Session 4aAAa

Architectural Acoustics, Noise, ASA Committee on Standards, and Speech Communication: Acoustics in Classrooms and Other Educational Spaces

Kenneth W. Good, Chair
Armstrong, 2500 Columbia Ave., Lancaster, PA 17601

Chair's Introduction—7:55

Invited Papers

8:00

4aAAa1. China green campus school standard. Kenneth P. Roy (Innovation, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrongceilings.com) and Xiao Li (Res. and Development, Armstrong World Industries, Shanghai, Shanghai, China)

Since 2011 I've been involved as a principal investigator (Armstrong World Industries) on several research programs encompassing approximately 100 primary and middle grade schools throughout China. Initially, this research was coordinated with Tongji University for the development of the Chinese Green Campus rating system for schools, and later, we have worked with the Beijing Ministry of Education and several additional Universities including Tsinghua Univ., South China Univ. of Technology, Chongqing Univ., and Huaqiao University for the development of the new China GB Green Campus Standard. The "learnings" from these research studies will be reviewed, and the outcomes in terms of implementation will be discussed. A "playbook" was developed for the government architects who design schools, as a simple process for meeting the design goals, and understanding the installed costs for specific use areas. School acoustics is an important IEQ factor in the design and performance of the school in meeting its mission—which is for teachers to teach and students to learn.

8:20

4aAAa2. A review of acoustic research in physical educational settings. Stuart Ryan (Univ. of West Florida, 11000 University Pkwy, Pensacola, FL 32514, sryan@uwf.edu)

This presentation will review past and current acoustic research in physical education settings. Research suggests that the acoustic issues in physical education settings are likely detrimental to student learning (Ryan, 2009a; Ryan, Grube, & Mokgwati, 2010; Ryan & Mendell, 2010). The problem is compounded if there are students with special needs including learning disabilities, attention and auditory disorders, and students who are listening and learning in a non-native language. In addition, typical classroom teachers are faced with many voice concerns; however, physical education teachers are faced with poor acoustically designed gymnasiums, covered areas, and loud outdoor teaching environments which are more challenging than a typical indoor classroom (Ryan, 2009a). Ryan (2010) viewed physical education settings as "hostile listening environments" and have the potential to effect student learning and damaging teacher's voices.

Contributed Papers

8:40

4aAAa3. Acoustical analysis of preschool classrooms. Tina M. Grieco-Calub (Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., FS, 2-231, Evanston, IL 60208, tinagc@northwestern.edu) and Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin-Madison, Wisconsin, United States Minor Outlying Islands)

The present study explores acoustical parameters, including unoccupied and occupied noise levels and reverberation time (RT), in typical preschool classrooms located in the northern suburbs of Chicago. The study was motivated by the following observations: (1) preschool classrooms are often established in buildings that were not initially constructed to be learning spaces; (2) poor classroom acoustics interfere with skills related to academic outcomes including speech perception, serial recall, literacy, and cognitive skills; and (3) younger children are at greater risk for impaired performance in acoustically complex environments. The study was designed to determine whether these preschool classrooms meet existing classroom acoustics standards (ANSI S12.60-2010). Measurements were made with microphones positioned in locations of the classrooms where children typically engage in activities during school season to reflect realistic building operation situations. Unoccupied and occupied noises were recorded over long durations, with intensity and spectral analysis of the recordings conducted off-line. RTs were measured and quantified at the same microphone positions. Preliminary results suggested that none of the preschool classrooms met the recommended unoccupied noise levels, while 3/11 classrooms were found to have longer than recommended RTs. Occupied noise in these preschool classrooms, and implications for learning, will be discussed.

8:55

4aAAa4. Classroom soundscape—Virtual and real. Andy Chung (Smart City Maker, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong, ac@smartcitymaker.com) and W M To (Macao Polytechnic Inst., Macao, Macao)

The sonic environment which both teachers and students are provided in a classroom is important for effective communication during the learning process. How to design and optimize for a classroom requires a study of its soundscape including the feedback of potential users on their perception of the sonic environment. This paper presents how virtual technology is used to characterize the classroom soundscape and how it is related to the reality.

9:10

4aAAa5. Noise in and around schools. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

Incoming noise is usually a major problem for many schools which can drastically affect the teaching process, and often times this is the case due mainly to the traffic and community noise, but also in many cases the schools become themselves large noise sources, affecting the community tranquility. A case study is presented which shows the noise levels and acoustics characteristics of a basic school located in the middle of a middle class suburb, emphasizing the main nuisances that the school receives from the surroundings, and also those produced by the school to the community throughout its normal activities from before opening the school doors in the morning, till the last student has leaved, and during the teaching and leisure activities of young students.

9:25

4aAAa6. Multifunction speech intelligibility in a renovated historic room. Sarah L. Bochat, Michael B. Doing, Peggy B. Nelson (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, bocha007@umn.edu), Ana M. Jaramillo (AFMG Services North America, LLC, Brooklyn Park, MN), and Bruce Olson (Olson Sound Design, Brooklyn Park, MN)

Shevlin Hall, a historic building erected in 1906, is presently home to the department of Speech-Language-Hearing Sciences on the University of Minnesota campus. The primary classroom space is a 20' x 50' room with a high coffered ceiling used for lecture-style classes, discussion groups, and social events. The space underwent renovation in 2013 to address room acoustics and sound design. Post-renovation acoustic measurements and subjective responses reflect favorable outcomes for lecture conditions. Challenges remain for small-group and active discussion classroom arrangements. The current study evaluated predicted and measured speech intelligibility for small-group and active discussions. Preliminary data suggest good intelligibility throughout the room for the condition of lecture-style presentation. Reduced intelligibility is suggested for the condition of talker in back and receiver at instructor or front center position. Implications for sound design will be discussed.

9:40

4aAAa7. Exploring relationships between measured acoustic, lighting, and indoor air quality metrics logged in K-12 classrooms. Kieren H. Smith and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, PKI 210C, 1110 S. 67th St., Omaha, NE 68104, kierenhs@gmail.com)

Comfort in classrooms engenders learning and countless factors contribute to the quality of an educational environment. A longitudinal study measuring classroom conditions, including indoor air quality, lighting, and acoustic components has amassed data from 220 schools in the Eastern Nebraska area. Measurements logged at time intervals no longer than 5 minutes were taken for two full days in each classroom, three times in a year. Using this data, the relationships over time that acoustic factors have with indoor air quality, thermal comfort, and lighting factors are explored. Correlations are investigated between average measured values, time and spatial variation of measured values, and spectral composition of the acoustic metrics. Besides correlations between the assorted measured quantities, other relationships between the design of the building mechanical and lighting systems and resulting acoustic conditions are discussed; for example, do heat pumps or VAV boxes contribute to significantly different acoustic conditions? [Work supported by the United States Environmental Protection Agency Grant No. R835633.]

