech Perception Index. Sungbeen Cho, Scott Aker, and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 1212, London, ON N6G 1H1, Canada, scho255@uwo.ca)

Significant differences have been found in hearing aid (HA) performance between laboratory and real world test environments. Virtual sound environments provide a degree of control and reproducibility which is lacking in real world testing but may require an impractical number of loudspeakers. We assessed the accuracy of a simulation approach in which sources' direct sound is delivered by single loudspeakers while room acoustics are reproduced using low-order Ambisonics and a small number of loudspeakers. In a large office, we recorded binaural hearing aid output in response to sentence targets and babble noise presented at various levels and from various combinations of four loudspeakers surrounding a manikin. We measured the loudspeakers' room impulse responses (IRs) using a 32-channel spherical microphone array (Eigenmike), and split the IRs into "direct sound" and "room sound" portions. In an anechoic chamber, the original acoustics were simulated using Ambisonics or discrete loudspeakers for each source's direct portion and Ambisonics for the room portion. Ambisonic order and/or number of playback loudspeakers were also varied. HA output in the simulations was recorded using the manikin and assessed by comparing Hearing-Aid Speech Perception Index (HASPI) values computed on the simulation recordings with those made in the original room.

THURSDAY MORNING, 16 MAY 2019 STOPHER, 8:00 A.M. TO 11:00 A.M.

Session 4aSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration II

Benjamin Shafer, Cochair
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Contributed Papers

8:00

4aSA1. Power transmission metrics and applications in the design of quiet structures. Jonathan D. Young and Kyle R. Myers (Appl. Res. Lab., Penn State Univ., 3220B G Thomas Water Tunl, P.O. Box 30, State College, PA 16804, krm25@arl.psu.edu)

Connected structures subject to applied dynamic loads transfer vibrational energy through their connecting junctions. Identifying the dominant paths of transmission and characterizing the power flow through those paths is important for designing a quiet structure. When the connection type is known, one way of characterizing the transmitted power flow is to identify which degrees of freedom (i.e., translational and rotational) is most responsible for transmission through the junction. Another way could identify vibrational modes that dominate the transmission. This research presents several examples that characterize power flow between structures in physical and modal space. The structures examined here may be connected through springs, point impedances, or a generalized impedance matrix. Key questions considered are how changes to the system affect transmitted power, and how the results can be used to design quieter structures.

8:15

4aSA2. Development of vibrational metrics for internal damage scenarios of a scaled Transnuclear-32 dry storage cask for spent nuclear fuel. Kevin Y. Lin (Phys. and Astronomy, and National Ctr. for Physical Acoust., Univ. of Mississippi, 145 Hill Dr., Oxford, MS 38677-1848, klin@go.olemiss.edu), Joel Mobley, Wayne E. Prather, Zhiqu Lu, Gautam Priyadarshan, and Josh R. Gladden (Phys. and Astronomy, and National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

The assessment of the internal structural integrity of dry storage casks for used high burnup nuclear fuel assemblies is of critical importance before transporting these to permanent repositories. The large size and structural complexity of the Transnuclear-32 (TN-32) cask as well as the inability to access its interior make this a challenging task. To address these difficulties, we use an active acoustics approach to develop metrics that are sensitive to the internal configuration of these casks. A 6:1 scaled model of the TN-32 cask was constructed in order to study the internal configuration of the fuel assemblies including various damage scenarios. Each mock-up fuel assembly consists of bundled steel rods, and their structural failure is mimicked

by steel shot of equal weight. This talk will report the amplitude- and phase-based active acoustics metrics that we developed to characterize different levels of internal damage. Our studies indicate that vibrometric signatures of various internal conditions can be measured using sources and sensors mounted on the exterior shell. Our current methodology is sensitive enough to detect structural failures at the single fuel assembly level. [Work supported by DOE NEUP Award No. DENE0008400.]

8:30

4aSA3. Working gases based scaling-down limitations for thermoacoustic coolers: Miniaturization approach. Anas M. Abdelrahman and Xiaoqing Zhang (Dept. of Refrigeration and Cryogenics, School of Energy and Power Eng., Huazhong Univ. of Sci. and Technolgy, 1037#, Luoyu Rd., Hong Shan District, Wuhan 430074, China, arahman@hust.edu.cn)

Regarding miniaturization of thermoacoustic coolers for thermal management purposes, working gases play a key role as the primary media responsible for producing the so-called "thermoacoustic effect" with their interaction with solid media (i.e., secondary media) within stacks or regenerators. However, the role of working gases in limiting scaling-down of thermoacoustic coolers still needs more investigations compared to addressed operational parameters (i.e., mean pressure, temperature difference across stack, etc.). In the present study, a theoretical computational analysis, based on published literature work, would be conducted to investigate allowable minimum sizes of standing-wave thermoacoustic coolers under the effects of working gases thermo-physical properties with considering adverse effects limitation of thermal conduction losses. Different working gases including air and either pure or mixture noble gases have been used for such geometrical scaling-down analysis under specific operating conditions. Moreover, cooling power was focused here as the more desirable performance indicator rather than the efficiency. The results had revealed the cooling capability at different scale levels based on different working gases, which make gases properties significantly contribute to scalability of thermoacoustic coolers to meet the cooling needs for micro-electronics. In addition, more research work will be devoted to other scaling-down issues of thermoacoustic systems.

8:45

4aSA4. Suspension optimization for a compressor assembled on a refrigerator. Alexandre A. Pescador Sarda (Mech. Eng., UFPR, Av. Cel. Francisco H. dos Santos, 100, Curitiba, PR 81530000, Brazil, pescador@ufpr.br) and Arcanjo Lenzi (Engenharia Mecânica, UFSC, Florianópolis, SC, Brazil)

Noise annoyance generated by electrical machines is evaluated based on the sound power level (SPL) parameter measured in a reverberationroom or using a semi-anechoic chamber. However, when the machine is assembled in a base plate, the SPL can be altered depending on the new system configuration and the way the machine is assembled in the final product. The vibration generated in the electrical machine is transmitted to the base resulting in noise at the surface. The aim of this study was to model the suspension of a general machine assembled in a flexible base and minimize the vibratory power flow transmitted to the base through an optimization process, taking into account parameters such as the spring inclination, stiffness, and suspension damping. Decreasing the power flow to the base results in a reduction in the global levels of noise and vibration at the base plate.

9:00

4aSA5. Broadband sound localization with gradient helical structure. Jie Zhu (Mech. Eng., Hong Kong Polytechnic Univ., FG603, Kowloon, Hong Kong, jiezhu@polyu.edu.hk)

Acoustic sensors or microphones are essential equipments for the detection of sound signals. However, sound signal suffers from the inevitable attenuation due to many reasons, such as diffusion, damping, thermal, and viscous loss. To solve such a problem, we introduce a gradient acoustic metamaterial to magnify the sound signal before it can be coverted to the electrical signal by introducing a gradient refractive index along the sound propagating route. In this case, the acoustic signal can be magnified as the wave is compressed by the gradient increasing index. The helicoid metamaterial has the advantage to adjust acoustic parameters flexibility and continuity by changing the pitch of the blades. This design is of great significance to improve the working condition of signal detection and may contribute to the design of other sensors.

9:15

4aSA6. Resonance frequencies of a spherical aluminum shell subject to prestress from internal fluid pressure. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu) and Natalie Harris (Phys., Whitman College, Walla Walla, WA)

Vibration measurements of a spherical aluminum shell (6 in. diameter) filled with water show that the resonance frequencies of the shell shift higher or lower with increasing water pressure, depending on the specific mode of vibration. For a given mode, the rate of frequency shift with pressure change $\Delta f/\Delta p$ is approximately linear for gauge pressures up to 100 psi. Frequency shifts were detected for pressure changes as small as 0.2 psi or 10 mm Hg. Observations of positive frequency shifts are consistent with previous studies (from the 1950s) involving submerged cylindrical shells subject to much larger pressures. Analysis from this era suggests that the phenomenon is due to geometric nonlinearity; however, the negative frequency shift observed with low order modes is not predicted by this theory. The feasibility of developing a noninvasive method for monitoring intracranial pressure using shifts in skull resonance frequencies will also be discussed.

9:30

4aSA7. On the investigation of the natural mode characteristics of an internal supporting substructure interacting with a submerged main structures in terms of acoustic radiations. Pei-Tai Chen (Dept. of System Eng. and Naval Architecture, National Taiwan Ocean Univ., No. 2,Pei-Ning Rd. Keelung 20224, Taiwan, ptchen@mail.ntou.edu.tw)

This paper investigates acoustic radiation characteristics of an internal structure supported in a submerged main structure. An exciting force is applied on the internal structure where vibration is transmitted across the interfacial boundary between these two structures and radiating acoustic power into water through the surface contacting with water. In the previous study (Chen, in *176th ASA meeting*, Vol. 144, p. 1680), compliant matrices describing the internal structure and the main submerged structure are proposed, which are both symmetric matrices where the imaginary part of the compliance matrix of the submerged main structure is responsible for acoustic radiation, whereas the matrix for the internal supporting structure is real. These two compliance matrices which are defined on the interfacial boundary fully describing the submerged structural dynamics of this coupled system. It was shown that the parameters, such as thickness and stiffened plate, of the internal structures are very sensitive variations of acoustic radiations, although the stiffness of the internal structures are much lower as compared with the main submerged structure. The present study addresses the natural mode characteristics of the internal structure, such as natural frequencies and mode shapes, affecting power flow across the connecting boundary and thus radiating into water. Two sets of modal expansions for forces and displacements defined on the connecting interfacial boundary are established to investigate acoustic radiations.

9:45–10:00 Break

10:00

4aSA8. A parametric resonance based capacitive ultrasonic transducer for wireless power transfer in air. Sushruta Surappa and F. Levent Degertekin (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Rm. 318, Atlanta 30332, GA, sushsurappa@gatech.edu)

Capacitive ultrasonic transducers have been in use for many years for various applications such as medical imaging, wireless power, sensing, and nondestructive testing. Typically, capacitive transducers require a DC bias or electret in order to operate efficiently and with high sensitivity. This makes them less desirable for applications such as wireless power transfer or energy harvesting, where a passive piezoelectric transducer may be preferred. Recently, it was shown that the requirement of a DC bias can be overcome by driving the capacitive transducer into parametric resonance

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using ultrasound (Surappa *et al.*, *Appl. Phys. Lett.* **111**(4), 043503). In this work, we present the operation of the first ever biasless, chargeless capacitive parametric ultrasonic transducer (CPUT) in air and demonstrate the utility of such a system for acoustic power transfer and sensing. Experiments performed in air show that the CPUT has an open-circuit sensitivity of more than 100 mV/Pa and is able to recover 31 μW at a distance of 10 cm from a 50 kHz ultrasonic source in the absence of a DC bias.

10:15

4aSA9. Micro-electro-mechanical system multi-resonant accelerometer for auditory prosthetics. Alison Hake and Karl Grosh (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, aehake@umich.edu)

Low usage of auditory prosthetics such as hearing aids can be attributed to several factors including appearance, stigma, acoustic feedback, and poor performance in noisy environments. Cochlear implants suffer from similar stigma; furthermore, external components of the system must be removed for sleeping, bathing, and physical exercise. Designing a completely implantable system that includes a sensor placed in the middle ear can mitigate these issues. Our approach is to use a miniature piezoelectric accelerometer to sense the vibration of the middle ear ossicles rather than employing an external or subcutaneous microphone to sense incoming sound. Results of our previous work showed the potential of this approach using a traditional single-resonance sensor. We seek to improve the low-frequency input referred noise (IRN) of the system by using a new architecture consisting of an array of piezoelectric MEMS beams with different resonant frequencies. The beams are connected electrically in a manner that increases the system sensitivity over the bandwidth of interest, thereby decreasing the IRN. Preliminary analytic studies have illustrated that 10 parallel-connected beams can improve the IRN by approximately 45 dB at 100 Hz. This method could further miniaturize sensors capable of detecting ossicular vibration from 100 Hz to 8 kHz.

10:30

4aSA10. Method for low frequency sound reflection coefficient measurements in a compact space. Xiuyuan Peng, Junfei Li, Chen Shen, Kiegan Lenihan, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27705, xp21@duke.edu)

Here, we present a method to measure the quasi-normal reflection coefficients of sound absorbing materials in a compact space. A short incident sound pulse (length < 3ms) is generated by a deconvolution method with the source speaker. Then, a stage-mounted microphone moves across the material surface and records the total (incident + scattered) sound field. By comparing the sound field with and without the presence of the sound absorbing material, the frequency-dependent reflection coefficients can be derived by extracting the corresponding frequency components from the sound pulse. Using this method, we can calculate the reflection coefficients of a 2 ft. by2 ft acoustic panel from 300 Hz to 2500 Hz within a 2 m by2 m by3 m lab space without anechoic coatings. Moreover, this method enables us to investigate the spatial inhomogeneity of the sound absorbing material by studying the amplitude/phase variation of reflection coefficients across the material surface. Compared with conventional measurement techniques for reflection/absorption coefficients, our method has the advantages of low cost, minimal requirements for the measurement environment and the ability to measure the reflection coefficients at different locations. The proposed method can be favorable for measuring reflection coefficients of two-dimensional acoustic panels/metamaterials at low frequencies.

10:45

4aSA11. Sound quality evaluation of automobile interior noise under transient and steady-state running conditions. Zhengwei Yang, Huihua Feng (Mech. Eng., Beijing Inst. of Technol., 5 South Zhongguancun St., Haidian District, Beijing 100081, China, yangzhengwei1991@126.com), and Shouwei Lu (Tech. Ctr., Baic Motor Powertrain Co., Ltd., Beijing, China)

Sound quality can reflect people's subjective auditory feelings; thus, it plays an important role in automobile interior noise evaluation in recent years. Most research focuses on steady-state running conditions. In this paper, automobile vibration and noise transfer paths were measured with the binaural transfer path analysis (BTPA) method under both transient and steady-state running conditions. Then, loudness, sharpness, roughness, and A-weighted sound pressure level were used for studying properties and differences of automobile interior noise among different running conditions. Moreover, an experiment was carried out for the subjects to mark the annoyance of all noise samples. After that, the artificial neural network was applied to create the sound quality model to assess automobile interior noise without subjective experiments. According to the scores and binaural transfer path synthesis(BTPS) results, structural improvement methods were proposed for better sound quality of the automobile.

Session 4aSP

Signal Processing in Acoustics: Emerging Techniques for Acoustic Signal Processing

Michael J. Roan, Chair
Mechanical Engineering, Virginia Tech, 111 Randolph Hall, 460 Old Turner St., Blacksburg, VA 24061

Contributed Papers

8:30

4aSP1. Non-stationary source localization using direction of arrival measurements from multiple in-air vector sensors. Miles B. Penhale and Andrew R. Barnard (Mech. Eng.-Eng. MecûÂnica, Michigan Technol. Univ., 1400 Townsend Dr., R.L. Smith MEEM Bldg., Houghton, MI 49931, mbpenhal@mtu.edu)

Vector sensors utilize a combination of pressure sensors, particle velocity sensors, or both, to determine the acoustic intensity magnitude and direction pointing toward an acoustic source. This acoustic intensity vector is often referred to as the Direction of Arrival (DOA). By combining DOA information from multiple vector sensor measurement locations, a sound source may be instantaneously localized. The majority of vector sensor research has been conducted for underwater applications. A few studies of in-air vector sensors, which utilize multiple microphones, have been conducted; however, the majority of them study stationary sound sources in a laboratory environment or non-real-world settings. The focus of this paper is to study in-air vector sensor capabilities when sensing non-stationary mechanical noise sources—specifically ground vehicles—in a non-laboratory environment where ambient noise may be present. The DOA measurements at multiple vector sensor locations are used to test the acoustic source localization potential for this method.

8:45

4aSP2. Audio and ultrasonic frequency response measurements using regularized inverse filtering. Yegor Sinelnikov (Acoust., Zebra, 126 Liberty Ave., Port Jefferson, NY 11777, ysinelnikov@yahoo.com), Andrew Boyden, Richard Lavery, and Russell Calvarese (Acoust., Zebra, Holtsville, NY)

Skyrocketing growth in digital voice communication technology prompts inclusion of ultrasonic microphones in modern smartphones. Inaudible ultrasonic waves provide safe and effective way for ambient intelligence in indoor environments. It is important to measure microphone response over combined audio and ultrasonic range. Conservative measurements based on frequency sweep present signal equalization challenge. Most ultrasonic tweeters produce narrowband ultrasonic tones, while audio speakers resonate below 1 kHz. We present a simple methodology based on emission of specially constructed noise stimulus that overcomes this limitation. The Ole Wolf 25 x16 mm 1 W 8 Ω speaker connected to B&K power amplifier delivered the stimulus. High frequency B&K type 2670 reference microphone was used to record responses at 192 kHz. Experiments were performed in two stages. First, the blue noise was emitted, its response was recorded, and new stimulus with flat spectra between 15–25 kHz was derived using regularized inverse filtering technique. Second, constructed stimulus was emitted and recorded by reference microphone and device under test. The frequency response was derived and found in good agreement with conventional stepped sweep measurements.

9:00

4aSP3. Classification of inter-floor noise type/position via supervised learning. Hwiyong Choi, Haesang Yang, Seungjun Lee, and Woojae Seong (Naval Architecture and Ocean Eng., Seoul National Univ., 1, Gwanak-ro, Bldg. 34, Rm. 305, Gwanak-gu 08826, Seoul, South Korea, its_me_chy@snu.ac.kr)

This work presents noise type/position classification of various inter-floor noises generated in a building which is a serious conflict issue in apartment complexes. For this study, a collection of inter-floor noise dataset is recorded with a single microphone. Noise types/positions are selected based on a report by the Floor management Center under Korea Environmental Corporation. Using a convolutional neural networks based classifier, the inter-floor noise signals converted to log-scaled Mel-spectrograms are classified into noise types or positions. Also, our model is evaluated on a standard environmental sound dataset ESC-50 to show extensibility on environmental sound classification.

9:15

4aSP4. Ultrasonic communications for real-time video-rate data transmission through tissue. Gizem Tabak, Michael L. Oelze (Univ. of Illinois at Urbana-Champaign, 1308 W Main St., 119 Coordinated Sci. Lab., Urbana, IL 61801, tabak2@illinois.edu), and Andrew Singer (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Wireless implanted medical devices (IMDs), which communicate data wirelessly from sensors within the body to a receiver outside of the body, are poised to be significant contributors to medical diagnostics and treatment procedures. Currently, radio frequency (RF) electromagnetic waves are the most frequently used communication method for wireless IMDs. However, high attenuation of RF in the body and strict regulations on the RF frequency spectrum limit the data rates to 267 kbps for *in vivo* applications. Considering standard definition video streaming requires 1.2 Mbps, and HD requires greater than 3 Mbps, it is not possible to use RF communication, for example, in applications that require real-time video transmission and, possibly, intervention such as real-time video capsule endoscopy. In our work, we use ultrasonic waves to communicate through tissue at video-capable data rates (>1.2 Mbps). Previously, we demonstrated a 4 Mbps data rate with BER less than 1 10^{-4}through beef liver using small, 2-mm biocompatible transducers at 1.3 MHz. In this study, we will investigate the effects of target motion on data rates and demonstrate real-time communication links *in situ* in a dead animal and *in vivo*.

9:30

4aSP5. Personalizing head-related transfer functions using anthropometric measurements by combining two machine-learning models. MingYang Lee, Martin S. Lawless, and Melody Baglione (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, 41 Cooper Sq., New York, NY 10003, lee11@cooper.edu)

Virtual reality (VR) requires rendering accurate head-related transfer functions (HRTF) to ensure a realistic and immersive virtual auditory space.

An HRTF characterizes how each ear receives sound from a certain location in space based on the shape of the head, torso, and pinnae, and provides a unique head-related impulse response (HRIR) for each given source location. Since HRTFs are person-specific and difficult to measure, recent research has utilized pre-existing HRTF databases and anthropometric measurements to generate personalized HRTFs with machine learning algorithms. This study investigates a personalization method that estimates the shape of each ear's HRIR and interaural time differences (ITD) between the two ears in separate models. In the proposed method, the shape of the HRIR is estimated with an artificial neural network (ANN) trained with time-aligned HRIRs from the CIPIC database, eliminating between-subject timing differences. A regression tree is used to estimate the ITDs, which are integer sample delays between the left and right ears. A localization test with a VR headset was conducted to evaluate the perceptual accuracy of the personalized HRTFs. Subjects completed the test with both a pre-selected average HRTF and their personalized HRTF to compare localization errors between the two conditions.

9:45–10:00 Break

10:00

4aSP6. Improving autonomous vehicle safety—The use of convolutional neural networks for the detection of warning sounds. Ethan Wagner and Eoin A. King (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

In cities across the world everyday, people use and process acoustic alerts to safely interact in and amongst traffic. With the advent of autonomous vehicles (AVs), the manner in which these new vehicles can use these acoustic cues to supplement their decision making process is unclear. This will be especially important during the prolonged period of mixed vehicles sharing the road. One solution may lie in the advancement of machine learning techniques; it has become possible to "teach" a machine (or a vehicle) to recognize certain sounds. This paper reports on an ongoing project with the objective of identifying emergency vehicles sirens in traffic and alerting the vehicle to take rapid evasive action. In particular, we report on the use of a deep layer Convolutional Neural Network (CNN) trained to recognize emergency sirens. We retrained a CNN (AlexNet) to recognize sirens in real time. To utilize this network, samples from the ESC-50 dataset for environmental sound classification were processed and each converted to a spectrogram. This CNN can be used in conjunction with a microphone array to accurately recognize sirens in traffic and identify the direction from which the emergency vehicle is approaching.

10:15

4aSP7. A review of techniques for ultrasonic indoor localization systems. Joaquín Aparicio (Dept. of Informatics, Univ. of Oslo, Gaustadalléen 23B, Ole-Johan Dahls hus, Oslo 0851, Norway, joaqapar@ifi.uio.no), Fernando J. Álvarez (Sensory Systems Res. Group, Univ. of Extremadura, Badajoz, Spain), Álvaro Hernández (Electronics Dept., Univ. of Alcalá, Alcalá de Henares, Spain), and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Oslo, Norway)

Accurate localization in indoor environments is crucial for the correct operation of location-aware and augmented reality applications, indoor navigation, and inventory management, among others. Magnetic, radiofrequency and inertial navigation systems typically provide room or meter-level accuracy. Despite being the most widely used, they are affected by error drifts or changes in the environment where they operate, and their accuracy is a drawback for certain applications, such as navigation. Optical-based systems provide better accuracy, but they can be expensive, and they are not privacy-oriented. Ultrasonic positioning systems can also give room-level accuracy, as acoustic propagation is contained within the room walls, helping resolve room or floor-level ambiguities of radio systems. They can even achieve centimeter-level accuracy and ensure privacy, while being low cost. These properties highlight acoustics as a versatile technology for different indoor localization applications, as stated by the research published almost over the last three decades. In this work, the operating principles of the different techniques employed by acoustic positioning systems are reviewed, covering narrowband and wideband systems (including the differences between coded and uncoded transmissions), fingerprinting, and the most recent systems based on machine learning.

10:30

4aSP8. Extending bandwidth for sound power measurements. Michael C. Mortenson, Suzanna Gilbert, Tracianne B. Neilsen, Kent L. Gee, and Scott D. Sommerfeldt (Brigham Young Univ., N 283 ESC, Provo, UT 84602, michaelcmort@byu.edu)

Sound power is often measured using the intensity-based engineering standard ANSI S12.12-1992. Traditional methods for intensity-based sound power estimation are limited in bandwidth at low frequencies by phase mismatch between microphones and at high frequencies by microphone spacing—with errors occurring well below the spatial Nyquist frequency. The PAGE (Phase and Amplitude Gradient Estimation) method has been used to extend the bandwidth of intensity calculations [Gee *et al.*, *J. Acoust. Soc. Am.* **141**(4), EL357–EL362 (2017)]. This paper examines the efficacy of the PAGE method to overcome bandwidth limitations in estimating sound power. Specifically, the sound fields from three sources—a blender, a vacuum cleaner, and a dodecahedron speaker—were measured according to ANSI S12.12-1992. The sound power was computed for each source using both the traditional and PAGE methods. The resulting intensity-based sound power estimates are compared against sound power measurements obtained according to the scientific-grade ISO 3741:1999 standard. The PAGE method increases the bandwidth over which reliable estimates are achievable for intensity-based sound power estimates, even exceeding the spatial Nyquist frequency when phase unwrapping is successful. Thus, using existing equipment, industry professionals can extend the bandwidth of sound power estimates with the PAGE method. [Work supported by NSF.]

10:45

4aSP9. On the viability of the complex-intensity-based near-field acoustical holography method. Caleb B. Goates, Scott D. Sommerfeldt, Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com)

Because of the instrumentation difficulties of Fourier acoustical holography, it is desirable to find a holography method that does not require reference signals during array scanning. Among the methods that have been investigated to remove the need for references are those based on acoustic intensity measurements, including the complex-intensity-based near-field acoustical holography (CIBNAH) method [A. Nejade, *J. Sound Vib.* **333**(16), 3598–3608 (2014); *Appl. Acoust.* **116**, 348–356 (2017)]. The CIBNAH method has previously been applied to simple contrived sources and real-world machinery but has not been verified using analytical source models. This work shows the application of CIBNAH to an analytical model of a simply-supported plate, revealing key shortcomings of the method. The theory behind CIBNAH is discussed in light of these shortcomings. It is shown that while CIBNAH may be useful for finding radiation hot spots, it is not an adequate method to overcome the need for references in scan-based acoustical holography. [Work supported by NSF.]

11:00

4aSP10. Employing braid theory to disentangle overlapping oceanic events in the shallow water acoustic channel. Ananya Sen Gupta and Ryan McCarthy (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sci., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

The shallow water acoustic channel is well-known to exhibit rapid temporal fluctuations due to multipath effects, fluid motion as well as focusing events in the ocean. These events often overlap temporally thus contributing in unpredictable ways to the channel delay spread. We will present recent results from employing geometric signal processing techniques to shallow water channel estimation and will introduce new ways to identify and disambiguate overlapping oceanic phenomena within the channel delay spread. Specifically, we will explore how geometric braid theory and related operations can be employed to disentangle overlapping channel braids, and explore the efficacy of such signal processing techniques across diverse channel representations. Finally, we will discuss the importance of capturing oceanic phenomena at the appropriate resolution of time, frequency, and sparsity and the trade-offs in accuracy involved based on the chosen channel representation(s).

Session 4aUW

Underwater Acoustics, Physical Acoustics, Structural Acoustics and Vibration, Acoustical Oceanography, Signal Processing in Acoustics, and Computational Acoustics: Uncertainty in Propagation Prediction

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D. Keith Wilson, Cochair
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Invited Papers

8:20

4aUW1. Uncertainty quantification for right-sizing computational models of sound propagation in the atmospheric boundary layer. Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., 590 Holloway Rd., Annapolis, MD 21402, pettitcl@usna.edu), D. K. Wilson, and Carl R. Hart (U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH)

Comprehensive modeling of sound propagation through the atmospheric boundary layer is viewed as a judicious combination of accurate computational mechanics models and uncertainty quantification (UQ) methods. The role of numerical models is to represent nominally deterministic phenomena, e.g., geometrical spreading, ground interactions, refraction by mean gradients of wind and temperature. The role of UQ is to characterize the consequences of fundamentally non-deterministic and imprecisely known factors that affect propagation, e.g., turbulence in the atmospheric boundary layer, complex terrain features, and overly sparse spatio-temporal sampling of propagation parameters. High-fidelity wave propagation mechanics cannot compensate for inherent randomness in the environment and insufficient data on the parameters. When uncertainty is significant, the computational cost of high-fidelity models might be better invested in more ensemble simulations with medium-fidelity models and quantifying the payoff from more data about the environment. Work in recent years along three thrusts to enable this form of comprehensive modeling is reviewed: (1) Surrogate modeling based on cluster-weighted models, which are a type of probabilistic generative model, and on statistical learning methods, (2) global sensitivity analysis for assessing the importance of model parameters, and (3) a computational mechanics error budget for rationally analyzing the importance of various sources of uncertainty.

8:40

4aUW2. Statistics of acoustic waves propagating through turbulent media. Philippe Blanc-Benon (Ecole Centrale de Lyon, Université de Lyon, LMFA UMR CNRS 5509, 36 Ave. Guy de Collongue, Ecully 69134 Cedex, France, Philippe.blanc-benon@ec-lyon.fr)

Propagation of acoustic waves through atmospheric turbulence is relevant to different problems : outdoor sound propagation, blast waves generated from explosions or gunshots, propagation of sonic booms. While propagating in turbulent air, acoustic waves are distorted by the combined effects of diffraction and scattering induced by atmospheric inhomogeneities. Accurately controlled experiments are needed to validate theoretical models for sound propagation in inhomogeneous media. In this paper, probability distribution functions will be presented for linear and nonlinear acoustic wave propagation through thermal or kinematic turbulence. Experimental data will be compared with numerical simulation using parabolic approaches. [Work supported by the Labex CeLyA of Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ ANR-11- IDEX-0007).]

Contributed Paper

9:00

4aUW3. Effect of uncertainty in meteorological conditions on aircraft noise levels. Harshal Prakash Patankar and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hpp103@psu.edu)

To comply with noise regulations and to plan infrastructure around airports, there is a need to accurately predict aircraft noise levels. Even when high fidelity noise propagation models are used, the accuracy of noise level predictions can be affected by uncertainty in the input parameters. This work looks at the change in SPLs due to these uncertainties. To incorporate the effect of meteorological uncertainties, the methodology presented in Wilson *et al*. [*JASA*(2014)] is extended to the geometry of aircraft noise propagation. To allow for faster computations, a simplified version of the NORD 2000 noise propagation method is used. This work explores the effect of individual meteorological uncertainties in the propagation path (such as temperature profile and wind profile) as well as their combined effect on the SPLs for terminal area aircraft altitudes. [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, project 40 through FAA Award No. 13-C-AJFE-PSU under the supervision of Hua He. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

Invited Papers

9:15

4aUW4. Methods for including uncertainty in modeling and simulation of acoustic propagation in ocean waveguides. Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Two main classes of uncertainty are associated with modeling and simulation of acoustic fields in ocean waveguides. These classes, generally known as aleatory and epistemic uncertainties, represent opposite ends of a spectrum of possible types of imperfect knowledge concerning the system. Aleatory uncertainty can be interpreted physically as natural ocean variability which is typically characterized by probability density functions, provided sufficient information is available to justify their specification. On the other hand, epistemic uncertainty is associated with incomplete scientific knowledge concerning some aspect(s) of the system under analysis. While epistemic uncertainty can be reduced by the inclusion of additional information, e.g., data refinements, acquisition of additional data or more realistic modeling of the system, aleatory uncertainty cannot be eliminated or reduced because it is judged by the modeler to be inherent in the structure of the system. Both types of uncertainty are discussed, with an emphasis on the aleatory contribution based on stochastic basis expansions. Tradeoffs with Monte-Carlo and Bayesian methods are also considered. When density functions are not available, weaker inferential estimates based on epistemic approaches are appropriate and a hybrid aleatory-epistemic framework is outlined for treating these situations. [Work supported by the Office of Naval Research.]

9:35

4aUW5. Strategies for linking ocean state uncertainty to acoustic uncertainty. Timothy F. Duda, Weifeng G. Zhang, Ying-Tsong Lin, and Arthur Newhall (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Estimating uncertainty of underwater sound fields caused by partially known environmental conditions is a broad topic with many branches because acoustic fields have many characteristics and many descriptors. Specific field characteristics, each uncertain, influence sound use in specific ways. Multiple descriptors (parameters) of signals of interest, and noise, need to be adequately known to examine use scenarios. This is also true for field simulation, processor simulation, field uncertainty simulation, and processor uncertainty simulation. Parameters should be prioritized for efficient quantification of uncertainty. For example, spatial coherence uncertainty applies to array processing but possibly not to single-sensor processing. Linking environmental uncertainty to field uncertainty, then to task performance uncertainty suffers from the many degrees of freedom present in the environment, and the interconnected effects of the many variable environmental parameters. Here, linkage frameworks for sound within internal waves propagating in variable conditions are examined. Methods appropriate for deterministically defined wave groups are explored, as well as statistically described wave fields. First, wave parameters and parameter uncertainty are specified, then effects on the sound parameters are estimated, as well as derived quantities like probability of detection and direction of arrival. Canyon environments that we have studied with models provide one test bed.

9:55–10:10 Break

10:10

4aUW6. Bayesian calibration of underwater propagation models under uncertainty. Ralph T. Muehleisen, S H. Swift, and Qi Li (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg 221, Lemont, IL 60439, rmuehleisen@anl.gov)

Propagation models are notorious for the uncertainty of important parameters such as source strength, speed of sound profiles, and reflecting surface profiles. In many cases, one calibrates a model to measured data (e.g., sound levels or transmission loss) for the purposes of estimating these model parameters, i.e., for inverse modeling. Bayesian calibration methods have been developed that are extremely useful for calibration of models where parameters have high levels of uncertainty and problems may be under or over determined. The Kennedy and O'Hagen framework which uses a Gaussian process surrogate model to replace the model under calibration is especially useful when the underlying model is computationally expensive, and so, it may be difficult to apply many optimization based calibration methods. In this talk, we describe the application of the Kennedy and O'Hagen Bayesian Calibration framework to the calibration of an underwater ray tracing propagation model. The source strength and parameters for the sound speed profile are considered as highly uncertain. The Bayesian calibration technique is shown to improve model prediction and reduce the uncertainty of the unknown propagation parameters.

10:30

4aUW7. Quantifying ocean acoustic propagation uncertainty due to environmental uncertainty from ship-noise inversion. Stan E. Dosso (School of Earth & Ocean Sci, Univ of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca) and Dag Tollefsen (Defence Systems Div., Norwegian Defence Res. Establishment (FFI), Horten, Norway)

This paper considers the uncertainty in underwater acoustic propagation predictions that results from uncertainty in oceanic environmental parameters (water column and seabed) as estimated from acoustic inversion methods. The approach is general but is considered here for the inversion of acoustic fields from ships-of-opportunity, which represents a convenient and unobtrusive approach to environmental assessment, but with significant uncertainties for some environmental parameters. In this work, environmental uncertainties are quantified using trans-dimensional (trans-D) Bayesian inversion based on reversible-jump Markov-chain Monte Carlo sampling. Trans-D inversion considers the number of sediment layers in the seabed and/or the number of points defining the water-column sound-speed profile as unknown parameters, sampled probabilistically in the inversion. This approach numerically characterizes the posterior probability density of the environment using a large ensemble of dependent random samples of the model. Predicting acoustic propagation for a much-smaller, randomly-chosen subset of these model samples transforms environmental uncertainty to propagation uncertainty, which can be characterized spatially with standard statistical measures (standard deviation, credibility intervals, etc.). The approach is illustrated using simulations and noise from a large commercial ship recorded on an a horizontal array of hydrophones as part of the 2017 Seabed Characterization Experiment (SBCEX17) at the New England Mud Patch.

10:50

4aUW8. Variability of the modal response of a shallow water acoustic waveguide in the presence of uncertain sediment properties. Sheri Martinelli, Andrew S. Wixom (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, Mailstop 3230D, State College, PA 16804-0030, slm77@psu.edu), Mark Langhirt (Graduate Program in Acoust., The Penn State Univ., University Park, PA), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

Physical properties of the seabed comprise an important input set for physics-based modeling of sound propagation in littoral environments. Unfortunately this knowledge is often incomplete due to inherent space-time variability, yet deterministic models are still very much the standard for complex environments. Normal mode decomposition provides a well-established and understood framework for the study of underwater propagation. This work applies a generalized polynomial chaos expansion with stochastic collocation to propagate uncertain variables through a normal mode model, thus constructing expressions for the transmission loss (TL) and mode shapes themselves in terms of the input random variables. The goal of this work is to demonstrate the impact of imperfect knowledge of physical parameters on a deterministic computational model of the in-water acoustic field, and further study how well important information about model performance is captured by considering only first and second moments of the output distribution. An emphasis on material properties and geometry of sediment layering in the ocean bottom serves to isolate the effects of sediment uncertainty. Such a study can provide guidance for the use of deterministic models in performance prediction and has consequences for geo-acoustic inversion.

11:10

4aUW9. Three-dimensional underwater sound pressure sensitivity in oceanic environments. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Physical oceanographic processes, marine geological features, sub-bottom geoacoustic structure, sea surface disturbances, etc., can either individually or jointly affect underwater sound propagation in the ocean and cause significant temporal and spatial variability in the sound pressure field. The primary goal of this study is to develop a numerical scheme to determine the sound pressure sensitivity in response to variations of index of refraction due to changes of environmental conditions. This sensitivity analysis is an exention of the Born approximation which assumes perturbations at infinitesimal points. To handle disturbance within a finite volume, an improved sensitivity kernel is derived from a higher-order parabolic-equation (PE) approximation. With this sensitivity kernel, we can analyze the spatial distribution and the temporal evolution of the acoustic sensitivity field in complex oceanic environments. This paper will present numerical examples of three-dimensional (3D) sound propagation in continental slopes, submarine canyons, and nonlinear internal wave fields. Discussions on other applications, including uncertainty quantification of transmission loss prediction and adjoint models for 3D acoustic inversions, will also be provided. [Work supported by the Office of Naval Research.]

Contributed Papers

11:30

4aUW10. Machine learning methods for estimating probability density functions of transmission loss: Robustness to source frequency and depth. Brandon M. Lee and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu)

Predicted values of transmission loss (TL) in ocean environments are sensitive to environmental uncertainties. The resulting predicted-TL uncertainty can be quantified via the probability density function (PDF) of TL. Monte Carlo methods can determine the PDF of TL but typically require thousands of field calculations, making them inappropriate for real-time applications. Thus, a variety of alternative techniques based on polynomial chaos, field shifting, modal propagation in ocean waveguides, and spatial variations of TL near the point(s) of interest have been proposed. Recently, an approach to estimating the PDF of TL based on nominal TL, ocean environmental parameters, and machine learning was found to have a success rate of 95% with constant source depth (91 m) and frequency (100 Hz) when tested on 657,775 receiver locations within 100 randomly selected ocean environments. This presentation describes an extension of this approach and its success predicting the PDF of TL for different source depths and frequencies for ranges up to 100 km. This increase in the size of the parameter space furthers the need for a sophisticated method of choosing training examples. Such a method is proposed, and its performance is compared to that of prior techniques. [Work supported by ONR.]

11:45

4aUW11. Model-data comparison of sound propagation in a glacierized fjord with a variable ice top-boundary layer. Matthew C. Zeh (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 204 East Dean Keeton, Stop C2200, Austin, TX 78712-1591, mzeh@utexas.edu), Oskar Glowacki (Marine Physical Lab, Scripps Inst. of Oceanogr., Warsaw, Poland), Grant B. Deane (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA), Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Erin C. Pettit (College of Earth, Ocean, and Atmospheric Sci., Oregon State Univ., Fairbanks, AL), and Preston S. Wilson (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Transmission loss measurements were conducted in the meltwater-modified surface layer near Hansbreen Glacier in Hornsund Fjord in southwestern Svalbard in September 2017 [Deane and Glowacki, *JASA* **143**, 1711 (2018)]. An m-sequence source signal (149 dB re 1 μPa, 11 kHz carrier frequency) was tethered at 7 m depth to a boat drifting from 0 to 200 m. This signal was received by two Hitech HTI-96 hydrophones at 8 and 17 m depth deployed from a stationary boat anchored 500 m from the glacier. Within this environment, and typical for a glacierized fjord, regular calving events contributed to an ice mélange top boundary layer with larger icebergs occasionally obstructing the signal transmission path. The propagation environment was upward refracting, causing propagation sound to repeatedly reflect from the surface layer. A ray-based approach was applied to model the measured data. The variability of the top boundary was included in the model by incorporating surface scattering and inserting icebergs. Comparisons between several increasingly complex iterations of this model with the collected data will be presented. [Work supported by the NDSEG Fellowship, ONR Grant Nos. N00014-17-1-2633 and N00014-14-1-0213, and the Polish National Science Centre Grant No. 2013/11/N/ST10/01729.]

Session 4pAA

Architectural Acoustics and Signal Processing in Acoustics: Room Acoustics Modeling and Auralization

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Invited Papers

1:15

4pAA1. Reverberation time and audibility in phased geometrical acoustics using plane or spherical wave reflection coefficients. Matthew Boucher (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681-2199, matthew.a.boucher@nasa.gov), Monika Rychtarikova (Faculty of Architecture, KU Leuven, Gent, Belgium), Lukas Zelem (Faculty of Civil Eng., Dept. of Architecture, STU Bratislava, Bratislava, Slovakia), Bert Pluymers, and Wim Desmet (Mech. Eng., Div. PMA, KU Leuven, Heverlee, Belgium)

In acoustical spaces, room acoustic parameters are often predicted using energy-based geometrical acoustics. For smaller rooms, interference among coherent reflections is taken into account by phased geometrical acoustics, which improves results for lower frequencies. The use of a spherical wave reflection coefficient improves the results further, yet the impact on room acoustic parameters is not fully known. This work focuses on the differences in predicted reverberation time when using plane or spherical wave reflection coefficients. The differences are analyzed for a variety of boundary conditions, including non-uniform distributions of absorption, in medium-sized rooms using a phased image source model. Since calculated differences are greater than the conventional just-noticeable-difference of 5% for reverberation time, a laboratory listening test is performed to confirm audibility of the modeled differences. Two narrow band noise stimuli (octave bands with central frequencies of 125 and 250 Hz) with a duration of 1 s were used for comparisons of 18 acoustic scenarios by means of a three-alternative forced choice method (3AFC). More than half of the listeners could hear the differences in all 36 cases. Statistically significant results (chi-squared test was used) were found in two thirds of the cases, corresponding to those with longer reverberation times.

1:35

4pAA2. Diffraction simulation from wedges to finite-sized plates based on the physical theory of diffraction. Ning Xiang, Anthony Savino (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu), and Aleksandra Rozynova (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Cambridge, MA)

Efficient predictions of sound diffraction around objects are of critical significance in room-acoustic simulations. An advanced diffraction theory has recently been investigated for potential applications in room acoustics for some semi-infinite, canonical wedges and for finite-sized rectangular plates [Rozynova and Xiang, *J. Acoust. Soc. Am.* **144** (to be published)]. The physical theory of diffraction (PTD) still relies on both geometrical and physical principles, yet emphasizes the physical one. Important features of the PTD approach are its computational efficiency and the high degree of accuracy for the diffracted sound field. This paper reviews the fringe field predictions of canonical semi-infinite wedges and further discusses solutions of diffraction problems on finite, rigid rectangular plates. The PTD is applied to approximate the solutions of a finite-sized, rigid rectangular plate that achieves high numerical efficiency. The PTD simulation allows sound diffraction contributions to be determined independently from two pairs of edges of the rigid plate, while ignoring the edge waves around the corner in far-field. This paper uses numerical implementations of the PTD predictions to demonstrate the simulation efficiency of the PTD in finite-sized objects. The numerical simulations are also validated by some preliminary experimental results carried out using an acoustic goniometer.

1:55

4pAA3. Individualization of head-related transfer functions using sparse representation approach. Zeng Xiangyang, Wang Lei, Lu Dongdong (Northwestern PolyTech. Univ., Xi'an, Shaanxi, 710070, Xi'an 710070, China, zenggxy@nwpu.edu.cn), and Huaizhen Cai (Dept. of Psych., Univ. at Buffalo, Buffalo, NY)

The individualization of Head-Related Transfer Functions (HRTFs) is an important issue for enhancing the performance of binaural auralization. In this paper, the HRTFs and anthropometric parameters of Chinese people were measured and analyzed. A sparse representation approach was suggested to synthesize individualized HRTFs with selected anthropometric features. The approach was compared with two other methods proposed in previous studies by comparing the spectrum distortion of each method for objective evaluation. Then, subjective experiments were conducted to investigate the performance of the optimized HRTFs in binaural localization. The evaluation results show that the proposed HRTFs individualization approach has smaller spectrum distortion and better localization performance than that of the reference methods.

2:15

4pAA4. Current trends in binaural auralization of microphone array recordings. Jens Ahrens, Carl Andersson (Chalmers Univ. of Technol., Sven Hultins Gata 8a, Gothenburg 412 58, Sweden, carl.andersson@chalmers.se), and Hannes Helmholz (Chalmers Univ. of Technol., Gotheburg, Sweden)

Many approaches for the capture and auralization of real acoustic spaces have been proposed over the past century. Limited spatial resolution on the capture side has typically been the factor that caused compromises in the achievable authenticity of the auralization. Recent advancements in the field of microphone arrays provide new perspective particularly for headphone-based auralization. It has been shown that head-tracked binaural auralization of the data captured by a bowling-ball-sized spherical array of around 90 microphones allows for creating signals at the ears of the listener that are perceptually almost indistinguishable from the ear signals that arise in the original space. Promising results have also been obtained based on smaller arrays with fewer microphones. In the present contribution, we provide an overview of the current activities in the research community and demonstrate the latest advancements and remaining challenges.

2:35

4pAA5. Simulation of a coupled room scenario based on geometrical acoustics simulation models. Lukas Aspöck and Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

As part of the most recent room acoustical simulation Round Robin, a coupled room scenario, consisting of a laboratory room and a reverberation chamber, is investigated. The evaluation of the participants' results, all using geometrical acoustics based simulation models, however, showed that the measured double slope of this scenario could not be matched by any of the six algorithms. In addition to the presentation of the measured and simulated results for this scenario, this work discusses the role the input data, especially the applied absorption coefficients, and the configuration of the simulation model. Eventually, additional results for one simulation model are presented to demonstrate the options and the limitations regarding the simulation of coupled volumes using geometrical acoustics models.

2:55–3:10 Break

3:10

4pAA6. Audibility of dispersion error in room acoustic finite-difference time-domain simulation in the presence of a single early reflection. Jukka Saarelma (Dept. of Comput. Sci., School of Sci., Aalto Univ., P.O. Box 13000, FI-00076 Aalto, Finland, jukka.saarelma@aalto.fi)

Finite-difference time-domain method has gained increasing interest for room acoustic prediction use. A well-known limitation of the method is a frequency and direction dependent dispersion error. In this study the audibility of dispersion error in the presence of a single surface reflection is measured. The threshold is measured for three different distance conditions with a fixed reflection arrival azimuth angle of 54.7 deg. The error is placed either in the direct path, or in the reflection path. Additionally, a qualitative follow-up experiment to evaluate how the measured thresholds reflect the audibility of error in short room responses is carried out. The results indicate that the threshold varies depending whether the error is in the direct path or in the reflection path. For transient signals, the threshold is higher when the error is located in the direct path, where as for speech signal the threshold is higher when it is located in the reflection path. Evidence is found that the error is detectable in rendered room responses at the measured threshold levels.

3:30

4pAA7. Benchmark analytical solutions for steady-state high frequency broadband sound fields in three rectangular enclosures. David Raudales (Physical Acoust. Branch Code 7130, Naval Res. Lab., 1420 W Abingdon Dr. Apt. 126, Alexandria, VA 22314, david.raudales@nrl.navy.mil), Donald B. Bliss (Mech. Eng., Duke Univ., Durham, NC), Krista Michalis (Mech. Eng., Duke Univ., Arlington, VA), Jerry W. Rouse (Mech. Eng., Duke Univ., Albuquerque, NM), and Linda P. Franzoni (Mech. Eng., Duke Univ., Durham, NC)

Analytical solutions are presented for interior broadband sound fields in three rectangular enclosures with absorption applied on the floor and ceiling, rigid sidewalls, and a vertically oriented dipole source. The solutions are intended to serve as benchmarks that can be used to assess the performance of broadband techniques, particularly energy-based methods, in a relatively straightforward configuration with precisely specified boundary conditions. A broadband Helmholtz solution is developed using a frequency-by-frequency modal approach to determine the exact band averaged mean-square pressures along spatial trajectories within each enclosure. Due to the specific choice of enclosure configuration and absorption distribution, an approximate specular solution can be obtained through a summation of uncorrelated image sources. Comparisons between the band averaged Helmholtz solution and the uncorrelated image solution reveal excellent agreement for a wide range of absorption levels and improve the understanding of correlation effects in broadband sound fields. A boundary element solution with diffuse boundaries is also presented which produces consistently higher mean-square pressures in comparison with the specular solution, emphasizing the careful attention that must be placed on correctly modeling reflecting boundaries and demonstrating the errors that can result from assuming a Lambertian surface.