Session 4aAAb

Architectural Acoustics: Topics in Architectural Acoustics—Measurements, Perceptions, and Metrics

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Peabody Institute: Johns Hopkins Univ., 1 E Mount Vernon Place, Baltimore, MD 21202

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—10:10

Contributed Papers

10:15

4aAAb1. Noise level prediction in a bakery using panel contribution analysis combined with scale modeling. Gong Cheng and David Herrin (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, gongcheng.vac@uky.edu)

Panel contribution analysis combined with scale modeling was utilized to determine the sound pressure level at different positions in a bakery. The primary sources of noise were three heating and air conditioning units. Each of the units was discretized into several patches and the volume velocity for each patch was measured. Transfer functions were measured between the patches and two different customer locations for the full-scale baseline case and using a 1/10 scale model. Transfer functions for the scale model were adjusted to account for air attenuation in the full-scale case. Panel contribution analysis was used to predict noise levels assuming both correlated and uncorrelated sources. Predicted results compared well with measurement. The suggested method can be used to assess the sound pressure levels in large interior spaces before they are constructed so long as the source positions are known. In addition, contributions from the individual sources can be assessed and ranked.

10:30

4aAAb2. Analyzing frequency responses from time-windowed impulse responses across assorted types of room usage. Brenna N. Boyd and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 137 N Jefferson St., Apt A, Papillion, NE 68046, bnboyd@unomaha.edu)

Lokki *et al*. (2015) examined how frequency responses in concert halls change over different time-windowed periods of the impulse response across low, mid, and high frequencies, and connected that with perceived quality in those halls. Time windows were applied to highlight the direct, early, late, or total sound energy, and the resulting frequency responses were overlaid on the same graph to view easily the difference and growth between the time periods across frequencies. In this project, such frequency response analyses across different time-windowed impulse responses are applied to spaces other than concert halls, including a variety of classrooms, worship spaces, and music-centered and speech-centered performance spaces. The objective is to explore the utility of this method of analysis for assorted room usages. In this presentation, interesting cases and trends within individual rooms, within one type of room usage, and between different types of room usage are discussed, as well as this analysis method's advantages and disadvantages.

10:45

4aAAb3. New research on low-frequency membrane absorbers. John Calder (Acoust. Surfaces, Inc., 123 Columbia Court North, Chaska, MN 55318, jcalder@acousticalsurfaces.com)

Low-frequency (LF) room modes are one of the greatest issues for accurate sound recording and reproduction. Effective LF absorbers can mitigate modes in professional and consumer audio rooms. However, fiber- and foam-based absorbers act on sound *velocity*; membrane absorbers act on sound *pressure* (greatest at hard surfaces and corners). Velocity at hard surfaces is zero; thus, fiber and foam absorbers work far less effectively than membrane absorbers under 200Hz. Additionally, most independent testing laboratories are only large enough to accurately measure absorption results above 160Hz (per Schroeder frequency) but not below. Only one lab is large enough to be accurate down to 40Hz. A new LF membrane-based absorber product was designed to compliment the frequency range of an existing product. Both were separately tested for LF absorption down to 40Hz at the above-referenced lab. Ten LF absorber tests revealed that the type of absorber, and its location and orientation in a room, are critical to LF absorber effectiveness. Some unexpected results, however, showed clearly that without standardized laboratory absorption testing in a lab capable of accurately testing down to 40Hz, it is not possible to state conclusively that low-frequency absorber products perform as claimed.

11:00

4aAAb4. A research about influences to piano players' perception and behavior control in brushing up processes of music pieces under the condition of different reverberation time sonic fields. Ayako Matsuo and Takeshi Akita (Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Asahi-cho Senju Adachi-ku, Tokyo 120-8551, Japan, matsuoaya.dendai@gmail.com)

Dividing piano players' proficiency processes of music pieces into three categories of early, middle and final stages, we have studied about the difference and characteristics of their perception for sonic environment and control of performance behaviour at each stage.In our previous studies, it has been shown that we could measure the difference and characteristics of their "perception and behaviour control" on performance mainly at the early and the final stages in brain activity, using NIRS (Near-infrared spectroscopy) method and having some supplementary interviews after the experiments.Moreover in this paper, we carried out experiments about players' "perception and behavior control" on performance at the two stages under some different reverberation time sonic fields using same method.We also

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made sure of the cognition of the players' performances by having some supplementary interviews after the performance experiments.As a result, we could obtain that players' "perception and behavior control" on performance is being influenced from the condition of reverberation time difference of sonic field.

11:15

4aAAb5. Amplified music and variable acoustics—Controlled low frequencies and envelopment at high. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

Subjective ratings from 25 professional musicians and sound engineers were obtained to assess two Danish rock venues of similar size and similar low frequency reverberation times, but different high frequency reverberation characteristics. The musicians rated one hall outstanding and significantly better than the other, substantiating a hypothesis that pop venues can have a longer reverberation time at mid to high frequencies at least in the empty condition with no seats installed. Possible causes for this is an enhancement of the musicians' ease of expressing dynamics as well as of their ability to sense the audience for whom they are playing. It has formerly been shown, that what characterizes the best halls for amplified music is a relatively short reverberation time at low frequencies. However, in this study, a fairly long reverberation time in the 63 Hz octave band is found to be acceptable, especially in the hall with the longer mid-high frequency reverberation time, pointing towards a tolerance for longer reverberation in this band in general. Further, this hints at the possibility that mid-high frequency reverberation can mask, at least 63 Hz band challenges. The 125 Hz octave band RT, which thereby remains the single most important to control for amplified music, is app. 1 sec in the two venues that hold some 700 standing audiences

11:30

4aAAb6. On the prediction of sound diffusion coefficient. Hassan Azad and Gary W. Siebein (Architecture, Univ. of Florida, 3527 SW, 20th Ave, 1132B, Gainesville, FL 32607, h.azad@ufl.edu)

Acoustic diffusers are one of the key elements in architectural acoustic design. The shape, placement and combination with absorbers along with the physical characteristics of the room such as the volume, size, and geometry make a complex design scenario full of variables and dependencies. The Prediction of scattered reflections with the help of computer modeling is one of the most recent areas of research. Currently the only software package that exist for the simulation of the scattered sound is Reflex [1] from AFMG which can predict scattering and diffusion coefficients of 1D acoustics diffusers using their cross sections. It employs Boundary Element Methods for the calculation purposes. This study intends to develop a sound diffusion prediction software that simulates the measurement laboratory environment of the diffusion coefficients in accordance with 17497-2 ISO standards to predict diffusion coefficients of different type of 2D and 3D acoustics diffusers such as Schroeder, fractal and volumetric diffusers. For the calculations, wave-based room acoustics modeling techniques such as FDTD and BEM will be used. There are plenty of partial differential solvers available that are used for acoustics purposes which will be used in this paper as well. [1] http://reflex.afmg.eu/

11:45

4aAAb7. First-order approximation of diffraction at a rectangular rigid plate using the physical theory of diffraction. Aleksandra Rozynova (Acentech, 33 Moulton St., Cambridge, MA 02138, arozynova@acentech.com) and Ning Xiang (Rensselaer Polytechnic Inst., Graduate Program in Architectural Acoust., Troy, NY)

Efficient predictions of sound diffraction around objects is vital in room-acoustics simulations. Recent efforts using the Physical Theory of Diffraction (PTD) [Rozynova, A. and Xiang, N., *J. Acoust. Soc. Am.*, **141**, pp. 3785 (2017)] have been reported on some canonical objects of infinite size. This research has been further developed based on solving diffraction problems on finite, rectangular plates which are free from grazing singularities. Separate solutions are considered for diffraction from two pairs of the rigid plate edges, the edge waves around the plate corner are ignored due to the lower order of diffraction contributions in the far-field. This paper will discuss numerical implementations of the theoretical predictions to demonstrate the efficiency of the PTD in room-acoustics simulations for finite sized rectangular plates. The implemented numerical results will also be compared with some preliminary experimental results carried out with a goniometer.

Session 4aAB

Animal Bioacoustics: Animal Sound Production and Hearing

Micheal L. Dent, Chair
Psychology, University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Chair's Introduction—8:00

Contributed Papers

8:05

4aAB1. Temporal fine structure and zebra finch vocalizations. Nora H. Prior, Edward Smith, Gregory F. Ball, and Robert Dooling (Psych., University of Maryland, 4094 Campus Dr, College Park, MD 20742, nhprior@umd.edu)

The temporal fine structure (TFS) of an acoustic signal refers to the smaller changes within the time waveform, independent of larger envelope changes. Bioacousticians typically have relied on sonographic analyses to describe the primary acoustic features of animal vocalizations and human speech. However, sonographic analyses do not reveal the temporal fine structure in complex vocal signals. Recent research on the perception of human speech by normal-hearing and hearing-impaired listeners has suggested that TFS is important for speech comprehension. Interestingly, birds have much better sensitivity to changes in TFS than do humans, suggesting TFS could also be important for communication in birds. TFS in complex signals is extremely difficult to study in isolation since it occurs along with other acoustic features of complex signals. But, here using a variety of techniques to reduce or eliminate other acoustic features of a complex acoustic signal, we show that zebra finches have particularly good perceptual sensitivity to the natural variation in TFS, and that there are very likely systematic differences in TFS that are unique to various classes of vocal signals. These results suggest that, as with human speech, TFS may be important in avian acoustic communication.

8:20

4aAB2. Calls of the bald eagle (*Haliaeetus leucocephalus*). Edward J. Walsh (Res., Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68132, edward.walsh@boystown.org), Peggy Nelson, Andrew Byrne, Jeff Marr, Christopher Feist, Christopher Milliren, Julia Ponder, Patrick Redig (Univ. of Minnesota, Minneapolis, MN), and JoAnn McGee (Res., Boys Town Natl. Res. Hospital, Omaha, NE)

Injury and mortality statistics suggest that bald eagles entering the air space of wind energy facilities face considerable risk. To mitigate the hazard, acoustic deterrent systems designed to discourage entry into such hazardous air spaces are under consideration. In this study, the acoustic properties of a collection of call types within the eagle vocal repertoire are reported as a first step in a larger program of study designed to assess the deterrent capacity of the bird's natural vocal utterances. To that end, calls were recorded from bald eagles housed in the Raptor Center at the University of Minnesota. Based on preliminary acoustic analyses, at least five calls were identified and are referred to here as peal/scream, chatter/cackle, snort, squeal and grunt. With the exception of the low frequency grunt, calls were uniformly high pitched, tonal in nature, exhibited harmonic spectral structure, and they were generally complex, exhibiting distinct nonlinear characteristics. In this presentation, the spectrotemporal properties of each call type will be described with the goal of generating an acoustic repository to enable the consistent classification of calls to be tested for their potential as deterrence signals. [This work was supported by Department of Energy grant #DE-EE0007881.]

8:35

4aAB3. Use of transition and abstract rules by budgerigars in auditory pattern processing. Adam Fishbein, Gregory F. Ball, and Robert Dooling (Neurosci. & Cognit. Sci., Univ. of Maryland, College Park, 4123 Biology-Psych. Bldg., University of Maryland, College Park, MD 20742, afishbei@umd.edu)

Budgerigars, a small species of parrot, are open-ended vocal learners that produce a long, rambling warble song. Past work has shown that these birds are sensitive to changes in the order of their song elements but it is not clear what rules they may use to process changes in sounds sequences. Here, we used operant conditioning and psychophysical techniques to ask how budgerigars discriminate among patterned auditory sequences. In these experiments, auditory sequences of the form AAB, composed of pure tones or natural warble elements, were played to the bird as a repeating background until the bird pecked a key to initiate a trial. During a trial, either a target sequence played that differed from the background in the order of the elements or a sham trial occurred where there was no change. Results show that budgerigars use rules about the transitions between elements to detect targets. Further experiments using multiple background sequences show that the birds are also sensitive to the underlying abstract pattern of sounds (same-same-different). Thus, with their ability to attend to surface transition rules (phonology-like) and deeper abstract patterns (syntax-like), budgerigars are a useful vocal learning model for studying the neurobiology and evolution of language.