3:50

4pAA8. Real-time auralization of a talker's own voice in virtual rooms. Jennifer Whiting, Timothy W. Leishman, and Traci Neilsen (Brigham Young Univ., 243 Eyring Sci. Cente, Provo, UT 84602, jkwhiting@outlook.com)

While much has been done in the field of sound auralization in virtual rooms, the problem of hearing one's own voice in these environments has received less attention. A robust and feasible system for real-time auralization of talkers who are also listeners is needed. To address this requirement, a real-time convolution system (RTCS) was designed with the specific goal of "placing" a talker/listener in virtual acoustic environments. This system necessitated the development of several tools and methods. Oral-binaural room impulse responses were measured and characterized for a variety of room. The RTCS improved on past systems, in part through the derivation and inclusion of compensation filters, which corrected the linear auditory distortions of the RTCS components. Objective measures in the time- and frequency-domains were developed to assess the validity of the system. A jury-based listening study also indicated that RTCS users could speak and listen to their own voices in the virtual acoustic environments in a natural manner.

4:10

4pAA9. Diffusion equation modeling in sound field analysis of disproportionate rooms and coupled spaces. Zühre Sü Gül (Architecture, Bilkent Univ., METU Technopolis KOSGEB-TEKMER No. 112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com)

This study discusses alternative models and methods to be applied in room acoustics estimations for specific room types including disproportionate rooms and rooms with coupled volumes. A recent prediction method, namely, diffusion equation model (DEM) in room acoustic applications is utilized in the methodology, and this method is compared to common models as of statistical theory, image-source or ray-tracing techniques. Both long enclosures as of subway stations and coupled volumes as of multi-domain monumental structures are special cases with specific interior sound fields. Statistical theory is not always reliable in such extraordinary room forms, while ray tracing may tend to over or under estimate certain acoustical parameters. Thus, the application of DEM in a finite element scheme for detailed sound energy decay and sound flow analysis are held over some case structures. The results are compared to field tested data and ray-tracing solutions. Pros and cons of DEM in comparison to different methods are searched in detail considering the efficiency in visualization, computational speed, and reliability of acoustical parameter results for specific room shapes.

4:30

4pAA10. Auralization of virtual concerts: A subjective evaluation comparing binaural and ambisonic rendering. David Thery (CHM, LIMSI-CNRS, Rue John Von Neumann, Orsay 75005, France, david.thery@limsi.fr)

Auralization renderings have reached a sufficient level of maturity that simulated auralizations can be comparable to measured ones. These auralizations can be rendered over a variety of sound systems, potentially combined with a visual model through VR interfaces. This study presents a perceptual evaluation of auralizations of a small ensemble virtual concert rendering, comparing a tracked binaural rendering to 2nd order Ambisonic rendering over a 32 loudspeaker array. The geometrical acoustic model of several actual performance spaces were created and then calibrated using *in situ* omni-directional room impulse response measurements. The performance stimuli consisted of 3 extracts of jazz anechoic recordings comprising trios and quartet ensembles, augmented by three-dimensional visual point-clouds of the musicians playing on stage. Participants of the listening test included a range of listening expertise level (acousticians, architects, students). Several room acoustical parameters were evaluated between rendering systems, seating positions, and rooms.

4:50

4pAA11. Discrete material optimization for wave-based acoustic design. Nicolas Morales (The Dept. of Comput. Sci., 201 S. Columbia St., Chapel Hill, NC 27599-3175, nmorales@cs.unc.edu) and Dinesh Manocha (Dept. of Comput. Sci., Univ. of Maryland, College Park, MD)

The problem of automatic design of acoustic spaces is prevalent in architecture and room acoustics. We present a novel algorithm to automatically compute the optimal materials of large architectural spaces. Our method uses discrete optimization techniques to determine the best material configuration for desired acoustic properties of a structure, while taking into account properties of real-world materials. An efficient acoustic wave solver is used to accurately compute the acoustic impulse responses that drive the optimization process. Our method is tested on various computer representations of real-world scenes where we show how new material characteristics can be computed to improve the scene's strength, clarity, and reverberation time.

5:10

4pAA12. Empirical evaluation of in-field, binaural record and playback reproduction. William Neale and Toby Terpstra (Visualization, Kineticorp, 6070 Greenwood Plaza Blvd., Ste. 200, Greenwood Village, CO 80111, wneale@kineticorp.com)

This research evaluates a methodology for calibrating in field, sounds for playback in a separate, interior environment. The ability to record sounds in the field and playback them accurately in a different environment is useful when the end user, or listener cannot be present at the location where the sound is being produced live. In forensics, for example, an expert or juror may need to evaluate an acoustic or auditory issue but not have access to the site where the sound is produced. The methodology presented here utilizes a binaural microphone where participants in the field, listen to a physical sound, and calibrate the binaural microphone by adjusting recording levels until the sound heard in their headphones matches the sound being produced live in the field. A second group of participants are presented with the same live physical sound but in an interior environment. Using the same setup, these participants compare the live sound with the recorded and reproduced sound calibrated by the in-field participants. Participants in this second group empirically evaluate the similarity of the reproduced sound to the live physical sound in the interior environment.

Session 4pBAa

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Inverse Problems in Biomedical Ultrasound II

T. Douglas Mast, Cochair
Biomedical Engineering, University of Cincinnati, 3938 Cardiovascular Research Center, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Kang Kim, Cochair
Medicine, University of Pittsburgh, 950 Scaife Hall, 3550 Terrace Street, Pittsburgh, PA 15261

Contributed Papers

1:00

4pBAa1. Photoacoustic tomography in a clinical linear accelerator for quantitative radiation dosimetry. David A. Johnstone (Radiation Oncology, Univ. of Cincinnati, 3960 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, johnstdt@mail.uc.edu), Michael T. Cox (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Dan Ionascu, Michael A. Lamba (Radiation Oncology, Univ. of Cincinnati, Cincinnati, OH), Charles L. Dumoulin (Imaging Res. Ctr., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Cancer is the second leading cause of death in the United States. Approximately half of all cancer patients receive radiation therapy, in which linear accelerators are used to deliver high doses of x-ray radiation to tumors, inducing cell death. X-ray energy deposition causes pressure changes that produce acoustic signals due to the photoacoustic effect. Here, clinical x-ray beams were directed at test objects made of antimonial lead and other metallic materials within a water tank. Photoacoustic signals were measured using a calibrated broadband hydrophone and validated using simulations in k-Wave. Linear and two-dimensional synthetic apertures were formed by mechanically scanning the x-ray source and test object within a single plane. Tomographic images of test objects, reconstructed from measured photoacoustic signals, show good agreement with object geometry. X-ray doses incurred by the test objects are mapped based on the reconstructed acoustic pressure sources and Grüneisen parameter of the material employed. Potential applications to *in vivo* dosimetry for x-ray and proton therapy, potentially enabling safer and more effective treatments, are discussed.

1:15

4pBAa2. Comparisons of inverse and forward problem approaches to elastography. Siavash Ghavami, Saba Adabi (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., 200 First St. SW, Rochester, MN 55902, roudsari.seyed@mayo.edu), Olalekan Babaniyi (Civil and Environ. Eng., Duke Univ., Durham, NC), Azra Alizad (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN), Wilkins Aquino (Civil and Environ. Eng., Duke Univ., Durham, NC), and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN)

We present a full-wave inversion approach with total variation regularization for elastography. The proposed method is based on the minimization of an error in constitutive equations functional augmented with a least squares data misfit term referred to as MECE for "modified error in constitutive equations." The main theme of this paper is to demonstrate several key strengths of the proposed method on experimental data. In addition, some illustrative examples are provided where the proposed method is compared with a common shear wave elastography (SWE) approach. To this end, ultrasonically tracked displacement data from an acoustic radiation force (ARF) pulse are used in different phantoms including phantom with layered inclusion and triangle inclusion. The results indicate that the MECE approach can produce accurate shear modulus reconstructions in comparison with SWE, especially around the sharp edges in the layered and triangle inclusions. We compare shear modulus reconstruction using MECE and SWE with original inclusion shapes using two-dimensional normalized zero mean cross correlation, edge preservation index and dice coefficient similarity index. [Work supported by NIH Grant R01 CA174723.]

1:30

4pBAa3. Repeatability of linear and nonlinear quantitative compression elastography in the breast. Paul E. Barbone, Daniel Gendin (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), Yuqi Wang (Univ. of Wisconsin, Madison, WI), Rohit Nayak (Mayo Clinic, Rochester, New York), Assad Oberai (Univ. of Southern California, Los Angeles, CA), Timothy J. Hall (Univ. of Wisconsin, Madison, WI), Azra Alizad, and Mostafa Fatemi (Mayo Clinic, Rochester, MN)

Compression elastography allows the precise measurement of large deformations of soft tissue *in vivo*. From a measured large deformation, an inverse problem for both the linear and nonlinear elastic moduli distributions can be solved. As part of a larger clinical study to evaluate NEMs in breast cancer, we evaluate the repeatability of linear and nonlinear modulus maps from repeat measurements. Within the cohort of 31 subjects scanned to date, several had repeated scans. These repeated scans were processed to evaluate NEM repeatability. *In vivo* data were acquired by a custom, digitally controlled, uniaxial compression device with force feedback. RF-data were acquired using plane wave imaging, at a frame-rate of 200 Hz, with a ramp-and-hold compressive force of 8N, applied at 8 N/s. A two-dimensional (2D) block-matching algorithm was used to obtain sample-level displacement fields which were then tracked at subsample resolution using 2D cross correlation. Linear and nonlinear elasticity parameters in the Blatz model of tissue elasticity are estimated using iterative optimization. Repeatability between both modes and elastic modulus maps is measured and compared. Preliminary results indicate that when images are acquired in the same region of tissue, the modulus maps are consistent. [Work supported by NIH R01CA195527.]

Session 4pBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics II

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Sangpil Yoon, Cochair
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Contributed Papers

1:55

4pBAb1. Gas stabilizing titanium dioxide nanocones against desmoplastic cancer by ultrasound cavitation induced tumor penetration and sonodynamic therapy. Reju G. Thomas and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technol Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, jameskwan@ntu.edu.sg)

Sonodynamic therapy is an emerging technique for treating tumors by utilising ultrasound mediated reactive oxygen species (ROS) production from sonodynamic agents. Here, we have manufactured titanium dioxide nanocones (TDN) for local on-demand ROS generation. These nanocones nucleate inertial cavitation during exposure to therapeutic ultrasound. Furthermore, inertial cavitation enhances the penetration of the TDN into tissue. The particles were synthesized by a hydrothermal method in the presence of 1,6-hexanediamine as stabiliser. Electron microscopy images confirm the formation of nanocone structures with a size of 300 nm (and confirmed with dynamic light scattering). The TDN displayed an inertial cavitation threshold of 1.9 MPa for a 0.5 MHz ultrasound transducer at 5 mg/ml concentration. We also show that FITC conjugated TDN penetrated 2% agarose mold upto 1.2 cm distance after exposure to ultrasound for 10 min. Finally, a ROS release profile of TDN under ultrasound exposure was established using ROS sensor 1,3-diphenylisobenzofuran (DPBF). After 15 min of exposure to high intensity focused ultrasound (0.5 MHz center frequency) time, TDN in the presence of DPBF showed a significant decrease in UV absorbance compared to control, verifying that ROS were generated under ultrasound exposure. TDN opens up the potential for targeted sonodynamic cancer therapy.

2:10

4pBAb2. Acoustic microstreaming due to an oscillating contrast microbubbles near a substrate: Velocity, vorticity and closed streamlines. Nima Mobadersany and Kausik Sarkar (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Intravenously injected microbubbles used as ultrasound contrast enhancing as well as drug delivery agents are encapsulated by a nanometer-thick layer of lipids, proteins, or polymers to stabilize them against premature dissolution. Here, acoustic microstreaming due to an oscillating microbubble, either coated or free, responsible for sonoporation and other bioeffects is analytically investigated. The detailed flow field is obtained, and the closed streamlines due to the ring vortex are plotted in both Eulerian and Lagrangian descriptions. Analytical expressions are found for the ring vortex showing that its length depends only on the separation of the microbubble from the wall and the dependence is linear. The circulation as a scalar measure of the vortex is computed quantitatively identifying its spatial location. The functional dependence of circulation on bubble separation and coating parameters was shown to be similar to that of the shear stress. [Work supported partially by NSF CBET 1602884 and GWU.]

2:25

4pBAb3. Time-dependent nanobubble stability: Correlating bubble size and concentration with ultrasound performance. Eric C. Abenojar, Christopher Hernandez (Dept. of Radiology, Case Western Reserve Univ., 2185 S Overlook Rd., Cleveland Heights, OH 44106, eca20@case.edu), Judith Hadley (Malvern Panalytical, Westborough, MA), Al C. De Leon (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), Robert Coyne (Malvern Panalytical, Westborough, MA), Michael C. Kolios (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada), and Agata Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH)

Lipid shell-stabilized nanobubbles (NB, <500 nm) are widely explored as next-generation contrast agents for diagnostic ultrasound (US) imaging and drug delivery. For a successful clinical translation, it is important to understand the factors which contribute to the stability and rate of signal decay from the NB over time. The small size and fragile nature of NB have limited the characterization of their stability to correlations with their loss of signal over time under US. Bubble oscillations in the acoustic field, however, can accelerate their dissolution process. In this study, the passive, non-acoustically driven dissolution of lipid-shelled, C_3F_8 NB, and the relationship between bubble size/concentration and US signal intensity were assessed. The change in the acoustic activity of NB over time was correlated with the changes in size and concentration of the buoyant (bubbles) and non-buoyant particle population, measured using a novel resonant mass measurement technique. Clinical US was used to measure signal enhancement at different time points in a tissue phantom (f = 12.0 MHz, MI: 0.29, 1 fp/s). Results demonstrate a clear nonlinear relationship between the rate of ultrasound signal decay and concentration. While US signal decayed significantly over time (from 0 to 5 h), bubble concentration did not change significantly. A statistically significant decrease in the NB diameter was observed 1 h after the NBs were prepared and isolated while no change in the size was observed between 1 and 5 h.

2:40

4pBAb4. Microbubble translation in plane-wave ultrasound transmission. Outi Supponen (Mech. Eng., Univ. of Colorado at Boulder, 1111 Eng. Dr., Boulder, CO 80309, outi.supponen@colorado.edu), Francesco Guidi (Information Eng., Univ. of Florence, Florence, Italy), Awaneesh Upadhyay (Mech. Eng., Univ. of Colorado, Boulder, CO), Hendrik J. Vos (Biomedical Eng., Erasmus MC, Rotterdam, The Netherlands), Mark Borden (Mech. Eng., Univ. of Colorado, Boulder, CO), and Piero Tortoli (Information Eng., Univ. of Florence, Firenze, Italy)

Primary radiation force is capable of translating microbubbles in the focal region of single-element and array ultrasound probes. This effect can be harnessed to enhance the contact between ligand-bearing microbubbles and targeted endothelium for applications in targeted drug delivery and ultrasound molecular imaging. In this study, displacements of lipid-coated

microbubbles associated with plane-wave transmission are investigated using the multi-gate Doppler approach, and compared with focused-wave transmission at equivalent peak negative pressures. In plane wave transmission, the radiation force is nearly uniform over the field of view and therefore allows for a more uniform translation of microbubbles compared to focused wave. Statistically determined median displacements are in good agreement with the axial and lateral ultrasound beamplots both in plane-wave and focused-wave transmissions, while peak microbubble displacements reveal a number of discrepancies. Distinct size-isolated microbubble populations (diameters 1–2 μm, 3–4 μm, 4–5 μm, 5–8 μm, and polydisperse) were tested, showing important differences in their displacements and a strong driving frequency dependence thereof. These findings help tune the ultrasound transmission parameters for uniform and size-selective microbubble translations.

2:55

4pBAb5. Microstructural anisotropy evaluation in trabecular bone structure using the mode-converted (longitudinal to transverse, L-T) ultrasonic scattering. Omid Yousefian (North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27606, oyousef@ncsu.edu), Hualong Du (North Carolina State Univ., Lincoln, NE), Timothy Horn, and Marie M. Muller (North Carolina State Univ., Raleigh, NC)

The mode-converted ultrasonic scattering method is utilized to characterize the structural anisotropy of a phantom mimicking trabecular bone, fabricated using metal additive manufacturing from a high resolution CT image of trabecular horse bone. A normal incidence transducer transmits longitudinal waves into the sample, while the scattered transverse signals are received by an oblique incidence transducer. Four L-T measurements are performed by collecting scattering from four directions. The results show that the L-T converted scattering amplitude is highly dependent on the microstructural anisotropy direction. The ratios of L-T converted amplitudes for measurements in different directions is calculated to characterize the anisotropy of sample. The results show that the anisotropy is changing along the sample, which coincides with simulation results previously obtained on the same structures, as well as with the anisotropy estimated using image processing of the CT scans. The anisotropy was shown to increase monotonously along the sample from 0.48 to 0.7 depending on the location. At the same time, the ratio of LT scattering amplitude measured in two perpendicular directions was shown to increase monotonously from 0.6 to 0.67. These results suggest the potential of mode-converted methods to assess the anisotropy of structures including trabecular bone.

3:10

4pBAb6. Evaluation of bone fracture healing in children using acoustic radiation force: Initial *in vivo* results. Siavash Ghavami, Adriana Gregory, Jeremy Webb (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., 200 First St. SW, Rochester, MN 55902, roudsari.seyed@mayo.edu), Max Denis (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Lowell, MA), Viksit Kumar (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), Todd A. Milbrandt, A. Noelle Larson (Orthopedic Surgery, Mayo Clinic College of Medicine & Sci., Rochester, MN), Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), and Azra Alizad (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN)

Vibrational characteristics of the bone are directly dependent on bone's physical properties. In this paper, a vibrational method for bone evaluation is introduced. We propose a new type of quantitative vibro-acoustic method based on acoustic radiation force of ultrasound for bone characterization in patients with fracture. In this method, we excite the clavicle and ulna by an ultrasound radiation force (URF) pulse. The URF pulse induces vibrations in the bone, resulting in an acoustic wave that is measured by a hydrophone placed on the skin. The resulting acoustic signals were used for wave velocity estimation based on cross-correlation technique. To further separate different vibration characteristics, we adopt a variational mode decomposition (VMD) technique to decompose the received signal into an ensemble of band-limited intrinsic mode functions, which allows analyzing the acoustic signals in terms of their constitutive components. We conducted a prospective study that included a total of 15 patients, 12 with clavicle fractures and 3 with ulna fractures. The contralateral intact bones were used as control. Statistical analysis demonstrated that fracture bones can be differentiated from intact bone with a detection probability of 80%. Also, we introduce a "healing factor" that quantifies the progress of healing in clavicle bones. Statistical analysis showed that healing factor can track the progress of healing in clavicle in 80% of fractures.

3:25

4pBAb7. Ultrasonic bone assessment using backscatter difference measurements at 1 MHz. Brent K. Hoffmeister, Evan N. Main, and Phoebe C. Sharp (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, hoffmeister@rhodes.edu)

There is interest in developing ultrasonic techniques that can be used to detect changes in bone caused by osteoporosis. One approach, called the backscatter difference technique, measures the power difference between two portions of a backscatter signal from cancellous bone. Previous laboratory studies have tested the technique using transducers with center frequencies $>$ 2 MHz. The present study uses a 1 MHz transducer which may improve performance at central skeletal sites such as the hip and spine. Measurements were performed *in vitro* on 54 cube shaped specimens of cancellous bone from 14 human femurs using a broadband, single element 1 MHz transducer. Received backscatter signals were analyzed to determine the normalized mean of the backscatter difference (nMBD) which was computed by measuring the power difference between two gated portions of the backscatter signal in decibels and dividing by the gate separation in microseconds. Linear regression analysis found weak to moderate correlations ($0.13 \leqq R \leqq 0.66$) between nMBD and bone density, depending on which portions of the signals were analyzed. These results suggest that backscatter difference measurements using a 1 MHz transducer may be able to detect changes in bone caused by osteoporosis.

3:40–3:55 Break

3:55

4pBAb8. Development of high frequency ultrasound-based technique to increase cell permeability. Sangpil Yoon (Aerosp. and Mech. Eng., Univ. of Notre Dame, 151 Multidisciplinary Res. Bldg., Notre Dame, IN 46556, syoon4@nd.edu), Yingxiao Wang (BIoEng., Univ. of California, San Diego, La Jolla, CA), and K. K. Shung (Biomedical Eng., Univ. of Southern California, Los Angeles, CA)

Ultra-high frequency ultrasonic transducers have been developed by limited groups for cellular applications and high resolution imaging purposes. We have developed 150 MHz ultrasonic transducers with a focal size of smaller than 10 μm to increase permeability of cells to introduce macromolecules into cell cytoplasm. Cell-based therapy has enormous potential to treat neurodegenerative disease and cancer by engineering cells. One of the main challenges in cell-based therapy has been the safe intracellular delivery of macromolecules such as proteins and nucleic acids. We have developed a high frequency ultrasound-based technique for simultaneous and targeted single cell intracellular delivery of diverse types of macromolecules by increasing permeability of cell. High frequency ultrasound has a focus with area smaller than a single cell and enough focusing gain to directly disrupt cell lipid bilayer without microbubbles. Extremely thin layer of lithium niobite single crystal was used to generate 150 MHz sound waves. The transducer was integrated with microscope to apply acoustic pulses to increase permeability. CRISPR-Cas9, programmable gene editing tools, were delivered into single cells after the permeability was increased by high frequency ultrasound beam transmitted from the developed high frequency ultrasonic transducers. This study showed that the direct disturbance of cell membrane without microbubbles can be achieved by high frequency ultrasound for the safe delivery of macromolecules by increasing cell permeability.

4:10

4pBAb9. Standing acoustic waves in microfluidic channels for enhanced intracellular delivery of molecular compounds. Connor S. Centner (BioEng., Univ. of Louisville, 580 S Preston, Louisville, KY 40202, connor.centner@louisville.edu), Mariah C. Priddy (BioEng., Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (BioEng., Univ. of Louisville, Louisville, KY)

Intracellular delivery of molecular compounds is required for many *in vitro* research applications. Ultrasound-induced cavitation has been shown to enhance intracellular delivery of molecular compounds via mechanisms that may include sonoporation or endocytosis. Recently, acoustofluidic approaches have been developed to utilize standing acoustic waves (SAW) for cell manipulation in microfluidic channels. In this study, the effect of SAW on fluorescein release from perfluorocarbon double emulsion droplets in microfluidic channels was explored. Vaporization of perfluorocarbon double emulsion droplets induced by SAW may potentially enhance intracellular delivery of molecular compounds. In this study, fluorescein-loaded double emulsions were passed through a microfluidic device and exposed to 8-MHz SAW. Fluorescein release was quantified by measuring the change in fluorescence of the supernatant before and after treatment. Treatment with SAW in microfluidic channels increased the fluorescence by 8.8-fold compared to the baseline level. Fluorescein release was also higher after treatment with SAW compared to samples that passed through the microfluidic device without exposure to SAW ($p = 0.03$). These results suggest that SAW and perfluorocarbon double emulsions in microfluidic channels could potentially enhance the efficiency and consistency of intracellular molecular delivery *in vitro*.

4:25

4pBAb10. Sounding out bacteria: Microstructural effects of therapeutic ultrasound on bacterial biofilms. Lakshmi Deepika Bharatula (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore), Scott Rice, Enrico Marsili (Singapore Ctr. for Environ. Life Sci. Eng., Nanyang Technolog. Univ., Singapore, Singapore), and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, jameskwan@ntu.edu.sg)

Treatment of chronic infections due to formation of bacterial biofilms are a huge risk due to the growing concerns with antimicrobial resistance. Biofilms grow in a complex and dynamic environment that weaken the effect of antimicrobials. Yet, the current strategy to tackle the problem is the development of novel drugs. However, the increasing prevalence of antimicrobial resistance suggests that an alternative treatment strategy without, or in synergy with, antibiotics is necessary to combat the biofilm infections. We and others have proposed high intensity focused ultrasound (HIFU) as a means to disrupt the biofilm matrix and improve therapy. Yet to date, there is limited knowledge on the cellular activity triggered by the biofilm-acoustic interactions. Here, we report the effect of HIFU at 500 kHz center frequency in absence of antibiotics or microbubbles on the microstructure of biofilms formed by *Pseudomonas aeruginosa*. Changes to the biofilm after acoustic exposure were characterized by confocal microscopy and electrochemical impedance spectroscopy. We observed a drop in the biomass at pressures where non-linear acoustics were dominant, and an increase in cellular activity. Our results suggest that there are acoustic bio-effects present in these bacteria that have not yet been reported.