8:50

4aAB4. Ultrasonic vocalization production by socially isolated and experienced mice. Laurel A. Screven and Micheal L. Dent (Psych., Univ. at Buffalo, B80 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu)

Mice produce ultrasonic vocalizations (USVs) in a wide variety of social situations, including courtship, investigation, and territorial defense. Although production of USVs is believed to be innate, suggesting mice do not need acoustic experience with conspecifics to produce calls, it is possible mice require experience to emit calls in the proper behavioral context. Male mice require social experience with a female to produce USVs in response to olfactory signals present in dirty bedding (e.g., Nyby *et al.*, 1983), and isolated male mice will produce more USVs than socially experienced males (Keesom *et al.*, 2017), however the influence of social experience on vocal production by female mice has yet to be investigated. The present experiment aimed to determine if social isolation or experience with conspecifics influences the vocal repertoire of adult female CBA/CaJ mice. Mouse calls were recorded during exposure to an unknown male or female mouse and analyzed to determine if isolation or experience affected call number, rate, or spectrotemporal complexity. Chronic social isolation did influence the vocal repertoire, suggesting that social experience plays a role in the production of USVs by female mice.

9:05

4aAB5. The effects of noise and age on the detection of ultrasonic vocalizations and pure tones by laboratory mice. Anastasiya Kobrina and Micheal L. Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, SUNY University at Buffalo, Amherst, NY 14261, akobrina@buffalo.edu)

Mice are frequently used to study and model age-related hearing loss (ARHL) due to similarities in human and mouse cochleae and in genetic makeup. Mice emit ultrasonic vocalizations (USVs) that may be used for communication. Behavioral research has established that mice exhibit ARHL for detection of USVs and pure tones, however later in life than previously estimated by electrophysiological methods. In addition, mice showed sex-related ARHL differences for USVs but not for 42 kHz tones. In order to fully understand whether mice can be used to model ARHL it is critical to examine hearing loss for simple and complex stimuli in a variety of listening environments. In this study, detection thresholds were measured in adult mice using operant conditioning procedures and positive reinforcement. Mice were trained to detect USVs and high frequency pure tones in silence and in 50 dB white noise. The results revealed that adult mice retain hearing late into their lifespan, with very old mice exhibiting higher thresholds for USVs and pure tones in noise than in silence. The results highlight the importance of measuring hearing in awake, trained, behaving subjects when comparing ARHL across different species. [This work was supported by F31 DC016545 and R01 DC012302.]

9:20

4aAB6. The use of background noise to mask objectionable sounds to rodents. Robert D. Bruce, Arno S. Bommer, Isaac Harwell, Adam Young, and Max Meynig (CSTI Acoust., 16155 Park Row # 150, Houston, TX 77084-6971, bob@cstiacoustics.com)

Construction noise and vibration can interrupt sleep/concentration of humans during renovations. Noise treatments or vibration breaks can reduce the impact. But could background noise cover up a moderate level of construction noise? Innocuous sounds are used to mask quieter but more annoying sounds. The concept is commonly used in offices to aid in concentration and reduce annoyance in places where people work or sleep. But could it be used to mask construction noise if the recipients were rodents rather than people? In theory, the concept of sound masking should be valid for other species of animals. Excessive noise and vibration has been shown to affect rodent behavior including breeding. Their hearing, extending into the ultrasound region, complicates efforts to recognize and measure the rodents' acoustical environment. During the past several years, we have had occasion to study the literature on rodents' hearing and the use of masking to prevent the creatures from being bothered by intruding sound. In this paper, we summarize the literature and investigate the concept of masking for rodents with respect to regularly encountered sound sources and sounds from construction activities.

9:35

4aAB7. Optical coherence tomography reveals complex motion between the basilar membrane and organ of corti in the gerbil cochlea. Clark E. Strimbu (Dept. of Otolaryngol., Head and Neck Surgery, Columbia Univ. Medical Ctr., P&S 11-452, 630 West 168 St., New York, NY 10032, ces2243@cumc.columbia.edu), Nathan C. Lin (Dept. of Elec. Eng., Columbia Univ., New York, NY), and Elizabeth S. Olson (Dept. of Otolaryngol., Head and Neck Surgery, Columbia Univ. Medical Ctr., New York, NY)

The mechanical basis of auditory transduction within the mammalian cochlea is not fully understood but relies on complex mechanical interactions between different structures within the Organ of Corti (OoC). We used spectral domain phase microscopy, a functional extension of optical coherence tomography (OCT), to measure the sound-evoked motion in response to "Zwuis" frequency complexes, *in vivo*, in the basal region of the gerbil cochlea. Because OCT allows for measurements from multiple axial positions along the optical path, we are able to simultaneously record vibration at different points within the OoC between the basilar membrane (BM) and tectorial membrane. Imaging through the round window membrane, we could record from several longitudinal locations along the cochlea and at each location, we measured from several radial positions. Both the BM and intra-OoC vibrations are tuned and show nonlinear amplification, or compressive growth, with the applied sound pressure level. Similar to recent reports (Ren *et al*, PNAS 2016; Ren and He, MoH 2017), we find that the motions of surfaces within the OoC can exhibit a greater degree of compressive nonlinearity (amplification) and physiological vulnerability than the motion of the BM. Additionally, some responses show sharp notches near the onset of the nonlinear regime.

9:50

4aAB8. Copulation calls of female chimpanzees (Pan troglodytes) signal fertility. Anthony P. Massaro (Ecology, Evolution, and Behavior, Univ. of Minnesota, 1987 Upper Buford Cir, Falcon Heights, MN 55108, massa060@umn.edu), Emily E. Boehm (Evolutionary Anthropology, Duke, Durham, NC), Elizabeth V. Lonsdorf (Psych. and Biological Foundations of Behavior Program, Franklin & Marshall College, Lancaster, PA), Carson M. Murray, Margaret A. Stanton (Ctr. for the Adv. Study of Human Paleobiology, The George Washington Univ., Washington DC, DC), and Michael Wilson (Anthropology, Univ. of Minnesota, Minneapolis, MN)

In many species, females produce vocalizations during or soon after mating. In chimpanzees, these calls accompany only some matings while visual signals of sexual receptivity are always visible. Proposed explanations for chimpanzee copulation calls include (1) increasing paternity confusion and thus reducing infanticide risk, and (2) competing more effectively with other females for male mating interest. We used information theoretic model selection to test competing hypotheses using a record of 1,168 copulation events recorded by the mother-infant project team at Gombe National Park, Tanzania (1976-2015). We found that females produced copulation calls more frequently in the last week of their reproductive cycle, when they are most fertile. This timing is consistent with the hypothesis that copulation calls function as an honest signal of fertility. While females also signal fertility visually (with sexual swellings), copulation calls may signal to a broader audience, encouraging males to invest in mating effort. Male primates are often portrayed as always being ready to mate, but in reality, mating entails potential costs for males, including the risk of aggression from other males, and opportunity costs that must be weighed against the risk of mating with a non-ovulating female.