4:40

4pBAb11. Therapeutic ultrasound-induced insulin release *in vivo*. Tania Singh (Biomedical Eng., The George Washington Univ., 800 22nd St. NW, Ste. 5290, Washington, DC 20052, taniasingh@gwu.edu), Ivan Suarez Castellanos (INSERM, Washington, District of Columbia), Diti Chatterjee Bhowmick (Biological Sci., The George Washington Univ., Washington, DC), Joshua Cohen (GW Medical Faculty Assoc., Washington, DC), Aleksandar Jeremic (Biological Sci., The George Washington Univ., Washington, DC), and Vesna Zderic (Biomedical Eng., The George Washington Univ., Washington, DC)

We have previously shown that therapeutic ultrasound is capable of stimulating insulin release from pancreatic beta cells, non-invasively, safely and effectively. The aim of this work is to conduct preliminary animal studies to evaluate the feasibility of controlled insulin release *in vivo* using therapeutic ultrasound. Wild type hIAPP+/+ white FVB mice were randomly assigned to either the ultrasound treatment group or the sham group. Mice in the ultrasound treatment group received one five-minute treatment of continuous 1 MHz ultrasound at 1 W/cm^2 . Blood samples were collected via tail nick immediately prior to ultrasound application and immediately after ultrasound application. The pancreas was excised for histological analysis using H&E staining. No gross damage—including any burns on the skin—in the treatment area were observed and there was no evidence of skin burning or internal damage of the abdominal organs, especially the pancreas, found during necropsy. As measured by ELISA, the experimental group treated with ultrasound exhibited an increase of 0.43 ng/ml in blood insulin concentration compared to a 0.60 ng/ml decrease in the control group after 5 min ($p < 0.01$). Our preliminary results show promise in the translational potential of therapeutic ultrasound in the treatment of type 2 diabetes. We expect that our approach, with careful selection of ultrasound parameters, may provide a safe, controlled and targeted stimulation of insulin release from the pancreatic beta cells.

4:55

4pBAb12. Development and characterization of acoustically responsive exosomes for simultaneous imaging and drug delivery applications. Jenna Osborn (George Washington Univ., Ste. 3000, 800 22nd St. NW, Washington, DC 20052, jennakosborn@gwu.edu), Jessica Pullan, James Froberg, Yongki Choi, Sanku Mallik (North Dakota State Univ., Fargo, ND), and Kausik Sarkar (George Washington Univ., Washington, DC)

Exosomes are naturally secreted bilayer vesicles ranging in size from 40 to 200 nm that play a critical role in cell-to-cell communications and protein and RNA delivery. Researchers have explored exosomes as potential drug delivery vehicles due to their natural morphology and small size. Here, for the first time, bovine milk derived exosomes have been modified to be acoustically responsive as potential ultrasound contrast agents or a drug carrier. The echogenic exosomes were formed through a freeze-drying process in the presence of mannitol. The size and morphology of the particles were assessed with a qNano^{TM} and atomic force microscopy (AFM). The ultrasound response of these particles was characterized through linear and nonlinear scattering behaviors. The presence of the echogenic exosomes enhances the scattered signal by 11.4 ± 6.3 dB. The stability of these particles under constant ultrasound exposure were assessed to be similar to that of echogenic polymersomes. The variation of mannitol concentration was assessed. To assess the imaging improvement of ultrasound imaging, the exosomes were injected through a tail vein in mice. The modification of the echogenic exosomes shows to have great promise as potential ultrasound contrast agents or ultrasound responsive drug delivery system.

5:10

4pBAb13. Therapeutic ultrasound for improving the tropical corneal delivery of macromolecules. Hanaa H. Almogbil, Camille Daszynski (School of Eng. and Appl. Sci., The George Washington Univ., Sci. & Eng. Hall, 800 22nd St. NW, Washington, DC 20052, Hanaa@gwu.edu), Erik A. Rodriguez (Dept. of Chemistry, The George Washington Univ., Washington, DC), Tania Singh (School of Eng. and Appl. Sci., The George Washington Univ., Washington, DC), Mary Ann Stepp (School of Medicine, The George Washington Univ., Washington, DC), and Vesna Zderic (School of Eng. and Appl. Sci., The George Washington Univ., Washington, DC)

The goal of this project is to facilitate the delivery of topical drugs into the cornea and anterior segment of the eye using therapeutic ultrasound which could present a promising treatment for keratoconus and other corneal diseases. Each cornea is dissected and placed in a diffusion cell. smURFP-blue, a blue fluorescent chromophore, was used as the drug. The experimental groups of corneas were treated with 1 and 0.8 W/cm^2 continuous ultrasound for 5 min at frequencies of 400 kHz and 600 kHz, respectively, then left in the diffusion cell for another 55 min. Fluorescence images of the fixed corneas were obtained to determine the relative amount of smURFP-blue that remained in the tissue. Safety of ultrasound application was tested by comparing the damage in the corneal layers. Spectroscopy measurements indicated no statistical difference in the presence of the chromophore in the receiver compartment in ultrasound- and sham-

treatment groups. Preliminary results showed greater fluorescent intensity in the cornea when smURFP-blue is delivered with ultrasound compared to smURFP-blue added without ultrasound. The histology studies did not show any significant damage in ultrasound-treated corneas. This work may allow for the development of an inexpensive, clinically applicable, and minimally invasive ultrasound method for corneal drug delivery.

5:25

4pBAb14. Optimization of molecular delivery to red blood cells using an ultrasound-integrated microfluidic system. Emily M. Murphy, Mariah C. Priddy (BioEng., Univ. of Louisville, 200 E. Shipp Ave., Louisville, KY 40208, emmurp09@louisville.edu), Brett R. Janis, Michael A. Menze (Biology, Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (BioEng., Univ. of Louisville, Louisville, KY)

The shelf-life of donated red blood cells (RBCs) for transfusions is currently limited to six weeks when stored under refrigeration. This causes supply shortages worldwide and prevents transfusions in locations that lack access to cold-chain storage. Recently, a new approach to store RBCs as a dried powder at ambient temperature was developed. This method utilizes an ultrasound-integrated microfluidic platform to induce intracellular delivery of compounds that protect cells during desiccation and rehydration. The objective of this study was to detect cavitation emissions in order to optimize parameters for molecular delivery to RBCs in this system. Ultrasound was continuously generated in the microfluidic channels using an 8-MHz PZT plate and acoustic emissions were passively detected with an identical PZT plate aligned coaxially. Fluorescein and lipid-coated microbubbles were added to RBC solutions in order to nucleate cavitation and enhance intracellular molecular uptake as measured by flow cytometry. Increased levels of broadband emissions were detected at microfluidic flow rates associated with higher fluorescein delivery to RBCs. These results suggest that inertial cavitation plays an important role in enhancing molecular delivery to RBCs in the microfluidic channels. Optimization of this system may enhance delivery of protective compounds for long-term preservation of blood.

THURSDAY AFTERNOON, 16 MAY 2019 WILLIS, 1:30 P.M. TO 3:30 P.M.

Session 4pEA

Engineering Acoustics: General Topics in Engineering Acoustics: Characterization and Measurement

Matthew D. Guild, Cochair
Acosutics Div., Naval Research Lab., Code 7165, 4555 Overlook Avenue, SW, Washington, DC 20375

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Contributed Papers

1:30

4pEA1. Modeling the vibration of a thin bar using SimScape. Carter J. Childs and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ.University Park, PA 16802, cjc357@psu.edu)

The standard treatment of longitudinal and flexural vibrations of thin bars follow the methods described by Raleigh. The solution for longitudinal vibrations is completely analogous to electrical signals in electrical transmission lines. Thus, longitudinal mechanical vibrations can be modeled using lumped or distributed parameter analog circuits. The same is true of transverse vibrations of perfectly flexible strings. However, when bending stiffness is included for transverse vibrations, the simplicity of the transmission line analogies is not present. The authors are not aware of a lumped parameter model of the flexural vibration of a thin bar that includes the effects of bending stiffness. This paper presents a lumped parameter model for a thin bar that provides accuracy similar to that of a lumped parameter model of a flexible string. The differential length element of the bar is modeled in the same way as the differential element in the standard treatment of bra vibrations. The model will be demonstrated in the SimScape modeling software, though it can also be implemented in Modelica and possibly in some versions of SPICE.

1:45

4pEA2. Engine characterization using experimental method and prediction of insertion loss of the exhaust system. Manish Chhabra (Mech., Univ. of Cincinnati, 275N Marr Rd., Apartment 101, Columbus, IN 47201, chhabra.manish90@gmail.com)

The evaluation of the acoustic performance of an exhaust system at the design stage requires a correlated engine model and reasonably approximate input boundary conditions to simulate the end results, both of which are not easily available. It is known that the input boundary conditions for insertion loss analysis require two engine parameters, namely, source impedance and source strength spectra. This study describes experimental measurements for these parameters using in-duct measurement via the multi-load method for a six-cylinder diesel engine and calculation for insertion loss using GT-Power. The research discusses the approach taken to select the acoustic load cases considered for the multi-load method and then the execution of the test plan for different engine operating conditions The time domain data sets were processed to obtain the frequency spectra and was used to get the impedance of the acoustic load cases and finally the source impedance and source strength spectra for different engine operating conditions. The results obtained using all the acoustic load cases were optimized by filtering out unacceptable load cases and then re-evaluating the source characteristics to use them as input boundary conditions for insertion loss analysis. The analysis results were then compared to the experimental insertion loss.

2:00

4pEA3. Impedance measurements of acoustically nonlinear engine components. Eric Van horn, Luke Humphreys, and David E. Scarborough (Aerosp. Eng., Auburn Univ., 211 Davis Hall, AL 36832, hnh0004@auburn.edu)

Currently, the acoustic analysis of combustion systems generally consists of simplified impedance networks. For effective simulation, the impedance of these components must be known, for which there is a large gap in the current knowledge. This study examines the acoustic impedance for various different types of gas-turbine engine components, including acoustically nonlinear components such as swirlers and orifice plates. Experimental results are obtained using a specially designed impedance tube. Acoustic pressure and velocity measurements are made using a multiple microphone method. The impedance is then calculated using the reflection coefficient, and impedance subtraction used to isolate the test article. This method obtains accurate results within a relatively wide range of frequency values, 100–1500 Hz for this study. Amplitude sweeps conducted for each test article demonstrate the nonlinear aspects, if any, for each test article. These results are then compared to acoustic simulations conducted in COMSOL to assess the capabilities and shortcoming of COMSOL's linear acoustic package to provide predictions of the acoustic impedance of these components. The results for this are intended to provide valuable data for modeling the acoustics of combustion systems, as well as demonstrate an effective method for obtaining impedance data for various acoustic components.

2:15

4pEA4. Initial results in designing an acoustic sound simulator for heavy equipment. Nathaniel Wells, Scott D. Sommerfeldt, and Jonathan Blotter (Brigham Young Univ., N308 ESC, Provo, UT 84602, nateswells@gmail.com)

This paper focuses on the initial development of a vehicle cab sound simulator. This sound simulator has two objectives. First, it can be coupled with a visual simulator and used for operator training. Second, it will be used by designers such that when structural modifications are made to the vehicle, the acoustic response and sound quality in the cab can be predicted. To begin to understand and implement this simulator, transfer functions were measured for several structures progressing from simple to complex and used to generate a simulated signal. Post-processing techniques were used to improve the overall quality of the simulated responses. Similarly, the structures were recreated in a numerical software package, where the transfer functions were calculated numerically and used to generate simulated responses. These signals were compared to the measured response of the system and auralized to determine the effectiveness of the simulation.

2:30

4pEA5. On a cooling speed comparison a sound fire extinguisher with the blade. Bong Young Kim and Myungjin Bae (Commun. Eng., Soongsil Univ., 21-1, Garak-ro 23-gil, Songpa-gu, #203, Seoul 05669, South Korea, bykim8@ssu.ac.kr)

Sound Fire Extinguisher, which is actively studied at Sori Sound Engineering Research Institute (SSERI), is a new type of extinguishment facility that can be used for suppression and prevention of conflagration in various environments. Sound Fire Extinguisher uses acoustic lens to minimize the attenuation of sound energy and transfer energy to the target point. It can prevent conflagration by lowering ambient temperature even before conflagration. In this study, we experimented to see if the Sound Fire Extinguisher could prevent conflagration by lowering the ambient temperature. Experimental results show that when the Sound Fire Extinguisher sound component of the same wind speed is supplied, the heated tableware is cooled by 10%–20% faster than the wind speed of 2 m/s. These results show that the Sound Fire Extinguisher can be used to prevent conflagration, since the sound component of the Sound Fire Extinguisher itself promotes the surrounding thermal dissipation to cool quickly.

2:45

4pEA6. Design and characterization of tabletop anechoic and sonication chambers. Olivia Ott, Shane Lani, and Jim Sari (Johns Hopkins Univ. Appl. Phys. Lab. 11100 Johns Hopkins Rd., Laurel, MD 20723, olivia.ott@jhuapl.edu)

Access to anechoic chambers for acoustic testing can be an expensive and limited resource for acousticians. The purpose of this work was to develop small scale, portable anechoic and sonication chambers that could be used for rapid and simultaneous testing. Using consumer off the shelf (COTS) products, low cost anechoic and sonication chambers were designed and built. Upon completion the prototypes were characterized for their acoustic performance. Both the anechoic and sonication chambers were characterized by their ability reduce ambient noise. The characterization was completed by measuring transmission loss (TL) through the chamber walls. Second, the sonication chamber was tested for spatial sound pressure level (SPL) variation. The procedure for spatial testing inside the chamber included measuring SPL along a horizontal stage using mini microphones. The performance and characterization of these chambers will be presented and discussed.

3:00

4pEA7. Absorption measurement of acoustic materials using a spherical microphone array. Samuel Dupont (LAUM / GAUS, Universite du Mans, Ave. Olivier Messiaen, Le Mans 72085, France, samuel.dupont@univ-lemans.fr), Manuel Melon (LAUM, Le Mans, France), and Alain Berry (GAUS, Sherbrooke, QC, Canada)

Standardized methods for the sound absorption measurement are restricted to normal or diffuse incidence. The capture of the sound field over a plane material sample using a spherical microphone array has already been proposed as a solution to measure surface impedance or sound absorption. However, the proposed solutions use either modal beamforming or a direct computation of the pressure on the sphere. In this presentation, a method which uses the spherical harmonics expansion to calculate the absorption using the spherical holography principle is proposed. It allows to measure the characteristics of the material without relying on an accurate definition of the sound scene such as the exact position of the source but only on the distance of the spherical microphone to the material. Simulation and experimental result of surface impedance and sound absorption will be presented as well as comparisons with another approach from the literature based on sparse array processing of spherical antenna.

3:15

4pEA8. Ultimate wave decomposition method for characterization of symmetric and asymmetric materials. Reza Ghaffarivardavagh, Rui Wu (Mech. Eng., Boston Univ., 8 saint Mary St., Photonic Bldg., Rm. 832, Boston, MA 02215, ghaffari@bu.edu), Stephan Anderson (Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., Boston, MA)

To date, several approaches leveraging impedance tubes have been proposed to retrieve the complex acoustic reflection and transmission coefficient through the materials. In these approaches, commonly, accurate derivation of the reflection and transmission relies on the several assumptions such as material symmetry or the type of the terminations (load) used in the impedance tube. Herein, we have proposed an Ultimate wave decomposition approach (UWD) that may be employed for the precise derivation of the reflection and transmission coefficients, regardless of the termination type. Moreover, it is demonstrated that by employing the UWD method, utilizing single arbitrary termination, symmetric material, and two arbitrary terminations, asymmetric material can be fully characterized. The performance of the presented methodology has been compared experimentally with four different methods with respect to both symmetric and asymmetric cases, from which the superior performance of the UWD method is demonstrated. UWD represents a highly practical approach in which reflection and transmission and consequently acoustical properties of the material, can be precisely measured and calculated using a very simple procedure.

Session 4pNS

Noise, Signal Processing in Acoustics, and Psychological and Physiological Acoustics: Advances and Applications in Sound Quality Metrics

S. Hales Swift, Cochair
Physics and Astronomy, Brigham Young University, N221 ESC, Provo, UT 84602

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

Chair's Introduction—1:05

Invited Papers

1:10

4pNS1. The loudness model used in ISO532-3: Development, evaluation and prospects. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

ISO 532-2 is based on the loudness model for stationary sounds described by Moore and Glasberg [*JASA* (2007)]. This model is similar to that in ANSI S3.4-2007, except that the model in ISO 532-2 incorporates binaural inhibition: a strong input to one ear in a given frequency region reduces the effective level of a weaker input to the other ear in nearby frequency regions. ISO 532-3 has been proposed as an extension to ISO 532-2 to deal with time-varying sounds. It generates predictions of short-term loudness, the loudness of a short-segment of sound such as a word in a sentence or a single note in a piece of music, and of long-term loudness, the loudness of a longer segment of sound, lasting 1–5 s, such as a whole sentence or a musical phrase. The model gives reasonably accurate predictions of the overall loudness of technical sounds (e.g., factory noises), of speech whose dynamic range has been compressed or expanded, and of sounds whose time pattern and spectra differ at the two ears. However, the model needs to be extended to generate predictions of the overall loudness impression of a sound environment over a period of several minutes to an hour.

1:30

4pNS2. Rapid calculating of loudness according to ANSI S3.4-2007 with the Glasberg and Moore 2002 extension to time-varying signals in MATLAB. S. Hales Swift (Energy Systems Div., Argonne National Lab., N221 ESC, Provo, Utah 84602, hales.swift@gmail.com), and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The ANSI S3.4-2007 standard gives a method for calculating the predicted loudness of stationary sounds for an average listener. Glasberg and Moore (2002) provide an extension of the method to time-varying sounds. The mathematical structure of the excitation in the loudness calculation is amenable to significant acceleration in MATLAB by expressing portions of the calculation, notably those representing the cochlear filtering process, in terms of matrices. Thus, procedures to achieve rapid processing of loudness are set forth. Possible extensions of this approach to other metrics within the same family are considered.

1:50

4pNS3. The description of fan noise by indexes based on the specific loudness. Stephan Töpken and Steven van de Par (Acoust., Univ. of Oldenburg, Carl-von-Ossietzky-Str. 9-11, Oldenburg 26129, Germany, stephan.toepken@uni-oldenburg.de)

In a previous study of the authors, a broad variety of fan sounds that were equalized in overall A-weighted level was rated in listening experiments with a semantic differential. The factor analyses of the ratings indicated six perceptual dimensions and five groups of sounds, which shows the rich variety of sound characteristics covered by the tested fan sounds. The results showed that the groups of pleasant and unpleasant sounds differed mainly with respect to the first three perceptual dimensions, "pleasant," "humming/bass," and "shrill." An analysis of the specific loudness according to the DIN 45631 standard revealed systematic differences in the specific loudness patterns for the different groups of fan sounds. It was possible to define two psychoacoustic indexes that correlate highly with the factor values of the three most important perceptual dimensions of fan noise. The most important index, N_{ratio}, relates the amount of loudness resulting from low mid-frequency content between 2 Bark and 5 Bark to the loudness from high frequency content above 10 Bark. The identified boundaries of the frequency ranges employed in the indexes are in good agreement with those found for air conditioning noise, air cleaners and in the context of sound masking in offices.

2:10

4pNS4. Psychoacoustic Roughness Standard. Roland Sottek and Julian Becker (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Roughness of acoustic signals has been a focus of sound design for many years. A rough sound can contribute to a sporty character of an engine, but also to a strong annoyance. It is desired that a sound fits the product. This goal should be achievable as early as possible in the development process. For this purpose, perceptual evaluations in combination with model calculations and simulation tools must be used. Existing roughness calculation models work well for synthetic signals such as modulated tones and noise signals. However, the roughness prediction is much more challenging for technical sounds because of the more complex spectral and temporal patterns. Although the consideration of roughness is very common in practice, there is still no standardized roughness calculation method. This paper describes a method that is based on a model of human hearing according to Sottek. It has been optimized for non-linear processing and the weighting of the modulation spectra. Additionally, a proposal for calculating a binaural single value of roughness is given. This model allows to predict the perceived roughness very well. The standardization of this roughness calculation method is planned both as a German standard (DIN) and as part of ECMA-74.

2:30

4pNS5. Inconsistencies between the predicted qualities of enhanced speech signals from two objective metrics. Zhuohuang Zhang and Yi Shen (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

Objective speech-quality metrics have been used widely as a tool to evaluate the performance of speech enhancement algorithms. Two widely adopted metrics are Perceptual Evaluation of Speech Quality (PESQ) and Hearing-Aid Speech Quality Index (HASQI). While PESQ is based on a highly-simplified phenomenological model of auditory perception for normal-hearing listeners, HASQI contains processing steps that represent the physiology of the auditory periphery and is able to capture the perceived speech quality from hearing-impaired listeners. In the current study, the performance of deep-learning-based speech enhancement algorithms was evaluated using the two objective metrics. The algorithms were implemented so that the audio features were represented in either a linear frequency scale or a nonlinear frequency scale (i.e., Mel scale). Higher speech quality for linear- than Mel-scale processing was predicted by HASQI, but PESQ was less sensitive to the difference in frequency scale. To resolve the discrepancies between the two metrics, a behavioral experiment was conducted following the ITU recommended procedure for assessing speech quality (ITU-R BS.1534-1). Listeners strongly preferred the enhanced speech using linear- over Mel-scale processing, which is consistent with predictions from HASQI. The sources of the discrepancies between PESQ and HASQI were also explored via further acoustic analyses.

2:50

4pNS6. Reliability and validity of sound quality metrics versus objectivity. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

Often it is mentioned to get an objective measurement of the subjective evaluation. The result is a sound quality metric. The terms subjective and objective are traditionally associated with aspects related to human perception and physical measures respectively. Objectivity describes the independency of test results from the respective researcher, reliability considers whether the same results would be achieved, if the research procedure would be repeated, validity means whether a method measures what it is intended to measure. Physical measurements could have a high reliability but low validity, whereas perceptual measurements often possess a relatively low reliability but relatively high validity. For a sound quality metric, the use of predictors supported by theoretical considerations and plausibility is very important. Furthermore, a robustness analysis is needed indicating that the metric is not very susceptible to choice of input data. Be aware, the use of statistics does not replace thinking; the use of predictors must be plausible or needs theoretical background. Examples of typical sound quality metrics will be given.

3:10–3:25 Break

3:25

4pNS7. Identification of perceived sound discomfort contributed from partially correlated vibration and noise sources in vehicles. Yu Huang and Weikang Jiang (Shanghai Jiao Tong Univ., Dongchuan Rd. 800 Jidong A819, Minhang, Shanghai 200240, China, yu_huang@sjtu.edu.cn)

The interior noise sources are often complex in vehicles, including not only structure-borne and air-borne noise sources but also vibration sources. These sources may be partially correlated and cannot be calculated using traditional methods, e.g., transfer path analysis and operational path analysis. On the other hand, it is necessary to study the sound quality of the vehicle interior noise to improve the comfort of drivers and passengers in vehicles. An operational partial singular value decomposing method together with sound quality analyses was employed in this study to determine the influence of various partially correlated sources with the perceived discomfort of the subject. The vibration and noise in vehicles were measured in a car when it was running on asphalt, concrete, gravel and bumpy roads. Thirty subjects used the absolute magnitude estimation method to rate the discomfort produced by noise stimuli. A discomfort model was proposed based on the relations between subjective magnitudes and the objective parameters of noise (i.e., the SPL, loudness, roughness, sharpness, and articulation index). The contributions of various vibration and noise sources to the vehicle discomfort were predicted well by the operational partial singular value decomposing method based on the discomfort model.

3:45

4pNS8. Sound quality metric indicators of rotorcraft noise annoyance using multilevel regression analysis. Matthew Boucher, Siddhartha Krishnamurthy, Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681-2199, matthew.a.boucher@nasa.gov), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

It is hypothesized that sound quality metrics, particularly loudness, sharpness, tonality, impulsiveness, fluctuation strength, and roughness, could all be possible indicators of the reported annoyance to helicopter noise. To test this hypothesis, a psychoacoustic test was recently conducted in which subjects rated their annoyance levels to synthesized helicopter sounds [Krishnamurthy, InterNoise2018, Paper 1338]. After controlling for loudness, linear regression identified sharpness and tonality as important factors in predicting annoyance, followed by fluctuation strength. Current work focuses on multilevel regression techniques in which the regression slopes and intercepts are assumed to take on normal distributions across subjects. The importance of each metric is evaluated one-by-one, and the variation among subjects is evaluated using simple models. Then, more complete models are investigated, which include the combination of selected metrics and random effects. While the conclusions from linear regression analysis are affirmed by multilevel analysis, other important effects emerge. In particular, a random intercept is shown to be more important than a random slope. In this framework, the relative importance of sound quality metrics is re-examined, and the potential for the modeling of human annoyance to helicopter noise based on sound quality metrics is explored.