10:05–10:20 Break

10:20

4aAB9. Vessel sound cuts down communication space for vocalising fish and marine mammals. Rosalyn L. Putland (Leigh Marine Lab., Inst. of Marine Sci., Univ. of Auckland, 160 Goat Island Rd., Leigh, Auckland 0985, New Zealand, rput037@aucklanduni.ac.nz), Nathan D. Merchant, Adrian Farcas (Ctr. for Environment, Fisheries and Aquaculture Sci., Lowestoft, Suffolk, United Kingdom), and Craig A. Radford (Leigh Marine Lab., Inst. of Marine Sci., Univ. of Auckland, Warkworth, New Zealand)

Anthropogenic noise across the world's oceans threatens the ability of vocalizing marine species to communicate. Some species vocalize at key life stages or whilst foraging, and disruption to the acoustic habitat at these times could lead to adverse consequences at the population level. To explore the risk of such impacts, we investigated the effect of vessel noise on the communication space of the Bryde's whale *Balaenoptera edeni*, an endangered species which vocalizes at low frequencies and bigeye *Pempheris adspersa*, a nocturnal fish species which uses contact calls to maintain group cohesion while foraging. By combining long-term acoustic monitoring data with AIS vessel-tracking data and acoustic propagation modelling, a quantitative method for determining the impact of vessel noise on their communication space was established. Routine vessel passages cut down communication space by up to 61.5% for bigeyes and 87.4% for Bryde's whales. The influence of vessel noise on communication space also exceeded natural variability between 3.9 and 18.9% of the monitoring period. To combat potential effects of vessel sound, we propose the application or extension of ship speed restrictions in ecologically significant areas, since our results indicate a reduction in sound source levels for vessels transiting at lower speeds.

10:35

4aAB10. Analysis of Haddock (*Melanogrammus aeglefinus*) sounds recorded in the Northwest Atlantic. Xavier Mouy (School of Earth and Oceans Sci., Univ. of Victoria, PO Box 1700, Victoria, BC V8W3P6, Canada, Xavier.Mouy@jasco.com), Rodney A. Rountree (Biology, Univ. of Victoria, Waquoit, MA), Katie A. Burchard (Northeast Fisheries Sci. Ctr., NOAA Cooperative Res. Study Fleet Program, Narragansett, RI), Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada), and Stan E. Dosso (School of Earth and Oceans Sci., Univ. of Victoria, Victoria, BC, Canada)

Haddock (*Melanogrammus aeglefinus*) are present on both sides of the North Atlantic. Their distribution in the Northwest Atlantic ranges from Greenland to North Carolina. They are an important food resource that needs to be closely monitored to ensure a sustainable fishery. Research studies have reported that both male and female haddock produce sounds during courtship and spawning. These sounds can be used to monitor spawning activities non-intrusively and at large scale. The objective of this paper is to analyse the spatial and temporal occurrence of Haddock sounds in the Gulf of Maine. Passive acoustic data were collected in 2003, 2004, 2006, and 2007 in areas known to contain spawning haddock. To analyze the large amount of data collected, an automated haddock sound detector was developed based on a measure of kurtosis and the Dynamic Time Warping algorithm. The detector was trained using the 2006 and 2007 data and its performance was quantified and optimized by comparing detection results with manually annotated Haddock sounds. The detector was then used to analyze data collected in 2003 and 2004. Results provide information on the temporal and spatial distributions of courtship and spawning sounds.

10:50

4aAB11. A frog's lungs sharpen the frequency tuning of its peripheral auditory system: A pulmonary paradox resolved? Mark Bee (Dept. of Ecology, Evolution, and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, mbee@umn.edu), Norman Lee (Dept. of Biology, St. Olaf College, Northfield, MN), and Jakob Christensen-Dalsgaard (Biology Inst., Univ. of Southern Denmark, Odense M, Denmark)

In frogs, sound localization is facilitated by the directionality of their internally coupled ears, which are connected through the mouth cavity and Eustachian tubes. Acoustic input through the frog's body wall and lungs also shapes the ear's directionality. One hypothesized function of the lung input is to enhance directional hearing to improve localization of conspecific calls. However, an unresolved paradox exists: the lung input often improves directionality at frequencies that are intermediate between those emphasized in conspecific vocalizations. The present study of green treefrogs (*Hyla cinerea*) may resolve this paradox. Using laser vibrometry, we show that the lung input improves directionality at frequencies emphasized in conspecific calls by just 2 dB or less. In contrast, the lung input reduces the frontal field sensitivity of the tympanum by up to 7 dB in response to frequencies that are intermediate between those emphasized in conspecific calls and to which the two inner ear papillae are tuned. These results suggest the lung input may function in communication by sharpening the frequency tuning of the peripheral auditory system, which in turn could filter out frequencies in heterospecific calls in some mixed-species choruses. How the lung input achieves this "noise cancellation" is currently under investigation.

11:05

4aAB12. Temporal information processing in gray treefrogs: Compute within frequency then integrate across frequency. Saumya Gupta and Mark Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, gupta333@umn.edu)

In Cope's gray treefrog (*Hyla chrysoscelis*), the pulse rate of advertisement calls is a species recognition cue. This temporal information is redundant across two spectral bands in the call's bimodal spectrum that are primarily transduced by different inner ear papillae. Previous studies have shown that some neurons in the frog auditory midbrain are selective for bimodal spectra and that, compared with unimodal spectra, calls with bimodal spectra are more attractive. In addition, "interval-counting" neurons in the frog auditory midbrain are selective for conspecific pulse rates. However, it is not known whether pulse rate information is computed prior to, or subsequent to, spectral integration. We conducted behavioral experiments to investigate the serial order of pulse rate computations and spectral integration. In phonotaxis tests that used bimodal and unimodal stimuli with variable pulse rates, we tested the hypothesis that spectral information is integrated prior to pulse rate computations versus the alternative hypothesis that pulse rate can be computed independently in different frequency channels prior to spectral integration. Our results support the latter hypothesis, and suggest pulse rate can be computed independently in separate frequency channels. These results give insight into the mechanisms of temporal processing underlying the perception of key biological information.