4:05

4pNS9. Identifying metrics to predict annoyance due to Mach-cutoff flight ground signatures. Nicholas D. Ortega (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, njo5068@psu.edu), Michelle Vigeant (Acoust., The Penn State Univ., State College, PA), and Victor Sparrow (Acoust., The Penn State Univ., University Park, PA)

Theoretically, Mach-cutoff flight under ideal atmospheric conditions could lead to boomless supersonic flight observed under the flight path on the ground. Such ideal atmospheric conditions refract the sonic boom waves upwards at the caustic line, so they do not reach the ground. This presentation describes the perception of the evanescent sound field below the flight path. The work investigates perceptual attributes and metrics related to these unique sounds. Annoyance and three other perceptual factors (“Thunderous,” “Rumbly,” and “Swooshing”) were analyzed through subjective testing using pair-wise comparison. Stimuli used were from recordings made during NASA’s “Farfield Investigation of No-boom Thresholds” (FaINT). Linear regression with principal component analysis indicated which perceptual factors contribute to annoyance, and stepwise regression identified candidate metrics for predicting annoyance. Traditional loudness metrics (i.e., weighted Sound Exposure Level) were analyzed alongside sonic-boom specific metrics (i.e., Perceived Loudness) and sound quality metrics (i.e., Sharpness). [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, Project 42 through FAA Award No. 13-C-AJFE-PSU under the supervision of Sandy Liu. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

4:25

4pNS10. Comparison of sonic boom noise metrics from predictions and measurements under low atmospheric turbulence conditions. Alexandra Loubeau and William Doebler (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Six noise metrics have been identified as candidates for quantifying ground sonic boom levels from overflight of supersonic aircraft. Each of these metrics (PL, ASEL, BSEL, DSEL, ESEL, and ISBAP) has previously been investigated in meta-analyses using laboratory study data corresponding to perception of sonic booms in outdoor and indoor environments. These metrics are now computed and analyzed for a set of recorded outdoor sonic boom signatures under low atmospheric turbulence conditions. Predictions of the ground signatures are also computed, without inclusion of turbulence effects, and metrics are compared between measurements and predictions. Metrics least sensitive to atmospheric turbulence effects are identified as potentially more robust for quantifying the sonic boom level from a supersonic aircraft.

Contributed Papers

4:45

4pNS11. Stevens perceived levels of common impulsive noises, sonic booms, and sonic thumps. William Doebler and Jonathan Rathsam (NASA Langley Res. Ctr., M.S. 462, NASA Langley Res. Ctr., Hampton, VA 23681, william.j.doebler@nasa.gov)

NASA’s X-59 Quiet Supersonic Technology low boom flight demonstrator aircraft is being designed to produce a shaped sonic thump of 75 dB Perceived Level (PL) at the ground. The PL metric was chosen because it correlates well with human perception of sonic booms both outdoors and indoors. Members of the public often ask how loud 75 dB PL is. To communicate this level in terms of more familiar sounds, a PL reference scale was developed. Common impulsive sounds were recorded, and their PLs were computed. Some of the various impulsive sounds include distant thunder, basketball bounces, and car door slams (79, 81, and 89 dB PL, respectively). Concorde’s 105 dB PL traditional N-wave sonic boom is also included in the reference scale. Additionally, the impulsive sounds’ energy spectral densities and sone spectra are compared to that of a simulated X-59 ground waveform.

5:00

4pNS12. Developing a tone standard for air-conditioning and refrigeration equipment. Derrick P. Knight (Ingersoll Rand, 3600 Pammel Creek Rd., La Crosse, WI 54601, derrick.knight@irco.com)

AHRI Technical Committee on Sound is continuing the redevelopment of standard AHRI 1140—Sound Quality Evaluation Procedures for Air-Conditioning and Refrigeration Equipment. We are currently evaluating the feasibility of harmonizing this standard with a current ASHRAE funded study whose goal is to determine the threshold of annoyance due to tones in HVAC equipment. However, from a manufacturer’s perspective, it is very difficult to accommodate a metric measured in the listener’s space. Additionally, sound power test methods for HVAC equipment allow testing in a reverberant field, which poses significant challenges to measuring tones. This presentation will provide an update in regards to the development for

standard AHRI 1140 and will seek feedback from consultants and academics.

5:15

4pNS13. Effect of trading-off office background and intermittent noise levels on performance, annoyance, distraction, and stress. Martin S. Lawless, Zane T. Rusk, Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lawles@cooper.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

In open-plan offices, work performance is affected by the acoustic environment, which includes steady-state broadband noise and intermittent, occupancy-generated noise. High levels of broadband noise (e.g., HVAC noise) can mask intermittent sounds to reduce distraction, but risk causing fatigue and other noise-related symptoms that may be detrimental to performance. In this study, the impact of the acoustic environment on work performance was investigated by adjusting the relative levels of both broadband and intermittent noise. Participants were exposed to four different acoustic environments, either starting with high background noise and low intermittent levels or vice versa. While in each background condition, the subjects performed four cognitive tasks that evaluated memory, attention, reasoning, and planning skills, respectively. Heart rate variability and electrodermal activity (EDA) were measured to gauge arousal (stress levels) in each environment. After each exposure, participants were asked to rate annoyance, distraction, fatigue, and stress, among other subjective attributes. The EDA and ratings of distraction significantly increased as the intermittent noise levels increased, while noise annoyance ratings were consistent across each background condition. Additionally, performance on the cognitive tasks was impacted by the order in which the participants experienced the acoustic environments.

THURSDAY AFTERNOON, 16 MAY 2019 — JONES, 1:00 P.M. TO 3:30 P.M.

Session 4pPAa

Physical Acoustics and Signal Processing in Acoustics: Infrasound II

Roger M. Waxler, Cochair
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Philip Blom, Cochair
Los Alamos National Laboratory, Los Alamos National Laboratory, P.O. Box 1663, Los Alamos, NM 87545

Contributed Papers

1:00

4pPAa1. Infrasound propagation in multiple-scale random media using generalized polynomial chaos. Alexandre GOUPY (CMLA, ENS Paris-Saclay, CMLA, ENS Paris-Saclay, Cachan, France, alexandre.goupy@gmail.com), Christophe MILLET (CEA, DAM, DIF, Arpajon, France), and Didier LUCOR (LIMSI CNRS, Orsay, France)

Infrasound propagation in realistic environments is highly dependent on the information to specify the waveguide parameters. For real-world applications, there is considerable uncertainty regarding this information, and it is more realistic to consider the wind and temperature profiles as random functions, with associated probability distribution functions reflecting phenomena that are filtered out in the available data. Even though the numerical methods currently-in-use allow accurate results for a given atmosphere, high dimensionality of the random functions severely limits the ability to compute the random process representing the acoustic field, and some form of sampling reduction is necessary. In this work, we use polynomial chaos (gPC)-based metamodels to represent the effect of large-scale atmospheric variability onto the acoustic normal modes. The impact of small-scale atmospheric structures is modelled using a perturbative approach of the coupling matrix. This multi-level approach allows to estimate the statistical influence of each mode as the frequency varies. An excellent agreement is obtained with the gPC-based propagation model, with a few realizations of the random process, when compared with the Monte Carlo approach, with its thousands of realizations. Furthermore, the gPC framework allows computing easily the Sobol indices without supplementary cost, which is essential for sensitivity studies.

1:15

4pPAa2. Similarities and differences in infrasound propagation effects between arctic and temperate environments. Michelle E. Swearingen (U.S. Army ERDC, Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Sarah McComas (U.S. Army ERDC, Vicksburg, MS), D. K. Wilson, and Vladimir Ostashev (U.S. Army ERDC, Hanover, New Hampshire)

Meteorological conditions in an arctic environment differ significantly from those in a temperate environment. Atmospheric phenomena particular to polar regions, including wind patterns such as the polar vortex and low-level jets above strongly stable layers, strong temperature and humidity gradients, and density currents, could have unique impacts on infrasound propagation that are not observed in temperate locations. In this study, parabolic-equation simulations of sound propagation are performed using measured meteorological conditions for summer and winter conditions in temperate and arctic locations. The similarities and differences in environmental conditions between these two locations and their relative impact on the predicted transmission loss are examined. For summer conditions, a comparison to measured data from explosive sources is performed for both temperate and arctic locations.

1:30

4pPAa3. Dependence of Rayleigh integral predictions on the ground motion model for acoustic signals produced by underground explosions. Philip Blom (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, pblom@lanl.gov)

Seismoacoustic signatures are produced by above- and below-ground explosions and are often observed at local, regional, and global distances. In the case of an underground explosion, seismic waves propagate to the surface and produce acoustic signatures via pumping of the atmosphere by the ground motion that can be predicted using a Rayleigh integral analysis. Acoustic signature predictions will be discussed and compared with observations from the Source Physics Experiment (SPE) in two scenarios. First, a rigid piston model of the ground motion will be highlighted as a first order model. Second, a more realistic model treating the ground as an elastic medium with finite compressional wave speed will be developed and discussed to demonstrate how such a model changes predicted acoustic signals at local distances. Acoustic signals predicted using each methodology will be compared with observations from SPE to identify how characteristics of the acoustic signal can be leveraged to improve characterization of the underground explosive source.

1:45

4pPAa4. Validation of a generalized least squares beamformer for infrasonic data analysis. Fransiska K. Dannemann, Philip Blom, and Omar Marcillo (Earth and Environ. Sci., Los Alamos National Lab., P.O. Box 1663, MS D446, Los Alamos, NM 87545, fransiska@lanl.gov)

Current seismoacoustic signal detectors including the traditional F-detector, the Progressive Multi-Channel Correlation detector (PMCC), and the adaptive F-Detector (AFD) statistically separate signals of interest from noise based upon a user-defined threshold, however, in regions of high background noise or in the presence of multiple transient signals, a signal's SNR decreases and is often missed by the detector. The adaptive F-detector addresses this problem of coherent noise across array elements by re-mapping a noise threshold over a user-defined window. While application of the adaptive F-detector successfully reduces false detection rates attributed to coherent noise across array elements, the detector is applied post-processing following array analysis using a standard (Bartlett) beamforming approach. Processing of low SNR infrasonic signals can further be enhanced through the application of a generalized least squares (GLS) approach to beamforming which adaptively accounts for background noise characteristics. A characterization of background noise environments will be presented, along with the statistical significance of enhanced detection capabilities compared to traditional beamforming approaches.

2:00

4pPAa5. An investigation of the scattering of infrasonic acoustic waves by turbulent fluctuations generated by the breaking of gravity waves. Roberto Sabatini, Jonathan B. Snively, and Michael P. Hickey (Embry-Riddle Aeronautical Univ., 1400 Hancock Blvd., Daytona Beach, FL 32114, roberto87sabatini@gmail.com)

Infrasonic waves generated below and above the Earth's surface can travel up to ionospheric heights and also reach very large radial distances, spanning from hundreds to thousands of kilometers. As a result, the signals recorded at ground level far from the source location and at high altitudes are strongly influenced by the spatial and temporal variations of the temperature and winds. In most propagation models, acoustic waves are treated as perturbations of a stationary mean atmosphere, which varies only along the vertical coordinate. Hence, horizontal and temporal small-scale fluctuations of temperature and winds induced by gravity waves are inherently excluded by such methods. The objective of the present work is two-fold. First, a model based on the compressible unsteady Navier-Stokes equations, is applied to simultaneously investigate the propagation and breaking of gravity waves and the propagation of infrasonic waves (here emphasizing frequencies in the range [0.001,0.1] Hz) through their induced fluctuations. Second, simulations are performed to investigate the effects of small-scale turbulent inhomogeneities on infrasonic recordings at the ground and within the thermosphere-ionosphere (e.g., by radio remote sensing). More specifically, the influence on the observable signatures are studied, and the interaction between the spectrum of the scattered acoustic waves and the spectral properties of the inhomogeneities is highlighted. Applications to detection of weak natural and anthropogenic signals are discussed.

2:15–2:30 Break

2:30

4pPAa6. Use of the spectral-finite element method for infrasound propagation in a 3D heterogeneous environment. Katrina Burch (USACE-ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, katrina.m.burch@erdc.dren.mil), Michelle E. Swearingen (USACE-ERDC-CERL, Champaign, IL), and Ross E. Alter (USACE-ERDC-CRREL, Hanover, NH)

The use of the spectral element method (SEM), as implemented in the open-source software SpecFEM3D, is explored for the application of longitudinal wave propagation. Infrasound, <20 Hz, propagation using local atmospheric data and a numerical weather forecast model through a heterogeneous environment is examined. The spectral-finite element method simulates acoustic and seismic waves by particle displacement in the earth and velocity potential through the atmosphere. This method also allows for inclusion of air-to-ground coupling plus realistic topography. Modifications to the code that allow for a moving atmosphere are described. Models are designed at local distances no greater than 15 km. Furthermore, high density/velocity regions for lateral heterogeneities are implemented. Absorbing boundary conditions are applied to each of the model's sides. Simulation results are compared to real-world data collected at separate test sites and discussed. This presentation will discuss the applicability of SpecFEM3D to realistic infrasound modeling.

2:45

4pPAa7. Comparison of infrasound emissions observed during a tornado with potential fluid mechanisms. Christopher Petrin and Brian R. Elbing (Mech. & Aerosp. Eng., Oklahoma State Univ., OSU-MAE, Eng. North 218, Stillwater, OK 74078, cepetri@okstate.edu)

Infrasound may be emitted by tornado-producing storms up to 2 h before tornadogenesis, and due to their low atmospheric attenuation, these low frequencies may be detected several hundreds of kilometers away. Therefore, passive infrasound monitoring shows potential for the study and prediction of tornadoes, provided received infrasound signals can be correlated with the flow-fields of tornadoes. Literature indicates that tornadoes do cause high infrasound levels between 0.5 Hz and 10 Hz, but the radially vibrating vortex mechanism commonly proposed to explain this production [Abdullah, *Mon. Weather Rev.* **94**, 213–220] has been shown to be non-physical. Schecter [*Mon. Weather Rev.* **140**, 2080–2089] showed these limitations and showed using numerical experiments of a tornado-like vortex produced infrasound from around the storm's melting level. As this level contains diabatic processes involving hail, it also appears that hail production could be connected to tornado infrasound. In the current work, observations of atmospheric infrasound during a small tornado that occurred on 11 May 2017, will be evaluated in light of these previous studies.

3:00

4pPAa8. Exploring the use of exploding oxy-acetylene balloons for field-scale infrasound. Tracianne B. Neilsen (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Sean Maher (Univ. of California, Santa Barbara, San Diego, CA), Eric J. Lysenko, Julio A. Escobedo, Sarah A. Shaw, Margaret G. McKay, Menley R. Hawkes, Christian A. Lopez, Carla Butts (Brigham Young Univ., Provo, UT), and Robin S. Matoza (Univ. of California, Santa Barbara, Santa Barbara, CA)

Fundamental aspects of volcano-acoustic wave propagation and source effects, such as seismo-acoustic coupling, can be explored with field-scale experiments. Buried explosives are often used for these tests but require special permission and personnel. As an alternative, we explored using an environmentally friendly source of high-amplitude noise: balloons filled with a stoichiometric oxy-acetylene mix detonated with an electric match. A field-scale experiment was conducted to test the efficacy of exploding oxy-acetylene balloons to generate infrasound and modest seismic vibrations. In prior studies with these exploding balloons, the balloons were positioned above the ground to test nonlinear propagation theory [Young *et al.*, *J. Acoust.*

Soc. Am. **138**, EL305–EL310 (2015)] and quantify Mach Stem formation [Leete *et al.*, *J. Acoust. Soc. Am.* **138**, EL522–EL527 (2015)]. In the present experiment, the balloons were placed on the ground in "craters." Specifically, 17 in. oxy-acetylene balloons were exploded in three different "crater" shapes and while sitting on the ground as a control. The explosions were recorded at 100-160 m on colocated infrasound sensors, and broadband seismometers microphones at different heights. The 17 in. oxy-acetylene balloons produce significant infrasound with peak levels at 100 m of 5-13 Pa depending on type of "crater."

3:15

4pPAa9. Overview of ongoing infrasound research at the Georgia Institute of Technology. Alessio Medda, Krish Ahuja, Rob Funk, David Alvord, Darryl Dickey (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, alessio.medda@gtri.gatech.edu), Elliot Dowd (Georgia Tech Res. Inst., Smyrna, Georgia), and John Trostel (Georgia Tech Res. Inst., Smyrna, GA)

This presentation discusses ongoing research in Infrasound technologies at the Georgia Institute of Technology, Georgia Tech Research Institute (GTRI), the applied research arm of the Georgia Tech. In particular, results of a study that compared a number of commercially available infrasound sensors with several windscreen technologies are presented. Among them, comparisons obtained with a wind screen loaned to GTRI by NASA Langley and described by Ahuja and Shams in the 2017 Infrasound Workshop are also presented. Sources producing controlled infrasound under study at GTRI are also discussed. These include a sonic boom simulator, a propane vapor burner, oscillating jets, a nitrogen cannon and a low frequency acoustic driver. In addition, signatures from people moving through doorways are also presented. Each source was most effective in a given frequency range. Controlled infrasound at 0.1 Hz was obtained by several sources, among which a flame and a cold plume modulated at nominal frequencies of 0.1 Hz. Moreover, preliminary results of successful attempts at characterizing the infrasound sources and removing wind noise via wavelet analysis are also presented.

THURSDAY AFTERNOON, 16 MAY 2019 — MCCREARY, 3:15 P.M. TO 5:30 P.M.

Session 4pPAb

Physical Acoustics: General Topics in Physical Acoustics I

Sam Wallen, Chair

The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

3:15

4pPAb1. Comparison of one-way and full-wave linear propagation models in inhomogeneous medium. Petr V. Yuldashev, Pavel B. Rosnitskiy (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova, and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State University, Moscow 119991, Russia, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., oa.sapozhnikov@physics.msu.ru)

One-way wave propagation models based on the parabolic approximation or its wide-angle extensions are often used for describing bounded acoustic beams. Such models are highly demanded when solving nonlinear problems that are computationally intensive and thus technically difficult to solve using full-wave approaches. When describing the propagation of a wave beam in a homogeneous medium, the one-way assumption is fulfilled exactly, and therefore, the inaccuracy is caused only by the limitations of the parabolic approximation. Such an error is significantly reduced within the wide-angle approach and completely disappears when using the exact propagator in the framework of the angular spectrum method. The situation is less obvious in the case of a heterogeneous environment, when a part of the wave energy is inevitably reflected becoming a counter-propagating wave and thus is not taken into account in the one-way approximation. The degree this phenomenon affects the accuracy of the one-way approach is still under discussion. In the current paper, a one-way propagator based on the pseudo-differential wide-angle equation is proposed. The propagator is tested for the homogeneous medium and for several configurations of media with regular and random inhomogeneities. The corresponding solutions are compared with those obtained using the k-Wave toolbox. Results of comparison show how the one-way propagator accuracy depends on the contrast and smoothness of the inhomogeneities. [Work supported by RSF No. 18-72-00196.]

3:30

4pPAb2. Acoustic wave propagation in a toroidal waveguide carrying a mean flow. Charles Thompson, Sarah Kamal, Zaineb Abdulmagid, Eyobel Haile, Samusha Najjuuko, and Carlos Araujo (ECE, UMASS, UMASS Lowell, CACT fa203, Lowell, MA 01854, sarah_kamal@student.uml.edu)

This paper describes the analysis of acoustic wave propagation in an attenuating toroidal waveguide carrying a circumferentially directed mean flow. The disposition of the standing pressure waves driven into resonance by time harmonic excitation is of particular interest. The relationship between the mean flow velocity amplitude and location of peak pressure response relative to the source position is evaluated. The conditions required for non-reciprocal scattering behavior is given.

3:45

4pPAb3. Diffuse ultrasonic transport in an unconsolidated glass bead pack. Richard Weaver and John Y. Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL, r-weaver@uiuc.edu)

We study the transport of diffuse ultrasound with frequencies of hundreds of kHz through random aggregates of d = 3.0 and 1.0 mm diameter spherical glass beads in air under static loads of 100 to 300 kPa. Highdensity polystyrene foam on top and bottom transmits the static loads while maintaining ultrasonic isolation. A floating polystyrene foam wall helps establish a uniform hydrostatic load through the 10 to 70 mm depths. Findings include a band gap extending—for the 3 mm beads—from a lower edge at about 200 kHz (that scales weakly with load and inversely with bead diameter.) Amongst the 3mm beads, we observe an upper edge to the band gap at about 900 kHz corresponding to an optical branch passband associated with the lowest internal resonance of an isolated bead. Higher optical branches are observed also. The lower edge at 200 kHz corresponds well with estimates of the upper band edge for the rotational-wave vibrations of a hexagonal close packed array of beads in Hertzian contact. The observed first arrival times correspond well with Hertzian predictions for low frequency effective longitudinal wavespeeds. Within the low frequency pass band we see diffuse transport, with diffusivities comparable to simple theoretical expectations.

4:00

4pPAb4. Impacted waves in granular media: A laboratory scale asteroids experiment. Thomas Gallot, Gonzalo Tancredi, and Alejandro Ginares (Instituto de Física, Facultad de Ciencias, Universidad de la República, Igua 4225, Montevideo 11400, Uruguay, tgallot@fisica.edu.uy)

Asteroids and small bodies of the Solar System can be considered as agglomerates of irregular boulders, therefore cataloged as granular media. Ejections of particles and dust, resulting in a cometary-type plume, can result from impacts on their surface generating waves within these bodies and potentially causing modifications in the rocks distribution. Since no asteroid seismicity data are available, we propose a laboratory scale experiment of impact-induced seismic waves in granular media. Our study focuses on the influence of static compression mimicking pressure variations induced by self-gravity on the asteroid interior. A cubic box (50 x50 x50 cm) filled with different natural and artificial granular matter is impacted with low velocity projectiles (40 to 200 m/s). An array of accelerometers records the resulting wavefield while the box is compressed to understand its dependence with the monitored internal pressure. This study is relevant to understand how asteroids reacts to kinetic energy, as is will be tested at real scale during the Asteroid Impact and Deflection Assessment mission (2022).

4:15

4pPAb5. A comparison of optical and acoustical resonances: The bisphere telescope. Cleon E. Dean and Maxim Durach (Phys. and Astronomy, Georgia Southern Univ., P.O. Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu)

A previous presentation compared acoustical and optical resonances of a Mie regime double sphere system that focused on a side scattering phenomenon that roughly mimicked a mirror [C. E. Dean and R. M. Hodges, *JASA*, **143**, 1844 (2018)]. If one thinks of these scatterers as lenses the presence of a photonic or phononic "jet" suggests a caustic region with a concentration of energy near the tip of the jet, a point analogous to the focus of a lens. Since both light and sound are reversible, there are two foci on either side of such a scatterer, arrranged symmetrically about each scatterer on the axis of the line between the centers of the two sphere system. The current research examines the case when two variable sized Mie regime scatterers are arranged so as to have the backward focus of a second scatterer on or near the forward focus of the first scatterer. This is effectively a Mie regime double sphere "telescope." Changes to far field scattering in and around the forward scattering direction are examined. This talk attempts to answer these and other questions through the use of theoretical computational acoustics models.

4:30

4pPAb6. An outdoor sound propagation model in concert with geographic information system software. Nathan D. Tipton and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 43040, ntt8@psu.edu)

As industrial technology advances, man-made noise has increasingly contributed to natural environment soundscapes. To predict how this anthropogenic noise can affect these natural environments, engineers build acoustic models over given terrain; however many current models are not compatible with common Geographic Information System (GIS) software, could become outdated due to software version updates, or are written as proprietary packages unavailable to park management. The goal of this study was to create a true open source outdoor sound propagation model compatible with (but not dependent on) outside GIS software. The model was developed to include uneven terrain, atmospheric absorption, screening, wind effects, and ground effects using ISO 9613-2, an international standard for attenuation of sound during propagation outdoors. Given sound source inputs and locations over an input Digital Elevation Map, GIS compatible file types of spatially explicit sound pressure level predictions can be produced by this model. This tool allows for ecologists and park managers to get a better understanding of how anthropogenic noise is affecting soundscapes in natural environments. [Work supported by United Technologies Corporation Professorship.]

4:45

4pPAb7. Effects of perturbing a reference atmosphere on sonic boom propagation and metrics. Lucas Wade and Victor Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, law591@psu.edu)

There is substantial interest in the accurate noise prediction that ranges from sonic boom noise from conventional supersonic aircraft to low-boom noise (a sonic thump sound) from future aircraft designed for quiet flight. A carefully designed reference atmosphere was developed for comparing multiple sonic boom propagation programs. In the current work, that reference atmosphere was perturbed in a number of ways to assess the importance of each perturbation. The code PCBoom was used for this study, and the perturbations were for the temperature, humidity, and wind profiles. One result is that in the absence of winds, perturbing the temperature profile does not substantially affect the metrics on the ground, but perturbing the humidity profile does. [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, Project 41 through FAA Award No. 13-C-AJFE-PSU under the supervision of Sandy Liu. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

5:00

4pPAb8. Spatio-temporal observations of temperature and wind velocity using drone-based acoustic atmospheric tomography. Anthony Finn, Kevin Rogers, Joshua Meade, Jarrod Skinner, and Amir Zhargarian (Defence & Systems Inst., Univ. of South Australia, Bldg. W, Mawson Lakes, Mawson Lakes, SA 5095, Australia, anthony.finn@unisa.edu.au)

This paper reports on a sequence of trials in which the acoustic signature of a small remotely piloted vehicle (drone) has been used to obtain spatio-temporal estimates of atmospheric temperature and wind vectors. Sound fields are recorded onboard the aircraft and by microphones on the ground. Observations are compared and the resulting propagation delays computed for each intersecting ray path transecting the intervening atmosphere. A linear model of sound speed corresponds to virtual temperature and wind velocity, plus tomographic inversion combined with regularisation, then allows vertical cross-sections and volumes of temperature and wind profile to be computed. These two- and three-dimensional profiles are represented as a lattice of elliptical radial basis functions, which enables the medium to be visualised at high levels of resolution. The technique has been used to provide spatio-temporal visualisation of atmospheric dynamics up to altitudes of 1200 m over baselines of 600 m. Independent measurements taken by co-located instruments such as a Doppler SODAR, ZephIR 300 LIDAR

and temperature sensors carried onboard drones flying within the remotely sensed atmosphere show real world performance suggests accuracies of around 0.3 °C, 0.5 m/s and 0.2 m/s for temperature, horizontal and vertical wind speeds respectively may be anticipated. The real world performance also compares very favourably to error envelopes anticipated from propagation models based on large eddy simulation.

5:15

4pPAb9. Preliminary attempts to isolate ground-radiated noise from exploding balloons. Eric J. Lysenko and Traci Neilsen (Phys., Brigham Young Univ., 700 N 56 W, 232, Provo, UT 84604, eric.lysenko@yahoo.com)

Seismo-acoustic coupling occurs when seismic wave propagation creates air-borne acoustic signals. Research is ongoing to determine methods to distinguish between sound due to seismo-acoustic coupling and purely air-borne transmission. In a field experiment, we detonated 17 in. balloons filled with a stoichiometric oxy-acetylene mix placed both on and in the ground. We attempted to isolate ground-radiated waves by constructing a portable soundproof box to deaden air-borne sound wave. The box was constructed from mass-loaded vinyl, soundproofing composite board, liquid nails, and Green Glue. This design incorporated soundproofing through decoupling, absorption, and insulation techniques. Signals observed from a microphone placed in the box are compared with those obtained on microphones outside the box at various heights. The initial blast wave was not evident inside the box. However, the loudest sound measured in the box matches a subsequent portion of signals on microphones near the ground. Testing in a reverberation chamber is done to measure the frequency response in the transmission loss through the box. These results could suggest a viable technique for isolating ground-borne acoustic waves, which could be useful in experiments where calculating the coupling effect is impractical.

THURSDAY AFTERNOON, 16 MAY 2019 CARROLL FORD, 1:15 P.M. TO 5:20 P.M.

Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: Perceptual Consequences of Hearing Loss Across the Lifespan: From Children to Adults (Physiology Meets Perception)

Antje Ihlefeld, Cochair
Biomedical Engineering, New Jersey Institute of Technology, 323 Martin Luther King Blvd., Fenster Hall, Room 645, Newark, NJ 07102

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, Communication Sciences & Disorders, 516 High St., MS 9171, Bellingham, WA 98225

Sarah Verhulst, Cochair
Dept. Information Technology, Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Chair's Introduction—1:15

Invited Papers

1:20

4pPP1. Factors affecting speech-in-noise and speech-in-speech recognition for school-age children with hearing loss. Lori Leibold, Jenna Browning, Ryan W. McCreery (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68124, lori.leibold@boystown.org), and Emily Buss (Univ. of North Carolina, Chapel HIll, NC)

Children with hearing loss require a more advantageous SNR than children with normal hearing to achieve comparable masked speech recognition performance. Although children with hearing loss continue to have difficulties even when appropriately fitted with hearing aids, individual differences are often substantial. This study evaluated the influence of age, degree of hearing loss, aided audibility, and daily hearing aid use on speech-in-noise and speech-in-speech recognition on a sample of 47 children (5-17 years) with bilateral, sensorineural hearing loss. Age-matched children with normal hearing were also tested. Each child completed open-set monosyllabic word recognition in two masker conditions; speech-shaped noise and two-talker speech. Thresholds for 50% correct were measured in each masker using an adaptive tracking procedure. Compared to children with normal hearing, thresholds for children with hearing loss were elevated by an average of 7.4 dB in the noise masker and 6.5 dB in the speech masker. Preliminary results indicate that both age and aided audibility were significant predictors of performance for children with hearing loss in both masker conditions. Hearing aid use was a significant predictor of performance in noise. Degree of hearing loss was not associated with performance in either masker when aided audibility was taken into account.

1:35

4pPP2. Developmental hearing loss in conjunction with early life stress: Perceptual deficits and central auditory correlates. Merri Rosen, Yi Ye, David B. Green, and Michelle M. Mattingly (Anatomy and Neurobiology, Northeast Ohio Medical Univ., 4209 State Rte. 44, P.O. Box 95, Rootstown, OH 44272, mrosen@neomed.edu)

In children with otitis media, the conductive hearing loss (CHL) accompanying infection is a risk factor for later problems with speech perception. These perceptual deficits can persist long after auditory thresholds return to normal, suggesting they may be mediated by changes within the central auditory system. Using animal models of developmental CHL, we have demonstrated perceptual deficits for several temporally-varying signals that comprise speech. Furthermore, these perceptual deficits are correlated with impaired encoding in auditory cortex, indicating that central changes emerge from early auditory deprivation. In our transient developmental CHL model, which mimics the intermittent bouts of hearing loss experienced by children with otitis media, deficits are much alleviated by adulthood. However, early-life stress (ELS) has been described as an additional risk factor for speech problems arising from otitis media. Our data indicate that ELS alone induces deficits in the perception of temporally varying signals. Furthermore, animals experiencing both early transient CHL and ELS have perceptual deficits lasting into adulthood, the magnitude of which is greater than the sum of the individual deficits. These results raise the possibility that early life stress, alone or in conjunction with early CHL, may adverely impact speech perception in humans.

1:50

4pPP3. Consequences of auditory experience and cochlear implant stimulation on tuning and other measures obtained in pre-lingually deaf children and postlingually deaf adults. Julie G. Arenberg (Otolaryngol., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, jbierer@uw.edu), Kelly N. Jahn (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Lindsay A. Devries (Univ. of Maryland, College Park, MD), and Mishaela DiNino (Carnegie Mellon Univ., Pittsburgh, PA)

Children and adults with moderate to severe hearing loss may obtain cochlear implants (CIs) to restore auditory perception, but auditory development differs among them. We have compared various peripheral measures across CI listeners differing in etiology, age of onset and duration of hearing loss, and duration of CI use. Several measures assess the efficacy with which CI electrodes activate their target auditory neurons in individuals with varying hearing demographics. In addition to peripheral contributions to auditory perception, central reorganization might occur when the auditory nerve is stimulated with coarse, electrical input from CIs. Evidence from neurophysiological studies in cats suggest that the central representation of spectral/spatial resolution is altered by chronic CI stimulation. In humans, psychophysical tuning curves might reflect both the spread of electrical current in the cochlea and the central representation of electrical stimuli. Understanding how chronic, electrical stimulation during auditory system development affects spectral resolution may be useful for optimizing CI programming in children and adults.

2:05

4pPP4. Neural correlates of sound-learning experiences in the auditory system: Translational candidates for hearing rehabilitation. Kasia M. Bieszczad (Psych., Rutgers The State Univ. of New Jersey, 152 Frelinghuysen Rd., Psych. Bldg. 224, Piscataway, NJ 08854, kasia.bie@rutgers.edu)

A major disconnect between traditional auditory perception research and recent neuroscience is the high propensity in the auditory system for neuroplasticity. Altered processing of reward-associated sound stimuli can contribute to adaptive behavior, such as hearing, listening, and attending appropriately to sound cues. I will the present work from animal models of learning-induced neuroplasticity in the cortical and subcortical auditory system. The data show how receptive fields and tonotopic maps in primary auditory cortex (A1) as well as the auditory brainstem response (ABR) can change when adult animals trained by pairing a tone with the availability to obtain reward alters sound coding in the auditory system. Over the course of conditioning, increases and reductions, respectively, in ABR amplitude and peak latencies predict how well animals can pick out the learned sound-frequency acoustic cue from other frequencies following conditioning. Furthermore, receptive fields in A1 have narrower tuning for a remembered sound frequency—and only in animals who successfully remember that frequency over others assessed by behavioral test. Therefore, learned sounds are preferentially processed over novel and distractor sounds following conditioning. Significant behavioral preferences for learned tones may be due, in part, to the observed changes in auditory processing across the auditory system.

2:20–2:35 Break

2:35

4pPP5. Development of language, cognition and spatial hearing abilities in children with bilateral cochlear implants. Sara Misurelli (Dept. of Surgery, Div. of Otolaryngol., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, misurelli@surgery.wisc.edu), McKenzie Klein (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI), Christi Hess (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI), and Ruth Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

Cochlear implants (CIs) provide access to sound and the opportunity to communicate through spoken language for individuals who are deaf. Children who receive CIs most often have had little to no acoustic hearing; they have developed, learned, and communicated only via electrical auditory stimulation. In contrast, the majority of adults who receive CIs have had some access to acoustic hearing, either through normal hearing or through amplification via hearing aids. Our work investigates the development of language, cognition, and spatial hearing abilities in children who have bilateral CIs (BiCIs), many from an early age. The following will be discussed: (1) factors that may predict better and faster language development and cognition in children with CIs and (2) development of the ability to use cues for spatial hearing with degraded electrical signals. Our results thus far indicate that children with BiCIs are able to develop the ability to use spatial cues to aid in source segregation, particularly if they receive BiCIs at an early age. In addition, preliminary results reveal that maternal education and IQ may predict development of improved language scores over time for children with BiCIs.

Contributed Paper

2:50

4pPP6. Neural sensitivity to dynamic binaural cues: Human electroencephalogram and chinchilla single-unit responses. Ravinderjit Singh, Hari M. Bharadwaj, and Mark Sayles (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, singh415@purdue.edu)

Animals encounter dynamic binaural timing information in broadband sounds such as speech and background noise due to moving sound sources, self motion, or reverberation. Most physiological studies of interaural time delay (ITD) or interaural correlation (IAC) sensitivity have used static stimuli; neural sensitivity to *dynamic* ITD and IAC is rarely systematically addressed. We used a system-identification approach using maximum-length sequences (MLS) to characterize neural responses to dynamically changing ITDs and IACs in broadband sounds. Responses were recorded from humans (electroencephalogram; EEG) and from single neurons in terminally anesthetized chinchillas (auditory nerve fibers; ANFs). Chinchilla medial superior olive (MSO) responses were simulated based on binaural coincidence from recorded ANF spike times in response to left- and right-channel input. Estimated ITD and IAC transfer functions were low-pass, with corner frequencies in the range of hundreds of Hz. Human EEG-based transfer functions, likely reflecting cortical responses, were also low-pass, but with much lower corner frequencies in the region of tens of Hz. Human behavioral *detection* of dynamic IAC extended beyond 100 Hz consistent with the higher brainstem limits. On the other hand, binaural unmasking effects were only evident for low-frequency ITD/IAC dynamics in the masking noise.

Invited Paper

3:05

4pPP7. Visual influences on auditory spatial processing. Yi Zhou (College of Health Solutions, Arizona State Univ., 975 S. Myrtle Ave., Coor 3470, Tempe, AZ 85287, yizhou@asu.edu)

Sensory experience is the result of a multisensory analysis of the environment around us. When information is properly integrated, visual cues facilitate auditory localization. To investigate the spatial and temporal rules of contingency in multisensory integration, a majority of previous studies have focused on sensory space within the field of vision. But the spaces encoded by vision and audition do not always align with each other. For foveal species such as humans and monkeys, the visual field is restricted to frontal space, whereas the auditory field is panoramic, covering the entire frontal and rear space. The rear sensitivity provided by spatial hearing is critical for avoiding unseen dangers coming from behind. The rear space, however, has been largely overlooked in multisensory research. In this talk, I will present recent work related to vision's role in panoramic spatial hearing in humans, the changes in visual bias observed in human listeners with chronic unilateral hearing loss, and findings concerning visual modulation of spatial responses of single neurons in the marmoset auditory cortex. Based on these results, I will discuss the challenges of implementing existing theories of multisensory spatial perception in neural circuits.

Contributed Paper

3:20

4pPP8. Cortical reorganization following auditory spatial training in listeners with sensorineural hearing impairment: A high-density electroencephalography study. K. V. Nisha and Ajith U. Kumar (Dept. of Audiol., All India Inst. of Speech and Hearing, Manasagangothri, Mysore, Karnataka 570006, India, nishakv1989@gmail.com
)

The present study is intervention-based research aimed at remediation of spatial deficits in listeners with sensorineural hearing impairment (SNHI), through the use of virtual acoustic technology. A mixed group design comprising both within (pre-test, post-test control group design) and across the groups (standard group) comparisons were performed. The study included 37 participants, who were divided into three groups. Groups I and II consisted of SNHI listeners, while group III comprised normal hearing (NH) listeners. The study was conducted in three phases. At the pre-training phase, electroencephalographic (EEG) recordings were acquired from all the three groups using spatial deviants presented in P300 paradigm. Following this, group I listeners underwent virtual acoustic space training (VAST), and post-training EEG recordings were obtained. EEG recordings were also acquired from group II listeners in second evaluation without subjecting them to any formal spatial training. Results of unpaired t-tests, grand average waveforms and scalp topographies of offline processed waveforms revealed significant differences between SNHI and NH listeners. Furthermore, spatio-temporal analyses showed the emergence of new scalp maps in post-training phase in trained listeners and no topographic changes in untrained SNHI group, suggestive of benefit derived from VAST right at the fundamental level (cortical) of spatial processing.

Invited Paper

3:35

4pPP9. Speech-in-noise recognition examined in individuals with normal hearing sensitivity and tinnitus using behavioral and brain imaging methods. Fatima T. Husain (Beckman Inst. of Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, husainf@illinois.edu) and Yihsin Tai (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Although we know tinnitus can cause concentration problems, its contribution to communication difficulties has not been well-studied. In a series of experiments, we have investigated Speech-in-noise (SiN) performance, in those with tinnitus and normal hearing sensitivity using a variety of methods, including behavioral, otoacoustic emissions, structural MRI and in a proposed study, ERP/ EEG. We found that tinnitus patients with normal hearing sensitivity did not have a general speech-in-noise deficit (Tai and Husain, *JARO*, 2018). Instead, our findings indicated that the tinnitus group performed significantly worse only under the 5-dB signal-to-noise ratio (SNR) condition. Additionally, the SiN performance in tinnitus patients was found to be significantly correlated with the perceptual factors related to tinnitus, such as perceived loudness, and was worse in the left ear. We are currently investigating (1) how the left ear appears to be more affected in tinnitus by using structural MRI and (2) whether there is any correlation between tinnitus pitch and consonant recognition. For both the latter studies, we are also contrasting the normal hearing tinnitus group with a hearing loss tinnitus group and other control groups.

Contributed Paper

3:50

4pPP10. Speech auditory brainstem responses in adult hearing aid users: Effects of aiding and background noise, and prediction of behavioral measures. Karolina Kluk, Ghada BinKhamis (Manchester Ctr. for Audiol. and Deafness (ManCAD), The Univ. of Manchester, Oxford Rd., Manchester m13 9pl, United Kingdom, karolina.kluk@manchester.ac.uk), Antonio Elia Forte, Tobias Reichenbach (Dept. of BioEng., Ctr. for Neuro-Technol., Imperial College London, London, United Kingdom), and Martin O'Driscoll (Manchester Ctr. for Audiol. and Deafness (ManCAD), The Univ. of Manchester, Manchester, United Kingdom)

The aim of the study was to investigate the effect of aiding (hearing aids) and background noise on Auditory Brainstem Responses to short consonant vowel speech (Speech-ABRs), and to assess the predictive value of these responses in adults with a bilateral sensorineural hearing loss. Speech-ABRs evoked by a 40-ms [da] were recorded from 98 adult hearing-aid users via loudspeaker stimulus presentation with and without a hearing aid, in quiet and in 2-talker babble using a two-channel vertical electrode montage. Behavioral speech perception in noise and/or aided self-reported speech understanding were assessed. Aided speech-ABRs had earlier peak latencies, larger peak amplitudes, and larger F0 encoding amplitudes compared to unaided speech-ABRs. Background noise resulted in later F0 encoding latencies but did not have an effect on peak latencies and amplitudes, or on F0 encoding amplitudes. Speech-ABRs were not a significant predictor of any of the behavioral or self-report measures. Speech-ABRs are not a good predictor measure of speech-in-noise performance or self-reported speech understanding with hearing aids. However, they may have potential for clinical application as an objective measure of speech detection with hearing aids. [Work supported by EPSRC EP/M026728/1, Saudi Arabian Ministry of Education, NIHR MBRC.]

Invited Papers

4:05

4pPP11. Characterizing noise-induced hidden hearing loss in veterans. Naomi Bramhall (VA RR&D NCRAR, 3710 SW US Veterans Hospital Rd., P5 NCRAR, Portland, OR 97239, naomi.bramhall@va.gov)

Cochlear synaptopathy, the partial loss of auditory nerve synapses onto inner hair cells, has been proposed as a possible source of hyperacusis, some forms of tinnitus, and difficulty understanding speech in background noise. In animal models, cochlear synaptopathy is associated with a reduction in the amplitude of wave I of the auditory brainstem response (ABR) and can occur even when auditory thresholds are normal. This presentation will discuss noise exposure-related changes to several auditory physiological measures, including the ABR, in young military Veterans with clinically normal pure tone thresholds. Veterans show differences from non-Veteran controls even after statistically adjusting for group differences in sex and otoacoustic emissions, suggestive of synaptic or neuronal loss. While these physiological changes do not appear to be associated with decreased performance on standard speech-in-noise tests, they are associated with the report of frequent or constant tinnitus. Although post-mortem histological analysis would be necessary for confirmation, these data are consistent with animal models of cochlear synaptopathy and suggest that synaptopathy or "hidden hearing loss" may occur in response to high intensity noise exposure in humans and be correlated with tinnitus.

4:20–4:35 Break

4:35

4pPP12. Using perceptually relevant speech stimuli for making physiological measurements of the human auditory brainstem. Ross K. Maddox (Biomedical Eng. and Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rm. 5.7425, Rochester, NY 14642, ross.maddox@rochester.edu)

Speech perception is one of the most important functions of the auditory system. The brainstem is an essential part of this process. While studies of natural speech processing typically employ behavioral tasks, and more recently cortical electroencephalography, studies of the human brainstem have been limited (by necessity) to short stimuli like clicks, tonebursts, and single syllables. We recently described a method for presenting continuous naturally uttered speech and deriving the auditory brainstem response. This method makes it possible to create engaging tasks using natural speech while making simultaneous physiological measurements, with applications to a wide range of scientific questions. One such question, with a history of mixed findings, is that of selective attention's role in brainstem processing. We will discuss our work using the speech-derived auditory brainstem response in a two-talker listening task. In keeping with history, our results seem to differ from those of other recent studies using a similar technique. We will also discuss preliminary work at adapting the technique for audiological purposes, in hopes that using speech stimuli will provide a more accurate clinical predictor of speech perception.

4:50

4pPP13. Aging and hearing loss effects on neural speech processing. Samira B. Anderson (Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., 0100 Lefrak Hall, College Park, MD 20742, sander22@umd.edu)

The effects of age-related hearing loss are pervasive, resulting in declines in social, emotional, and cognitive function. As the world becomes a quieter place, the decrease in sensory input from the auditory periphery may trigger homeostatic mechanisms to preserve a stable rate of neural firing at higher levels of the auditory system from brainstem to cortex. For example, decreased inhibitory neurotransmission increases neural excitability, preserving the sensation of loudness for moderate to moderately loud conversational speech levels. However, this change in the balance of excitatory and inhibitory neurotransmission may disrupt the brain's ability to follow the rapid acoustic changes that are characteristic of running speech. Age-related disruptions in auditory processing of synthesized syllables and naturally-produced words and sentences have been demonstrated using electrophysiology (EEG) and magnetoencephalography (MEG). This presentation will review a series of EEG and MEG studies demonstrating effects of aging and/or hearing loss that vary depending on factors associated with type of hearing loss, stimulus choice, and primary neural source (midbrain versus cortex). Clinical implications for hearing loss management will be discussed. [Work supported by NIH-NIDCD, R21 DC015843-01.]

Contributed Paper

5:05

4pPP14. Nano, micro, and macro-scale effects on cochlear tuning. Aritra Sasmal and Karl Grosh (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, asasmal@umich.edu)

Reconciling the highly tuned and nonlinear basilar membrane (BM) response at the base with the nearly low-pass and weakly nonlinear response at the apex has presented a longstanding challenge to cochlear mechanics modelers. Recent experiments have shown that the BM centric view of cochlear mechanics is incomplete and have highlighted the importance of modeling and measuring the dynamics of the organ of Corti (OoC). Here, we describe a new computational model of the guinea pig cochlea that can correctly simulate the response at all frequencies. The model shows that the electromotile force from the outer hair cells modulate the differential motion between the reticular lamina and the BM. Model calculations at the apex show that the geometric taper of the scalae duct as well as the cytoarchitecture of the OoC breaks the scaling symmetry observed at the base. Further, the model predicts that the neural tuning at the base is primarily governed by the macroscopic dynamics of the cochlear partition, while the micro-scale fluid dynamics and the nano-scale channel dynamics dominate the neural tuning at the apex. Overall, the model provides a physiological explanation for the differences between high and low frequency hearing observed in psychophysical experiments. [Work supported by NIH-R01-04084.]

THURSDAY AFTERNOON, 16 MAY 2019 GRAND BALLROOM C, 1:30 P.M. TO 5:30 P.M.

Session 4pSC

Speech Communication: Perception (Poster Session)

Kevin B. McGowan, Chair
Linguistics, University of Kentucky, 1415 Patterson Office Tower, Lexington, KY 40506

All posters will be on display from 1:30 p.m. to 5:30 p.m. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:30 p.m. and contributors of even-numbered papers will be at their posters from 3:30 p.m. to 5:30 p.m.

4pSC1. Auditory sensation of impulsivity and tonality in vocal fry. Philipp Aichinger, Imme Roesner (Dept. of Otorhinolaryngology, Div. of Phoniatrics-LogopeDC, Medical Univ. of Vienna,Waehringer Guertel 18-20, Univ. Klinik HNO, Vienna 1090, Austria, philipp.aichinger@meduniwien.ac.at), and Jean Schoentgen (Dept. of Bio-, Electro- and Mech. Systems, Faculty of Appl. Sci., Université libre de Bruxelles, Brussels, Belgium)

Vocal fry is a phonation type characterized by nearly complete vocal tract damping during the closed glottal cycle phase, caused by a low vocal frequency in combination with a long glottal closed phase. Auditorily, vocal fry is characterized by the sensation of individual glottal cycles. Vocal fry may be (para-)linguistically relevant, but it may also be a symptom of a voice disorder. The aim of the study is to develop predictive models of the presence of vocal fry based on data from auditory experimentation using synthetic stimuli. Predictors are the vocal frequency, the glottal open quotient, and the glottal pulse skewness. The vocal tract is kept constant. Tests are conducted with stimuli that are temporally homogeneous in terms of voice quality, as well as with stimuli that contain neutral-fry-neutral voice quality transitions. Listeners rate tonality, impulsivity, and naturalness of stimuli on 7-point scales, as well as the presence of vocal fry on a dichotomous scale. Results show that perceived vocal fry is correlated with an increase in perceived impulsivity and a decrease in perceived tonality of the voice. The most important predictor of vocal fry is vocal frequency, whereas open quotient and skewness appear to play a minor role.