11:20

4aAB13. Within-individual variation in signal production limits female choice in noisy treefrog choruses. Jessie C. Tanner (Ecology, Evolution, and Behavior, Univ. of Minnesota - Twin Cities, 1479 Gortner Ave, 140 Gortner Labs, Saint Paul, MN 55108, tanne123@umn.edu) and Mark Bee (Ecology, Evolution, and Behavior, Univ. of Minnesota - Twin Cities, St. Paul, MN)

In acoustically communicating animals such as anurans and orthopterans, individual signalers can display remarkable variation in signal traits, even within a single bout of calling. This within-individual variation has the potential to "mask" the between-individual variation that is the target of sexual selection. Thus, receivers may not always be able to discriminate among males based on properties of their acoustic signals, even when female choice is well documented in the lab. The effect of within-individual variation may be compounded by noise generated by breeding choruses of conspecifics, which can prevent signal detection or discrimination. We manipulated within-individual variation in two signal traits (call rate and pulse rate) that are critically important for mate choice in *Hyla chrysoscelis*, Cope's gray treefrog. We replicated our design at three realistic levels of chorus noise. Within-individual variation in signal traits reduced the strength of sexual selection on one of two traits, and receivers were less likely to discriminate against lower-quality signals under conditions of realistic chorus noise. Our results suggest that preference functions generated using stimuli that lack within-individual variation, under ideal listening conditions, may chronically over-estimate the strength of sexual selection on signal traits in natural populations.

11:35

4aAB14. Effect of broadband chirp signal on the sonic behavior of the /kwa/ fishes inhabiting Posidonia oceanica seagrass meadow. Jean-Pierre Hermand and Laura Velardi (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, Brussels 1050, Belgium, lvelardi@ulb.ac.be)

Early experiment to study acoustics of Posidonia oceanica seagrasses (USTICA99) transmitted acoustic chirp signals (0.2-16 kHz, 15.8 s) every minute during 3 days over a meadow. Receiving hydrophones recorded vocalizations from unknown fish inhabitants, which sound like /kwa/. Here, we address the impact of our measurements on this fish population. The authors' ears analyzed the recordings to identify all fish vocalizations and attempt to recognize /kwa/ individuals based on timbre. A comparative analysis of the /kwa/ sound production within time intervals between and during chirp transmission was made to understand if and how the sonic behavior is affected. As in few similar studies for other anthropogenic noise sources on a limited number of species, there was an observable effect. The major part of the vocalization patterns that were identified within the chirp time interval started near the onset of the frequency sweep, which first spans the range of /kwa/ sound production (0.4-2 kHz). The number of single vocalizations per unit time during that period was close to that during the no-chirp time interval while during higher swept frequencies was markedly smaller. This suggests the fish detection range may extend toward ultrasound to avoid dolphin predators. Quantitative results will be presented.

Session 4aBA

Biomedical Acoustics: Acoustic Imaging of Small Vessels and Low Speed Flow

Mahdi Bayat, Chair
Biomedical and physiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Chair's Introduction—8:00

Invited Papers

8:05

4aBA1. Ultrasonic non-contrast perfusion imaging with adaptive demodulation. Jaime Tierney and Brett Byram (Biomedical Eng., Vanderbilt Univ., 2301 Vanderbilt Pl.; PMB 351631, Nashville, TN 37235, jaime.e.tierney@vanderbilt.edu)

Ultrasonic non-contrast perfusion imaging remains challenging due to spectral broadening of the tissue clutter signal caused by patient and sonographer hand motion. Simply, the velocity of the slowest moving blood has similar spectral support to the moving tissue. To address this problem, we developed an adaptive demodulation (AD) scheme to suppress the bandwidth of tissue prior to high-pass filtering. The method works by directly estimating the modulation imposed by tissue motion, and then removing that motion from the signal. Our initial implementation used single plane wave power Doppler imaging sequence combined with a conventional high-pass IIR tissue filter. However, other recent advancements in beamforming and tissue filtering have been proposed for improved slow-flow imaging, including coherent flow power Doppler (CFPD) and singular value decomposition (SVD). Here, we aim to evaluate AD separately as well as in comparison and in conjunction with improvements in beamforming and filtering using simulations and an *in vivo* muscle contraction experiment. We show that simulated blood-to-background SNR is highest when using AD + CFPD and a 100 ms ensemble, which resulted in a 9.88 dB increase in SNR compared to CFPD by itself. Additionally, AD + SVD resulted in a 54.6% increase in mean power within the *in vivo* muscle after contraction compared to a 6.84% increase with AD and a conventional IIR filter.

8:30

4aBA2. Enhanced sensitivity of flow detection in small vessels with slow flow by coherent-flow power Doppler. Jeremy J. Dahl (Radiology, Stanford Univ., 3155 Porter Dr, Palo Alto, CA 94304, jeremy.dahl@stanford.edu)

Power Doppler imaging is a commonly used method for the detection of flow. However, the sensitivity of power Doppler in small vessel detection is limited in clinical ultraound scanners by a combination of factors, including the small-diameter of vessels, slow flow through the vessels, the size of the Doppler ensemble, the presence of motion, the presence of noise, and the type of filter used. We have developed an alternative power Doppler imaging technique that enhances flow detection of small vessels and slow flow. This technique utilizes the same principles as power Doppler, but uses the coherence of backscattered blood signal rather than the backscattered signal power. We show that this technique, called coherent flow power Doppler (CFPD), is less sensitive to clutter-based signals and thermal noise and improves signal-to-noise ratio of the flow signal. We show through simulations, phantoms, and *in vivo* experimentation in human liver that this additional signal-to-noise ratio (7.5-12.5 dB) can be utilized to detect slower flow, improved frame rate (smaller Doppler ensemble), or improve the Doppler image profile. In addition, the technique demonstrates better signal-to-noise ratio for small-sized vessels (less than 2 mm).

8:55

4aBA3. Spatiotemporal clutter filtering methods for *in vivo* microvessel imaging. Pengfei Song and Shigao Chen (Dept. of Radiology, Mayo Clinic College of Medicine, Med. Sci. 1-26, 200 First St. SW, Rochester, MN 55905, song.pengfei@mayo.edu)

Ultrafast plane wave imaging enables acquisition and accumulation of a large number of Doppler ensembles in a short period of time, which substantially boosts ultrasound Doppler sensitivity by at least an order of magnitude. As a key component, spatiotemporal clutter filtering based on singular value decomposition (SVD) has recently demonstrated superior performance in tissue clutter suppression and microvessel signal extraction over conventional high-pass-filter-based clutter filtering techniques. This presentation introduces several novel SVD clutter filter techniques from our recent work, focusing on addressing the technical challenges (complex tissue motion, high computational cost, and noise) of SVD clutter filter during clinical translations of the ultrafast microvessel imaging technology for human imaging. We present a block-wise adaptive SVD filter method that can robustly threshold singular values on a local level, a real-time feasible SVD clutter filtering method based on partial SVD and randomized spatial downsampling, and a noise equalization and suppression method that can significantly improve microvessel imaging quality at a low cost. *In vivo* performance of these methods will be demonstrated in different organs.