4pSC2. Effects of talker variability on categorization of spectrally degraded vowels. Emily Dickey and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., University of Louisville, Louisville, KY 40292, eadick01@louisville.edu)

When spectral properties differ between earlier (context) and later (target) sounds, categorization of later sounds becomes biased through *spectral contrast effects* (SCEs). Recent work has shown that talker variability diminishes SCEs: shifts in vowel categorization were smaller when context sentences were spoken by 200 talkers than one talker [Assgari and Stilp, *JASA* (2015)]. CI users' speech categorization is also influenced by SCEs [Feng and Oxenham, *JASA* (2018)] but are known to struggle with talker discrimination. Here, we tested whether talker variability affected context effects in spectrally degraded speech perception. Listeners categorized target vowels varying from "ih" as in "bit" to "eh" as in "bet" following 200 context sentences spoken by one or 200 talkers (from Assgari and Stilp, 2015). Sentences had 5-dB spectral peaks added to low-F1 (100–400 Hz) or high-F1 (550–850 Hz) frequencies (to produce SCEs) then were noise vocoded at different spectral resolutions. At 4 and 8 channels, the experiment was too difficult to produce reliable results (flat categorization functions). At 12 and 24 channels, SCEs occurred but did not significantly differ across one-talker and 200-talker conditions. Talker variability does not appear to affect perception of spectrally degraded speech in the same way it does for normal-hearing listeners.

4pSC3. Mandatory dichotic integration of second-formant information: Mismatched contralateral sine bleats have predictable effects on place judgments in consonant-vowel syllables. Brian Roberts, Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk), and Peter J. Bailey (Psych., Univ. of York, York, United Kingdom)

Speech-on-speech informational masking may arise because the interferer disrupts processing of the target (e.g., capacity limitations) or corrupts it (e.g., intrusions into the target percept). The latter should produce predictable errors. Listeners identified the consonant in monaural three-formant analogues of approximant-vowel syllables, lying along a place-of-articulation continuum ([w]-[l]-[y]). There were two eleven-member continua; the vowel was either high-front or low-back. Continuum members shared F1 and F3 frequency contours; they were distinguished solely by the F2 contour prior to the steady portion. Continuum members also shared amplitude contours and fundamental frequency (130 Hz). Targets were always presented in the left ear. For each continuum, the F2 frequency and amplitude contours were also used to generate interferers with different source properties—sine-wave analogues of F2 (sine bleats) RMS-matched to their buzz-excited counterparts. Accompanying each continuum member with a matched sine bleat in the contralateral ear had little effect, but accompanying each member by a fixed mismatched bleat (1, 6, or 11) produced systematic and predictable effects on category judgments. This outcome indicates that informational masking by interferers involved corruption of target processing as a result of mandatory dichotic integration of F2 information, despite the grouping cues disfavoring this integration. [Work supported by ESRC.]

4pSC4. Influence of semantics on the perception of gender and femininity. Serena Piol and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, sp4864@nyu.edu)

Research on the perception of femininity of a speaker has either focused on acoustic parameters (e.g., F0) or on lexical differences (e.g., mauve/purple). This study simultaneously examined how acoustic parameters and semantic content affect the perception of a speaker's gender and femininity. Three speaker groups (cismen, ciswomen, and transwomen) produced sentences that were categorized as containing masculine lexical items (e.g., The boy gave the football a kick), feminine items (e.g., The little girl cuddled her doll), or neutral items (e.g., Airmail requires a special stamp). Listeners were first asked to identify the gender (male/female) and then asked to rate femininity on a visual analog scale. Results revealed no significant differences in femininity based on the lexical category of the sentences. Consistent with previous findings, average F0 predicted femininity ratings. Despite previous research showing differences in speaking rate for cismale and cisfemale speakers, our data revealed no effect of speaking rate on femininity rating.

4pSC5. The effect of altered sentence rhythm on timing judgments. Dylan V. Pearson, Yi Shen (Speech and Hearing Sci., Indiana Univ.-Bloomington, 200 S. Jordan Ave., Bloomington, IN 47401, dylpears@iu.edu), J. Devin McAuley (Dept. of Psych., Michigan State Univ., East Lansing, MI), and Gary R. Kidd (Speech and Hearing Sci., Indiana Univ.-Bloomington, Bloomington, IN)

Successful speech understanding requires the listener to accurately anticipate the temporal onsets of individual words in running speech. The present study investigated listeners' sensitivity to temporal deviations in sentences with natural or modified speech timing. Subjects listened to sentences in which a portion of speech preceding the final word was replaced by a silent gap. On each trial, an intact sentence was presented, followed by two versions of the sentence with a silent gap: one with the correct timing for the gap (i.e., equal to the duration of the missing speech) and one with altered gap timing (longer or shorter than the missing speech). Listeners judged which version had the altered timing. An adaptive procedure was used to estimate thresholds for the detection of altered timing for early-onset (shortened gap) and late-onset (lengthened gap) final words. In separate conditions, the rhythm of the sentence preceding the gap was either unaltered or rate-modulated according to a sinusoidal modulator. Results showed that the ability to identify the correct gap timing was adversely affected by the manipulation of sentence rhythm, and in both intact and altered rhythmic contexts, listeners were better at detecting early final word onsets than late onsets.

4pSC6. A database of English multisyllabic words for speech perception research. Cody Elston (Commun. Disord. & Sci., Rush Univ., 212 14th Ave., Sterling, IL 61081, codyelston@gmail.com), Brendan Prendergast, Madeleine Thomas, Mark Partain, Elizabeth Butler, Stanley Sheft, and Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ., Chicago, IL)

Recognition of individual words is frequently used to investigate speech intelligibility and underlying perceptual processing. Traditionally, the majority of such studies in English have utilized monosyllabic and, on occasion, disyllabic words and spondees. Although multisyllabic words have

been used extensively to investigate visual processing and lexical organization in reading, little research exists on the auditory perception of spoken multisyllabic English words. The present database was designed to provide materials to facilitate further research into the intelligibility and perceptual processing of spoken multisyllabic words. The database consists of five sections of 1–5 syllable words each. Individual words in each section were generated from the English Lexicon Project website. All words in the database were recorded by a male native speaker of American English, separated into individual word audio files and equalized in root-mean-square energy. Each syllable section contains 1125 words that vary in duration, frequency of occurrence and phonological neighborhood density. With a large number of words in each syllable section, shorter word lists can be selected from each section and matched on specific lexical characteristics. The database recordings are available free of charge for research purposes to improve understanding of perceptual processing of multisyllabic words.

4pSC7. The role of gender expectations on word recognition. Dylan V. Pearson and Tessa Bent (Speech and Hearing Sci., Indiana Univ.-Bloomington, 200 S. Jordan Ave., Bloomington, IN 47401, dylpears@iu.edu)

Socio-indexical and linguistic information bi-directionally interact during speech processing. Information about a speaker's age, gender, or ethnicity, conveyed through speech or visual cues, can influence how acoustic-phonetic cues are mapped to phoneme categories. For example, in McGowan (2015), Chinese-accented English sentences were presented along with a Chinese face (congruent), Caucasian face (incongruent), or no detailed visual information. Intelligibility scores were significantly higher in the congruent than the incongruent condition. Here, we investigate whether similar effects are observed for talker gender. Participants orthographically transcribed sentences mixed with noise from native American English male and female talkers. A gender congruent or incongruent visual face prime was presented before each sentence. In a control condition, different participants completed the task without the inclusion of visual face primes. Results showed that female talkers were significantly more intelligible than male talkers. Further, a gender congruency benefit was observed for female talkers, but not for male talkers. No incongruency cost was found; intelligibility scores in the incongruent and no-face control conditions did not differ. Although congruency effects were only observed with female talkers, the results suggest that expectations about speaker gender can influence word recognition accuracy similar to previously reported ethnicity effects.

4pSC8. Segmental duration as a cue to sentence structure. Sten Knutsen, Karin Stromswold, and Dave F. Kleinschmidt (Psych., Rutgers Univ., 152 Frelinghuysen Rd., Piscataway, NJ 08854, sten.knutsen@rutgers.edu)

In order to parse speech in real time, listeners should use any informative cues available. Here, we investigate the role of segmental duration. Previous work has found statistically significant differences in the mean durations of analogous segments across different lexical/syntactic structures. However, a difference in means does not necessarily mean that the distributions of these durations make individual token durations sufficiently informative to be a useful cue. The goal of this work is to use production data to quantify how informative segmental duration is about syntactic/lexical structure. Our model is based on an ideal listener model, where we assume listeners have implicit knowledge of segmental duration distributions for active and passive sentences. Given these distributions, the model can infer the posterior probability that a particular token belongs to one distribution or the other. After implementing our model in a Bayesian classifier, our results indicate there is indeed sufficient information contained in individual token durations so as to be useful in real-time sentence processing. Furthermore, we modeled listener behavior in a gating task with syntactically ambiguous sentences truncated before disambiguating morphosyntax and achieved 74% accuracy in predicting syntactic outcome, similar to accuracy reported in behavioral studies (62%–84%).

4pSC9. Seeing is believing: The role of the visual stimulus in perception of rounded vowels in Canadian French. John M. Sances (Linguist, Univ. of New Mexico, 1 University of New Mexico, MSC03 2130, Albuquerque, NM 87131, jsances@unm.edu)

For face-to-face communication, the visual stimulus has been shown to be important in speech perception. For vowels, lip protrusion of rounded vowels is the most visually salient signal. Rounding is a contrastive feature of French vowels, both front and back. Tests of native French speakers' lip-reading ability show that front rounded vowels are perceived poorly; the vowel perceived tends to be the back rounded counterpart (Tseva and Cathiard, 1990). Other work (Benoit *et al.*, 1994) has found that native French speakers often perceive the auditory signal for front rounded /y/ as /i/, the unrounded version. Adding visual information to the stimulus drastically increases accuracy. Another study corroborates this in showing that rounding is the least salient perceivable feature in the auditory stimulus, but the most salient in the visual stimulus (Robert-Ribes *et al.*, 1998). The current work extends these findings in a comprehensive experiment using audio, visual, and audio-visual stimuli with the two sources both matched and mismatched. As found previously, rounding was the most salient feature visually. However, rounding was also very salient in the auditory stimulus, contradicting previous research. In the audio-visual mismatched stimuli, listeners tended to favor the auditory signal over the visual signal almost exclusively.

4pSC10. Angry prosody slows responses to simple commands. Aleah D. Combs (Linguist, Univ. of Kentucky, Lexington, KY), Emma Kate Calvert (Commun. Sci. and Disord., Univ. of Kentucky, Lexington, KY), and Kevin B. McGowan (Linguist, Univ. of Kentucky, 1415 Patterson Office Tower, Lexington, KY 40506, kbmcgowan@uky.edu)

Previous research has found that emotional prosody can interact with speech perception and listeners' processing of the meaning of particular word/emotion pairings (Kim and Sumner, 2017). What remains unclear is how this interactive processing can affect behavioral responses such as responses to imperatives. To answer this question, 42 participants were presented with a series of commands read either with angry prosody, happy prosody, or neutral prosody and were instructed to press the requested button on a response box as quickly and accurately as possible. All emotional states were performed by a trained actor, rather than induced, and the stimuli were independently rated for accuracy of performance. On average, participants responded roughly 50ms slower to the commands which were performed with "angry" prosody. There was no difference between responses to "happy" and "neutral" prosody commands. This difference in response time may be due to the heightened neurological responses to angry stimuli (Frueholz and Didier, 2013). These results are consistent with a model of speech perception in which linguistic and social information are processed simultaneously and interactively (Sumner *et al.*, 2014) but not with a model in which emotional aspects of the speech signal or discarded or irrelevant to perception.

4pSC11. Dialect-specific features enhance perception of phonetic imitation of unfamiliar dialects. John P. Ross, Kevin D. Lilley, Cynthia G. Clopper (Linguist, Ohio State Univ., 1712 Neil Ave, Oxley Hall 100, Columbus, OH 43210, ross.1589@osu.edu), Jennifer Pardo (Psych., Montclair State Univ., Montclair, NJ), and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Talkers reproduce speech features of their interlocutors through phonetic imitation. In this study, the effects of experience with a dialect on phonetic imitation and the perceptibility of that imitation were explored. Talkers with New York City and General American accents repeated isolated words after model talkers with New York City and General American accents in a shadowing task. Half of the target words contained phonetic features differing between the two accents, including the stressed vowel in words like

"cauldron, the stressed vowel in words like carriage, and the initial fricative in words like stranger. The other half contained no distinguishing dialect features. Participants from the Midwestern United States completed an AXB task assessing the perceptual similarity of the repeated words to the original stimuli. The results demonstrated that accuracy was above chance overall, suggesting imitation across shadower and model talker accents. Additionally, a significant interaction between the presence of dialect-specific features and shadower dialect was observed: the presence of dialect-specific features facilitated identification of imitations by New York City shadowers, but had no effect on identification of imitations by General American shadowers. These findings suggest that the perception of phonetic imitation of unfamiliar dialects is enhanced by iconic dialect features.

4pSC12. Bidirectional effects of priming in speech perception: Social-to-lexical and lexical-to-social. Dominique A. Bouavichith, Ian C. Calloway, Justin T. Craft, Tamarae Hildebrandt, Stephen J. Tobin, and Patrice S. Beddor (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, dombouav@umich.edu)

Previous perceptual research demonstrates that providing listeners with a social prime, such as information about a speaker's gender, can affect how listeners categorize an ambiguous speech sound produced by that speaker. We report the results of an experiment testing whether, in turn, providing listeners with a linguistic prime, such as which word they are about to hear, affects categorization of that speaker's gender. In an eye-tracking study testing for these bidirectional effects, participants (i) saw a visual prime (gender or lexical), (ii) heard an auditory stimulus drawn from a matrix of gender (female-to-male) and sibilant frequency (shack-to-sack) continua, and (iii) looked to images of the non-primed category. Social prime results replicate earlier findings that listeners' /s-ʃ/ boundary can shift via visual gender information. Additionally, lexical prime results indicate that listeners' judgments of speaker gender can shift with visual linguistic information. These effects are strongest for listeners at category boundaries where linguistic and social information are least prototypical. In regions of high linguistic and social prototypicality, priming effects are weakened or reversed. The results provide evidence of a bidirectional link between social and linguistic categorization in speech perception and its modulation by the stimulus prototypicality.

4pSC13. Perceptual preference for falling tones over rising tones: A study of Mandarin Chinese. Yuyu Zeng, Allard Jongman, Joan A. Sereno, and Jie Zhang (Linguist, The Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, yzengae@ku.edu)

Typological studies have shown that there are more falling tones than rising tones in tone languages, including Chinese. We test the hypothesis that this may be due to a perceptually-based advantage for falling tones over rising tones. Two acoustically comparable (and matched for naturalness) tonal continua in Mandarin (level-falling T1-T4, and level-rising T1-T2) were created. Identification and discrimination results were obtained from 14 native Mandarin speakers. The results revealed that it is easier to identify a falling tone than a rising tone; that is, listeners require a smaller F0 difference between onset and offset to distinguish a falling tone from a level tone as compared to a rising tone from a level tone. Additionally, there are several hints of better discrimination for the falling continuum. This disagrees with our and others' Mandarin production data, which show that the rising tone is closer to the level tone than the falling tone is, hence a production-perception dissociation. We propose that, historically, Chinese listeners' greater sensitivity to the level-falling contrast has resulted in the preponderance of falling tones over rising tones found across Chinese languages, and this proposed explanation may be applicable to other tone languages as well.

4pSC14. Acoustic cues to perception of labialized stops in a merger in progress. John Culnan (Dept. of Linguist, Univ. of Arizona, Tucson, AZ 85721, jmculnan@email.arizona.edu) and Suki Yiu (Linguist, Univ. of Hong Kong, Hong Kong, Hong Kong)

In Hong Kong Cantonese, the labialized and plain velar (/k^w/ and /k/) are undergoing a merger where both may be produced as [k] before the vowel /ɔ/. This study examines the role of acoustic cues to labialization in the perception of velars in Hong Kong Cantonese, and whether these cues are utilized differently in the merger environment than other environments. Native listeners of Hong Kong Cantonese completed an identification task comprised of LPC resynthesized words of Cantonese in the merger environment and three other (unrounded) vowel environments. As anticipated initial results suggest that F2 transition is a less important cue for velars preceding /ɔ/ than those preceding other vowels, although surprisingly F1 transition may only be important for velars preceding /ɔ/. An examination of the importance of intensity suggests that this cue is not used in distinguishing labialized from plain velars. Finally, perception accuracy results on unchanged productions suggest that distinguishing labialized velars from plain velars is a more difficult task in the merger environment than other environments. Taken together, the pilot results suggest that native listeners of Hong Kong Cantonese rely more on multiple acoustic cues in environments where a single cue provides less certain information.

4pSC15. Reading aloud in a clear speaking style may interfere with sentence recognition memory. Sandie Keerstock and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, The University of Texas at Austin, 305 E. 23rd St. CLA 4.400 E9 Mail Code: B5100, Austin, TX 78712, keerstock@utexas.edu)

Previous research has shown that native and non-native listeners' recognition memory is higher for sentences previously heard in clear speech (CS) than in conversational speech (Keerstock and Smiljanic, 2018). The present study investigated whether speaking clearly also enhances *talkers'* sentence recognition memory. The production effect (MacLeod *et al*., 2010) revealed superior retention of material read aloud relative to material read silently during an encoding phase. Vocal production that included an additional distinct element, such as speaking loudly, produced even greater memory benefits than reading aloud (Quinlan and Taylor, 2013). Production of the exaggerated articulatory and acoustic cues in CS could thus lead to enhanced sentence recognition memory relative to conversational style. Native and non-native English speakers produced alternating blocks of 10 conversational and CS sentences for a total of 60 unique sentences. In the test phase, they identified 120 sentences as old (from exposure) or new (distractors). Unexpectedly, preliminary results show lower sentence recognition memory for sentences produced in CS than in conversational speech for both groups of talkers. The results suggest that producing CS, unlike perceiving it, interferes with recognition memory. Allocating cognitive resources to producing hyper-articulated speech may limit their availability for storing information in memory.

4pSC16. Speaker-listener dialect differences and spoken word recognition: Evidence from massive auditory lexical decision. Filip Nenadic, Matthew C. Kelley, and Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

The difficulty associated with perceiving an unfamiliar dialect has been shown in several studies using novel dialects, synthesized vowels, or recorded sentences (e.g., Goslin *et al*., 2012; Wright and Souza, 2012; Maye *et al*., 2007). One of the goals of the Massive Auditory Lexical Decision dataset (Tucker *et al*., 2018) is to investigate the effects of speaker-listener dialect mismatches on spoken word recognition. In three separate auditory lexical decision experiments, monolingual native speakers of English from different dialect regions (231 speakers of western Canadian English recruited in Edmonton, Alberta; 77 speakers of southwestern American English recruited in Tucson, Arizona; and 53 speakers of eastern Canadian English recruited in Halifax, Nova Scotia) each responded to a subset of the same word and pseudoword stimuli recorded by one male speaker of western Canadian English. Therefore, some of the participants had greater experience with the speaker's dialect than others, where the Edmonton participants had the most experience, the Tucson participants had the least, and the Halifax participants were in the middle. We discuss the results of the comparison of responses from these three dialect groups and their implications to speech perception and comprehension of less familiar dialects.

4pSC17. Investigating the comprehension and perception of reduced speech with pupillary response. Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Vincent Porretta (Psych., Univ. of Windsor, Windsor, ON, Canada), and Yoichi Mukai (Linguist, Univ. of AB, Edmonton, AB, Canada)

Spontaneous, casual speech is highly variable, in part due to reduction processes. Listeners handle these reductions in everyday communication; however, these forms present challenges for models of speech perception and lexical processing. Previous research has found that reaction times to reduced word-medial stops are longer, indicating that they are more difficult to process than words with unreduced word-medial stops (Tucker, 2011). The current study examines spoken word processing (as measured by pupil dilation) of reduced and unreduced word-medial stops to determine (a) if the pupillary response to reduced forms corresponds to reaction time results and (b) when in time any differences emerge. Thirty-nine native speakers of North American English completed a listen-and-repeat task in which 80 isolated disyllabic reduced and unreduced word-medial /d/ and /g/ items (40 of each phoneme) were presented. The pupil size data and speech productions are analyzed and will be reported. The results indicate significantly greater pupil dilation for reduced /d/ and /g/. Words containing /d/ elicited greater dilation than those containing /g/, for reduced and unreduced forms. This suggests that, although word-medial stop reduction is frequent in English, an increased processing load is incurred, mirroring previous reaction time results.

4pSC18. Experienced listeners' differential perception of enhancing and non-enhancing cue pairs. Meng Yang (Linguist, Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, mengyang@ucla.edu)

The perception of a pair of enhancing cues, pitch and breathiness, and non-enhancing cues, pitch and vowel duration, were compared in a cue-shifting experiment with Hani listeners who have experience with both cue pairs in the same tense-lax contrast. Results show that, when the correlation between the non-enhancing cues was the reverse of their experience, it was equally difficult for Hani listeners to shift attention from pitch to vowel duration as from vowel duration to pitch. For the enhancing cues, difficulty with cue-shifting was asymmetric; shifting from pitch to breathiness was easy, but shifting from breathiness to pitch was difficult. This asymmetry may occur because the perceptual dependence between pitch and breathiness is unidirectional: breathiness is associated with pitch but not vice versa. Thus, listeners experience interference only when shifting attention from breathiness to pitch. The difference between two cue pairs and the direction of asymmetry found for the enhancing cue pair is consistent with earlier results from English listeners who do not have phonemic experience with either pair. Collectively, these results indicate that enhancing cues are perceived differently from non-enhancing cues, even when listeners' language experience with cue pairs is equated.

4pSC19. Segmental and suprasegmental communicative information: Speech versus emotion recognition thresholds. Shae D. Morgan (Otolaryngol. Head and Neck Surgery and Communicative Disord., Univ. of Louisville, 627 S. Preston St., Ste. 220, Louisville, KY 40292, shae.morgan@louisville.edu)

Effective communication not only depends on what is said (segmental information), but how it is said (suprasegmental information). Research has clearly demonstrated the impact that intelligible segmental information has on communication. Less studied, however, is the availability of suprasegmental information during communicative interactions. Tests of speech recognition or word discrimination are commonly used to assess segmental information in the speech signal. No similar tests have been employed to detect the threshold for detection of suprasegmental information in speech. In this study, we examined thresholds of suprasegmental information (i.e., talker emotional state recognition) and compared them with thresholds obtained for segmental information (i.e., speech recognition). Implications will be discussed, including the availability of suprasegmental features of speech at levels below the threshold of segmental speech recognition. These results suggest that after speech in a signal becomes unintelligible, communication may still occur through the transmission of suprasegmental information, such as the talker's emotional state.

4pSC20. Experimentally testing the epenthetic bias in loanword adaptation. Zachary Jaggers (Linguist, Univ. of Oregon, University of Oregon Linguist, 161 Straub Hall, Eugene, OR 97403, zjaggers@uoregon.edu)

In loanword adaptation, epenthesis is the favored way to make non-native sound sequences pronounceable, over other options like deletion or substitution (Paradis and LaCharité, 1997). This epenthetic bias is also apparent at the phonetic level, such as the phonologization of excrescent bursts and vocoids as full vowels (Kang, 2003; Davidson, 2007). It is possible that loanword status in and of itself induces this bias, whether or not the source form of interest would be illicit in the speaker's native sound system. Weinberger's (1994) Recoverability Principle suggests that second-language learners prefer to preserve or insert sound material due to less awareness of what may be expendable while retaining word recoverability. The epenthetic bias may therefore hold even for sound sequences available in a speaker's native language but which they consider to be embedded in a word from a foreign language. The current study tests this prediction. Listeners transcribe nonce words manipulated along a [CCVC]-[CəCVC] continuum in which, crucially, both ends of the continuum are licit in their native language. Surrounding speech is manipulated between two framings of the nonce word as either an unfamiliar word in the native language or a word from a foreign language to test whether the latter framing induces a preference for <CVC…> transcription. This shines light on the phonetic roots of a common phonological pattern and how contextually mediated these are.