9:20

4aBA4. New sampling and filtering approaches to label-free ultrasonic perfusion imaging. Yang Zhu (Dept. of BioEng., Univ. of Illinois at Urbana-Champaign, B412 Beckman Inst., 405 North Mathews Ave., Urbana, IL 61801, yangzhu2@illinois.edu), Minwoo Kim (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Michael F. Insana (Dept. of BioEng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Color-flow and power Doppler imaging are safe, low-cost, non-invasive techniques for assessing blood flow throughout the body. Imaging microvascular flows and perfusion generally requires injectable contrast agents to separate tissue clutter from blood echoes. Our innovation is to modify echo acquisition to expand data dimensionality and increase sensitivity of non-enhanced PD imaging to very slow, disorganized flows. To separate perfusion from other echo signals, we developed a higher-order singular value decomposition filter that appropriately reduces data dimensionality. Effective elimination of clutter and noise results improved perfusion specificity. Our methods are applied to existing commercial US instruments, which makes them immediately valuable to medical practice. Our methods were previously tested at 24 MHz in a study involving muscle ischemia and highly-vascular melanoma lesions in mice. This study translates the methods to lower frequencies (5-10 MHz) for human trials. We developed a dialysis-cartridge device using TM-blood to quantifying very slow flow in the presence of tissue-like clutter and noise. We calibrated our technique to quantify the sensitivity, dynamic range, and spatial resolution of PD imaging. Results indicate an ability to separate flow speeds as low as 0.3 mm/s from clutter signals.

9:45

4aBA5. Multi-resolution rank analysis for non-contrast ultrasound imaging of slow flow. Mahdi Bayat (Biomedical and Physiol., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu), Azra Alizad (Radiology, Mayo Clinic, Rochester, MN), and Mostafa Fatemi (Biomedical and Physiol., Mayo Clinic, Rochester, MN)

While conventional filtering techniques can effectively exclude tissue-related Doppler shifts from those resulting from blood motions in large vessels (flow rates ~ cm/s), significant spectral overlap can hinder proper clutter filtering in imaging slow flows (flow rates ~ mm/s or tenths of mm/s). Recently, several groups have proposed clutter filtering via processing of echo ensembles in the rank space instead of frequency domain. Though significant Doppler content can result from fast moving tissues, global coherence of the induced variations enables a low rank representation. In this talk, we first provide a brief overview of the low rank clutter removal methods. We then present a new framework for the analysis of Doppler ensembles using multi-linear decomposition of a tensor comprising Doppler ensembles at different rates. The model results in a high dimensional tensor from which, after proper rank selection, images of flow at different speeds can be extracted along different fibers. Using simulation models, we demonstrate the superior performance of the proposed method over singular value clutter filtering in imaging small vessels embedded in moving tissue undergoing both translational and shearing motions. We also present *in vivo* examples where the proposed method provides significant blood to clutter+noise ratio enhancement compared to singular value clutter filtering.

10:10–10:25 Break

Contributed Papers

10:25

4aBA6. Phantom study of flow speed measurement using ultrafast ultrasound plane wave imaging. Boran Zhou and Xiaoming Zhang (Radiology, Mayo Clinic, 321 3rd Ave SW, Rochester, MN 55902, Zhou.Boran@mayo.edu)

Color flow imaging is an ultrasound imaging technique widely used to image the blood movement through arteries and veins. Quantification of velocity is urgently demanding. The objective of this study is to use ultrafast plane wave imaging to measure the flow in speed *in vitro*. For the flow phantom, the diagonal vessel (2-mm inner diameter at 40° angle) was studied with injection of blood mimic fluid via syringe pump at various volume rates (15, 20, 25, and 30 ml/min). Verasonics Vantage system (Verasonics Inc., Kirkland, WA, USA) and a linear array transducer L11-4v (Verasonics Inc., Kirkland, WA, USA) were used for ultrasound acquisition, a 8-angle ($-7°$ to $7°$) plane wave compounding with effective pulse-repetition frequency (PRF) of 1000 Hz was used (central frequency = 6.42 MHz, number of transmit cycles = 4). A total of 600 ensembles were acquired. The frame rate was 1000/s.[XZ1] [BZ2] The ultrasound in-phase quadrature data size was $195 \times 156 \times 600$ (axial x lateral x temporal) with data type of complex double. Singular value decomposition (SVD)-based ultrasound blood flow clutter filters was used for ultrafast plane wave imaging.[XZ3] [BZ4] Randomized SVD was utilized to filter noise and tissue motion. Power Doppler image was generated. The color flow images were generated by using the 2-D autocorrelation method. Results showed that color flow measurement can quantitatively evaluate the flow speed in the tube of the flow phantom.

10:40

4aBA7. Quantitative morphological analysis of microvasculature in thyroid nodules using non-contrast high-resolution Doppler imaging. Siavash Ghavami (Radiology, Mayo Clinic, College of Medicine, 200 First St. SW, Rochester, MN 55905, roudsari.seyed@mayo.edu), Mahdi Bayat, Adriana Gregory, Bae Hyeong Kim, Kumar Viksit (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), Jeremy Webb (Radiology, Mayo Clinic, College of Medicine, Rochester, MN), Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), and Azra Alizad (Radiology, Mayo Clinic, College of Medicine, Rochester, MN)

We hypothesize that pattern of microvasculature in the thyroid nodules can be a biomarker of thyroid cancer. In this study, morphological features of microvasculature networks are assessed as potential biomarkers for differentiation of benign, suspicious and malignant thyroid nodules. This prospective study included $n=81$ patients, with a total of 61 benign, 12 suspicious and 8 malignant nodules verfied by pathology and cytology findings. Thyroid nodules were first identified using conventional B-mode ultrasound. A sequence of raw ultrasound data were acquired from each nodule site in two perpendicular planes at ~600 frames/sec. Power Doppler image of the microvasculature was formed using singular value decomposition clutter removal filtering. Gross images were further processed using a combination of morphology and Hessian-based vessel enhancement filtering. Background-free images of the microvasculature networks were segmented (vessels segment length $>385\mu$m and diameter $>90\mu$m) for morphological analysis. Using a Bayesian framework, measures of vessels density,

4a THU. AM

diameter and tortuosity were tested for significant differences in the three groups. Among all measures, the probability of vessel diameter $>600\mu$m in benign cases was significantly higher than that in malignant cases with the p-value <0.01 using Wilcoxon-Rank-Sum test, suggesting reliable cancer detection performance of the proposed method. [Acknowledgment: NIH Grant EB17213.]