4pSC21. Prediction of listener perception of place-of-articulation in reduced speech. Megan M. Willi (Commun. Sci. and Disord., California State Univ., Chico, 400 W. 1st St., Chico, CA 95929, mwilli@csuchico.edu), Natasha L. Warner (Linguist, Univ. of Arizona, Tucson, AZ), and Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Previous research on stop consonant production found that less than 60% of the stops sampled from a connected speech corpus contained a clearly defined hold duration followed by a plosive release [Crystal and House, *JASA*(1988)]. How listeners perceive reduced, voiced stop consonant variants is not well understood. The purpose of the current study was to investigate whether an acoustic cue called a relative formant deflection pattern was capable of predicting listeners' perceptions of these approximant-like, voiced stop consonants variants. A new methodology motivated by a computational model of speech production was used to extract relative formant deflection patterns from excised VCV segments from a reduced speech database. Participants listened to a total of 56 excised VCV stimuli containing approximant-like, voice stop consonant variants and performed a force choice test (i.e., /b-d-g/). The agreement between the perceptions predicted by the relative formant deflection patterns and listeners' behavioral performance was compared. The expected relative formant deflection pattern correctly predicted listeners' primary response for percent /b/ and /g/ identifications, but not for listeners' percent /d/ identifications. The implications of these results on a possible invariant acoustic correlate for listeners' perceptions of place-of-articulation information will be discussed.

4pSC22. Speech exposure familiarizes listeners with talkers' vocal identity. Jayden Lee and Tyler K. Perrachione (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, jaydenl@bu.edu)

Talker identification experiments typically explicitly train listeners to identify voices. Ecologically, however, listeners learn to identify talkers without explicit practice. Here, we investigated whether listeners gain familiarity with voices even when they are not explicitly identifying them. Participants were assigned to three different exposure tasks with feedback, exposing them to identical stimuli but differentially directing their attention between talkers' vocal identity and verbal content: (1) identifying whether the talker and visual cue on each trial matched; (2) discriminating whether the talker was the same as the prior trial; (3) discriminating whether the speech content matched the previous trial. All three groups were then tested on their ability to learn to identify talkers from novel speech content. Critically, we manipulated whether the talkers during this post-test were the same or different than those heard during training. Regardless of exposure task, listeners were significantly more accurate at learning to identify talkers they had previously been exposed to versus novel talkers. The group that practiced identifying talkers during the exposure phase was only more accurate on exposed talkers. These results suggest that listeners learn talkers' vocal identity during speech perception even if they have not been directed to attend to talker identity.

4pSC23. Investigating the conditions on target-context assimilation in speech sound categorization. Amanda Rysling (Linguist, Univ. of California Santa Cruz, 1156 High St., Dept. of Linguist, Santa Cruz, CA 95064, rysling@ucsc.edu) and John Kingston (Linguist, Univ. of Massachusetts Amherst, Amherst, MA)

Many studies have shown that listeners perceptually differentiate target sounds in categorization tasks from their neighboring context sounds, but some have shown that targets are perceptually assimilated to their contexts. We test the hypothesis that differentiation occurs in context-target order because the context is taken as the criterion for categorizing the target, but assimilation occurs in target-context order because the context's acoustics are parsed as target information. In our experiments, the target was a labial-to-coronal consonant continuum or a front-to-back vowel continuum in VC and CV strings, and the contexts were the other continuum's endpoints. As the second sound in VC or CV, the target differentiated from the preceding context: listeners responded labial or back more often after front vowels and coronal consonants, respectively. With a target V in VC, the target assimilated to the following context: listeners responded back more often before labial consonants. For C in CV, some listeners assimilated the consonant to the following vowel: they responded labial more often before back vowels. Others instead differentiated the consonant from the vowel: they responded coronal more often before back vowels. Follow-up experiments will determine the conditions in which a consonant assimilates to or differentiates from a following vowel.

4pSC24. Speaking rate changes how duration informs phoneme categorization. Andrew Lamont, Rong Yin, Aneesh Naik, and John Kingston (Linguist Dept., Univ. of Massachusetts, 650 N. Pleasant St. ILC 434, Amherst, MA 01003, jkingston000@gmail.com)

Repp *et al. J. Exp. Psychol.* (1978)] reported that for a given duration of fricative noise, a longer silence was required to shift from a fricative to an affricate percept at a slower than a faster speaking rate. We crossed 5 fricative durations (90–208 ms, 29–30 ms steps) by 5 silence durations (0–120 ms, 30 ms steps) by two speaking rates (slow:fast ratio 1.51). Possible responses were grey ship, grey chip, great ship, or great chip. The likelihood of responding ch relative to sh decreased as the fricative lengthened, increased as the silence lengthened, and was more likely at the slow than the fast rate, but neither fricative nor silence duration interacted with speaking rate \— an apparent failure to replicate Repp, *et al.* The likelihood of responding great relative to grey increased with both fricative and silence duration and at the faster than the slower rate. Increasing fricative duration also increased the relative likelihood of responding great more at the slower than the fast rate, but increasing silence duration increased great likelihood less at the slower rate, which indirectly replicates Repp *et al.*, so long as fewer stop responses stand in for fewer affricate responses.

4pSC25. An eye-tracking investigation on the role of categorical perception and acoustic details in the processing of tonal alternations in context. Yu-Fu Chien (Chinese Lang. and Lit., Fudan Univ., Rm. 701, West Wing Guanghuag Bldg., N.220, Handan Rd. Yangpu District, Shanghai, Shanghai 200433, China, chien_yc@fudan.edu.cn) and Jung-Yueh Tu (Ctr. for Int. Chinese Education, Shanghai Jiao Tong Univ., Shanghai, China)

Neutralization is a phenomenon in which two different phonemes are realized as the same sound in certain phonetic environments. In Mandarin, a low-dipping Tone3 is converted to a high-rising Tone2 when followed by another Tone3, known as Third-Tone sandhi. Although previous studies showed statistically differences in F0 between a Sandhi-Tone3 (high-rising) and a Tone2, native Mandarin listeners failed to correctly categorize these two tones in perception tasks (Peng, 2000). The current study utilized the visual-world paradigm in eye-tracking to further investigate whether acoustic details in lexical tone aid lexical access in Mandarin. In the first experiment, we replicated previous studies in that production data of ten disyllabic minimal pairs of Sandhi-Tone3 + Tone3 and Tone2 + Tone3 words showed differences in F0 for the initial tones, but Mandarin listeners' accuracy in identifying them was only around 50%. In the eye-tracking experiment, results showed that proportion of looks to pictures corresponding to Sandhi-Tone3 + Tone3 words was significantly higher when Mandarin listeners heard Sandhi-Tone3 + Tone3 words. A similar pattern was found when auditory stimuli were Tone2 + Tone3 words. The eye-tracking results demonstrated that subtle acoustic details of F0 aid lexical access in a tone language. Mandarin listeners with or without musical training will also be compared.

4pSC26. Auditory integration in the perception of rhoticity. Molly F. Schenker and Anna M. Schmidt (Speech Path. & Aud., Kent State Univ., Speech Pathol. & Audiol., Kent State University, Kent, OH 44242, mdana@kent.edu)

Traditionally, descent of F3 below 2000 Hz at the midpoint has been considered an acoustic correlate for perceived rhoticity. Recent investigations by Heselwood of the auditory integration hypothesis related to rhotics and by Fox and colleagues of "center of gravity" (COG) for stops, diphthongs, and vowels suggested an application to rhotic perception. A resynthesized continuum containing manipulated formant amplitudes to create a high amplitude frequency band descending from above 2000 Hz to below 2000 Hz over 8 steps will be presented to graduate speech pathology students who will judge goodness of rhoticity.

4pSC27. Individual differences in distributional learning for speech: What's ideal for ideal observers? Nicholas R. Monto and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, nicholas.monto@uconn.edu)

Research demonstrates that efficient speech perception is supported by listeners' ability to dynamically modify the mapping to speech sounds to reflect cumulative experience with talkers' input distributions. Here we test the hypothesis that higher-level receptive language ability is linked to adaptation to low-level distributional cues in speech input. Listeners completed two blocks of phonetic categorization for stimuli that differed in voice-onset-time (VOT), a probabilistic cue to the voicing contrast in English stop consonants. In each block, two distributions were presented, one specifying /g/ and one specifying /k/. Across the two blocks, variance of the input distributions was manipulated to be either narrow or wide, reflecting distributions that were relatively more to relatively less consistent, respectively, in terms of how VOT cued the voicing contrast. As predicted by ideal observer computational frameworks, the participants in the aggregate showed steeper identification slopes for consistent compared to inconsistent input distributions. However, the magnitude of learning showed wide individual variability, which was predicted by receptive language ability as measured using standardized assessments. Individuals with poorer receptive language scores showed diminished distributional learning due to a failure to capitalize on consistent input distributions; instead, their perceptual decisions showed instability even the face of acoustic-phonetic certainty.

4pSC28. A deep neural network approach to investigate tone space in languages. Bing'er Jiang, Tim O'Donnell, and Meghan Clayards (McGill Univ., 1085 Dr. Penfield, Montreal, QC H3A 1A7, Canada, binger.jiang@mail.mcgill.ca)

Phonological contrasts are usually signaled by multiple cues, and tonal languages typically involve multiple dimensions to distinguish between tones (e.g., duration, pitch contour, and voice quality, etc.). While the topic has been extensively studied, research has mostly used small datasets. This study employs a deep neural network (DNN) based speech recognizer trained on the AISHELL-1 (Bu *et al.*, 2017) speech corpus (178 hours of read speech) to explore the tone space in Mandarin Chinese. A recent study shows that DNN models learn linguistically-interpretable information to distinguish between vowels (Weber *et al.*, 2016). Specifically, from a low-dimensional Bottleneck layer, the model learns features comparable to F1 and F2. In the current study, we propose a more complicated Long Short-

Term Memory (LSTM) model—with a Bottleneck layer implemented in the hidden layers—to account for variable duration, an important cue for tone discrimination. By interpreting the features learned in the Bottleneck layer, we explore what acoustic dimensions are involved in distinguishing tones. The large amount of data from the speech corpus also renders the results more convincing and provides additional insights not possible from studies with more limited data sets.

4pSC29. Bidirectional decay of auditory memory traces for pitch in speech sounds. Zhanao Fu (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, 1265 Military Trail, Scarborough, ON M1C 1A4, Canada, zhanao.fu@mail.utoronto.ca) and Philip J. Monahan (Dept. of Linguist, Univ. of Toronto, Toronto, ON, Canada)

Previous studies have shown human listeners have greater detection sensitivity to pitch increment than decrement in successive sounds. Assuming deviance detection is based on the comparison between the memory trace of a recent stimulus and the neural representation of a new stimulus, one hypothesis is that this differential sensitivity between increment and decrement is caused by the downward decay of pitch's memory trace. Under the same assumption, the present study found bidirectional—as opposed to the predicted unidirectional—decay of memory traces for pitch in speech sounds by measuring listeners' sensitivity to pitch change over varying time intervals with an AX discrimination task. Three properties in the AX task were randomly sampled from preset ranges: (1) f_0 of the A token (163:320 Hz), (2) difference between the f_0s of A and X (-30:30Hz), and (3) the inter-stimulus interval (ISI; 0:3 s). We found when the stimuli were in the lower portion of the speaker's pitch range, listeners were less sensitive to pitch increments at larger ISIs. Meanwhile, when the stimuli were in the higher pitch range, listeners were less sensitive to pitch decrements at larger ISIs. These results suggest memory traces for pitch in speech sounds decay toward a center pitch.

4pSC30. Directionality in sound change from asymmetries in acoustic distribution. Ollie Sayeed (Dept. of Linguist, Univ. of Pennsylvania, 3401 Walnut St., Philadelphia, PA 19104, sayeedo@sas.upenn.edu)

Following the work of John Ohala, historical sound changes are thought to take place by misperception of the input on the part of the listener. Any account of sound change based on misperception, though, faces a paradox: if X sounds like Y, Y should also sound like X, and yet we often see sound changes that are only attested in one direction. A potential solution is to think of phonetic categories as distributions in acoustic space, and so asymmetries in sound change (X > Y, *Y > X) come from asymmetries in the spread of the distribution of X and Y. If X is a very variable phonetic category with a thick-tailed distribution, a high proportion of its tokens should cross the perceptual boundary and be misperceived as Y; if Y has a narrow distribution, only a very small proportion of its tokens should be perceived as X. We predict that unidirectional sound changes should involve a change from a high-variance to a low-variance category. This experiment tests a case study of asymmetric nasal place assimilation (VnpV > VmpV, *VmpV > VnpV) on a sample of six speakers in three vowel contexts. In the contexts a_a and i_i, the sequence np before the change had a higher-variance distribution of F2 transition (a cue to nasal place) than the sequence mp after the change. In u_u, the sequence after the change had the higher variance. These results give partial support to our hypothesis.

4pSC31. On the articulation between acoustic and semantic uncertainty in speech perception: Investigating the interaction between sources of information in perceptual classification.. Olivier Crouzet (Laboratoire de Linguistique de Nantes (LLING), Université de Nantes / CNRS, Chemin de la Censive du Tertre, Laboratoire de Linguistique de Nantes (LLING), Nantes 44312 Cedex, France, olivier.crouzet@univ-nantes.fr) and Etienne Gaudrain (Ctr. de Recherche en NeuroSci. de Lyon – CRNL, CNRS / INSERM / Université Lyon 1, Groningen, The Netherlands)

Listeners processing speech signals have to deal with two main classes of uncertainty occurring in the vicinity of a given speech segment: both acoustic properties of the contextual environment (Ladefoged and Broadbent, 1957; Sjerps and McQueen, 2013) and lexical hypotheses based on word co-occurence probabilities or semantic relations (e.g., Connine, 1987; Gow and Olson, 2015) may affect the interpretation of a given sound. We investigate this issue by independently manipulating (1) semantic relationships between words using word embeddings estimations and (2) acoustic relations between a contextual part and the final word in the sentence. Based on word pairs that contrast on their vowel target only (e.g., french "balle" versus "belle", pronounced /bal/ vs. /bEl/ – eng. "ball" versus "beauty"), 3 types of sentences are generated: (1) a sentence that would semantically "prime" the word /bal/ ("Le joueur a dévié la", eng. "The player deflected the"), (2) a sentence that would favour the word /bEl/ "Le prince a charmé la", eng. "The prince charmed the"), and (3) a semantically incongruous sentence in both cases "Le journaliste a découvert la", eng. "The journalist discovered the." Listeners are presented with fully ambiguous final words (acoustically located between, e.g., /bal/ and /bEl/) in contexts where semantic influence varies (sentence-types 1/2/3) and is balanced with acoustic manipulations of formant frequencies favouring one word or the other. This will provide cues to modelling how both sources of entropy alter speech perception.

4pSC32. Real-time auditory feedback perturbation of German quantity contrasts. Miriam Oschkinat (Inst. of Phonet. and Speech Processing, Ludwig Maximilan Univ. of Munich, Schellingstraße 3, 80433, Munich 80433, Germany, miriam.oschkinat@phonetik.uni-muenchen.de), Eva Reinisch, and Philip Hoole (Inst. of Phonet. and Speech Processing, Ludwig Maximilan Univ. of Munich, Munich, Bavaria, Germany)

Online auditory feedback (OAF) perturbations have reviewed much about the interplay between acoustic and sensorimotor information during speech production. For spectral manipulations (e.g., formant frequencies), it was shown that people are sensitive to OAF, mainly reacting with a compensation in the opposite direction to the perturbation. This study investigates German speakers' reaction not to spectral but temporal OAF manipulations for the vowels /a/ and /a:/, a phoneme contrast that is realized as a quantity contrast without strong additional spectral cues. Participants were asked to produce the German words *Stab* (/ʃtaːp/ "pole") and *Staat* (/ʃtaːt/ "state") where the vowel was compressed in real-time, or *Stamm* (/ʃtam/ "trunk") and *Stadt* (/ʃtat/ "city") where the vowel was lengthened. While *Staat* and *Stadt* form a minimal pair in German, *Stamm* and *Stab* do not have lexical neighbours. Results showed compensatory responses in the opposite direction to the manipulation for *Staat*, *Stab* and *Stamm* with larger effects for *Staat* (with the lexical neighbour) than *Stab* (without lexical neighbour). Thus, participants react to manipulations of temporal feedback in a similar manner to spectral perturbations. These findings give more precise insight into the link between perception and production in the online-processing of the temporal structure of speech.

4pSC33. Adaptive measurement of crossover frequencies for intelligibility prediction. Nathaniel A. Whitmal (Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, nwhitmal@umass.edu)

In SII theory, frequencies where speech spectra can be divided into two equally-intelligible subbands are called crossover frequencies. These frequencies play a crucial role in SII calculations, and also designate spectral regions that contain important speech recognition cues. Typically, crossover frequencies are found by measuring psychometric curves for speech processed by a series of low-pass and high-pass filters, and then finding the two curves' intersection: an inefficient, time-consuming process. The present study introduces an up/down quantile estimation algorithm that adaptively steers filter cutoff frequencies toward the crossover frequency. Changes in cutoff frequency are governed by comparisons of block trials for low-pass and high-pass filtered speech that meet theoretical requirements for convergence toward the crossover frequency. Preliminary results for trials with nonsense syllables show that the proposed method's estimates match those obtained in published trials using the conventional method. Applications in SII measurements and speech recognition cue measurement will be discussed.

4pSC34. Lexically dependent estimation of acoustic information in speech II: Minimal pair confusability. Charles Redmon and Allard Jongman (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Rm. 427, Lawrence, KS 66046, redmon@ku.edu)

We aim to develop a framework for the analysis of phonetic contrast systems that is fundamentally lexical and does not depend on assumptions of inventory homogeneity and independence of distribution in words and higher-order systems. Previously (Redmon and Jongman, 2018, *JASA*) we reported results of an open-class identification experiment on a 240-word sample of the 26,793-word single-speaker database in Tucker *et al.* (2018). Here, we present results of the second experiment in the project: a 2AFC task where the choice set is limited to obstruent-contrastive minimal pairs. This task forms the opposite end of a continuum from least restricted utilization of acoustic or higher-order information (Exp. 1), to localized attention to a particular contrast in the lexicon. Just as the first experiment provided estimates of a lower bound on listeners' sensitivity to different cues in the signal, the results of this experiment provide an upper bound on those estimates. Participants were presented with 480 stimuli balanced between contrastive obstruents in #CV, VCV, and VC# positions. The results were then used to determine network edge weights on a phonological lexicon on the model of Vitevitch (2008), which emphasizes the interaction between acoustic features, neighborhood topology, and higher-order information in the lexicon.

4pSC35. What *can* make clear speech clear: Lessons learned from the Ferguson Clear Speech Database. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

Extensive acoustic and perceptual analyses have been carried out on the materials from the Ferguson Clear Speech Database (FCSD), which was recorded at Indiana University in 2002. The FCSD consists of 41 untrained talkers reading 188 sentences under instructions first to speak in a manner "as much like your normal conversational style as possible" and later to "speak clearly, so that a hearing-impaired person would be able to understand you." My intent in developing the FCSD was to exploit the expected wide acoustic and perceptual variability among the talkers and use a talker-differences approach to answer the question, "What makes clear speech clear?" In this presentation, I will summarize data from studies of vowel intelligibility, word intelligibility, and perceived sentence clarity along with global and fine-grained acoustic analyses, and discuss how all of these measures are related across the 41 talkers. My hope is that this birds-eye view of the FCSD data will reveal subgroups of talkers in which the talkers adopted certain "profiles" of clear speech acoustic changes that yielded specific helpful perceptual changes. If time permits, I will also review data on perceived talker indexical properties and how they change when talkers speak clearly.

4pSC36. A replication of a test of the metrical segmentation strategy in spoken word recognition. Natasha L. Warner, Seongjin Park (Univ. of Arizona, Dept. of Linguist, University of Arizona, Tucson, AZ 85721, seongjinpark@email.arizona.edu), James M. McQueen (Donders Inst., Radboud Univ., Nijmegen, The Netherlands), Richard A. Southee, Dongdong Zhang, and Iris Lin (Univ. of Arizona, Tucson, AZ)

Norris *et al.* (1995) tested the Metrical Segmentation Strategy (MSS; Cutler and Norris, 1988) as part of the spoken-word recognition model Shortlist. We replicate their study in a different dialect of English, with a different population and items. Norris *et al.* used a word-spotting task, in which listeners had to spot words within speech (e.g., stamp in [stæmpɪdʒ]). Target words were CVCC like champ or CVC like done, and were followed by a full vowel (champ in/tʃæmpoʊʃ/, done in /dʌnʤeɪb/) or a reduced vowel (champ in /tʃæmpəʃ/, done in /dʌnʤəb/). The original study found different behavior for CVCC versus CVC targets, with the results suggesting that listeners hypothesize a word onset at the start of a full-vowel strong syllable (the MSS). Doing so makes it harder to detect champ when it is followed by a full vowel than a weak vowel because the full vowel leads the listener to think the /p/ is the onset of the following word, while the following vowel has little influence for done, where the equivalent consonant is not part of the word. The results for the current study (underway) will show whether these effects generalize across English dialects, listener populations, and words.

4pSC37. Facilitation of speech processing by both expected and unexpected talker continuity. Yaminah D. Carter, Alexandra M. Kapadia, Sung-Joo Lim, and Tyler K. Perrachione (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ycarter@bu.edu)

Speech processing is faster for one continuous talker than mixed talkers. However, it is unknown whether listeners' expectations about talker continuity affect this facilitation. We measured response times during three speeded word identification experiments that manipulated listeners' expectations about talker continuity. First, we manipulated expectations about talker continuity by presenting words in pairs where both words were frequently produced by the same talker (talker-repeat trials) and rarely by different talkers (talker-change trials), or vice-versa. Word identification was faster in talker-repeat trials than talker-change trials, with equal facilitation from both expected and unexpected talker continuity. Unexpected talker changes did not slow processing more than expected changes. Second, a control experiment demonstrated that listeners' expectations about repetitions of the word itself did affect word identification speed. Third, listeners identified words in conditions with one talker, two talkers presented randomly, or two alternating talkers. Word identification was faster whenever the talker was repeated compared to when the talker switched between trials, even if listeners could perfectly predict the talker switch (i.e., alternating-talker condition); talker continuity also facilitated word identification in the random condition. These results provide converging evidence that talker continuity facilitates speech processing in an automatic, feed-forward way, irrespective of listeners' expectations.

4pSC38. Effects of type, token, and talker variability in speech processing efficiency. Alexandra M. Kapadia, Jessica Tin, and Tyler K. Perrachione (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, akapadia@bu.edu)

Phonetic variability across talkers imposes additional processing costs during speech perception, evident in performance decrements for mixed- versus single-talker speech. However, within-talker phonetic variation across different utterances is another, relatively unexplored source of variability in speech, and it is unknown how processing costs from within-talker variation compare to those from between-talker variation. Cognitive consequences of talker variability are also mostly measured from two-alternative forced-choice tasks, whereas naturalistic speech processing occurs in a much larger decision space. Do talker-variability effects scale when both the stimuli and the decision space are more complicated? Here, we measured response times in a speeded word identification task that factorially manipulated three dimensions of speech variability: number of talkers (one versus four), number of target word choices (two versus six), and number of talker-specific exemplars per word (one versus eight). Across all eight experimental levels, larger decision spaces led to significantly slower word identification. Word identification was also slower in conditions with mixed talkers and conditions with multiple exemplars. This pattern of interactions suggests complex processing relationships between type, token, and talker variability and provides preliminary evidence for how both within- and between-talker variability impose additional processing costs in speech perception.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:30 p.m.

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

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	McCreary
Signal Processing in Acoustics (4:30 p.m.)	Beckham
Acoustical Oceanography	McCreary
Animal Bioacoustics	Clements
Architectural Acoustics	French
Musical Acoustics	Breathitt
Physical Acoustics	Jones
Psychological and Physiological Acoustics	Carroll Ford
Structural Acoustics and Vibration	Stopher

Committees meeting on Wednesday

Biomedical Acoustics	Nunn

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Clements
Noise	Segell
Speech Communication	Carroll Ford
Underwater Acoustics	McCreary