10:55

4aBA8. Compound speckle model demonstrates two-phase wash-in of contrast-enhanced ultrasound cine loops. Matthew R. Lowerison (Medical Biophys., Univ. of Western ON, 95 Salem Pl., London, ON N6K 1T8, Canada, mloweri@uwo.ca), Ann F. Chambers (Oncology, Univ. of Western ON, London, ON, Canada), Hon S. Leong (Urology, Mayo Clinic, Rochester, MN), Nicholas E. Power (Surgery, Univ. of Western ON, London, ON, Canada), and James C. Lacefield (Robarts Res. Inst., London, ON, Canada)

During contrast-enhanced ultrasound (CEUS) imaging of tumor microvascular perfusion, nearby arteriole enhancement can be a dominant feature of the wash-in kinetics that obscures the perfusion characteristics of the capillary bed. This presentation demonstrates that statistical wash-in curves generated using our compound CEUS speckle model[1] exhibit two distinct phases corresponding to "fast-flow" and "slow-flow" enhancement that can be detected by fitting a linear combination of two monoexponential functions with different time constants. CEUS cine loops were acquired from a patient-derived xenograft model of renal cell carcinoma, where fresh tumor fragments were engrafted into the chicken embryo chorioallantoic membrane. Subharmonic CEUS images were acquired at 18 MHz using a destruction-replenishment protocol. Enhancement of manually segmented tumor cross-sections was analyzed offline in MATLAB. The CEUS cine loops from this xenograft model frequently exhibited in-plane arteriole enhancement. The two-phase fit discriminated a slow-flow component in 26.6% of time-intensity wash-in curves and 62.9% of the statistical wash-in curves, leading to a decrease in estimated tumor blood volume by 22.1% and 24.7%, respectively, compared to a simple monoexponential fit. These results suggest that conventional CEUS processing may frequently overestimate capillary blood volume. [1]M.R. Lowerison *et al.*, Med. Phys. **44,** 99-111 (2017).

11:10–11:55 Panel Discussion

THURSDAY MORNING, 10 MAY 2018 GREENWAY D, 10:10 A.M. TO 12:00 NOON

Session 4aEA

Engineering Acoustics, Signal Processing in Acoustics, and Underwater Acoustics: Advanced Transduction Technologies for Sonar Applications

Dehua Huang, Chair
NAVSEANPT, Howell St., Newport, RI 02841

Invited Papers

10:10

4aEA1. Transduction technologies for structural acoustics sonars. James Tressler (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, james.tressler@nrl.navy.mil)

This presentation will describe underwater acoustic transduction technologies that the Naval Research Laboratory (NRL) has developed for structural acoustics sonars. The NRL "Skyfish" downward-looking sonar system is deployed on a 21-inch diameter autonomous underwater vehicle (AUV) that features a pair of extendable/retractable wings, each of which houses a thirty-two element cylindrical hydrophone array. The well-behaved acoustic source consists of a pair of transducers, each covering a separate frequency band. The source is designed to provide a fairly wide acoustic aperture for looking both fore and aft as well as in the port and starboard directions. Architectural details of the source and receiver, materials considerations, amplifier design, and experimental data will be presented. [This work supported by the Office of Naval Research & ESTCP.]

10:30

4aEA2. Design issues for eigenbeamforming spherical microphone arrays. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoust., Fairfax, VT)

Approximating an analytic function by orthonormal bases functions has its origins in the theory of least squares developed by Gauss and Legendre. The use of orthonormal bases functions has been a standard tool in soundfield analysis since Rayleigh. One common set of orthonormal bases functions that naturally fit the solution of the spherical wave equation are the spherical harmonics. In this talk we describe the concept of building a microphone array on a rigid spherical baffle. We process the output signals from the spherical array such that we spatially decompose the acoustic soundfield into spherical harmonic Eigenbeam signals. Computationally efficient Eigenbeamforming, or modal beamforming, with these processed Eigenbeam signals will be discussed. Two main design issues are the growth in sensitivity to self-noise for higher-order spherical modes at lower values of k*r where, k is the wavenumber and r is the radius, and spatial aliasing due to the finite number of microphones covering the spherical surface at higher values of kr. We will discuss how to practically handle these design issues and how these design solutions impact the overall performance of the Eigenbeams and the use of these components in the design of modal beamformers.

10:50

4aEA3. The design of broadband constant beamwidth and constant beam pattern transducers. Dehua Huang (NAVSEANPT, Howell St., Newport, RI 02841, DHHuang@cox.net)

The broadband constant beamwidth (CBT) and the constant beam pattern (CBP) transducers attracted much attentions, because of their main lobes or even the whole beam patterns (CBP case) are independent of frequencies. For a spherical (or hemispherical) design, the far field angular beam patterns are the same as the normal directional radial particle velocity, or shading, distribution on the surface of the spherical transducer, if the *ka* product for the wave number and the sphere radius is greater than one, per spherical Hankel function asymptotic approximation to the solution of the Helmholtz wave equation. Any arbitrary velocity shading functions can be expanded by series of Legendre polynomials for optimization beamwidth and sidelobe levels. This paper introduced a new transducer or array design, where the methodology by CBP is employed, while the geometry of CBT spherical cap is applied, such that the shading functions are no longer confined by only one Legendre polynomial $P(\cos(\theta))$ of the CBT case. Here, the shading examples by various Legendre polynomials of $P(z_0*\cos(\theta))$ and the classic Dolph-Chebyshev $T(z_1*\cos(\theta))$ of different orders have been successfully simulated, where the z_0 and the z_1 are the design control parameters of the beams.

11:10

4aEA4. A review of single crystal underwater transducers. Harold Robinson (NUWC Div. Newport, 1176 Howell St., Newport, RI 02841, harold.c.robinson@navy.mil)

The unique material properties of lead magnesium niobate-lead titanate (PMN-PT) single crystals enable compact, broadband, high power sound projectors that cannot be realized using conventional ceramics. This paper shall highlight single crystal transducer prototypes developed at the Naval Undersea Warfare Center, Division Newport, and elsewhere demonstrating that the broadband, high power performance promised by single crystal's material properties can be realized in practical underwater transducers. Various single crystal transducer designs for underwater applications, including longitudinal vibrators, fully active and active-passive air-backed cylinder transducers, free-flooded ring transducers and bender bar transducers, will be shown. These transducer designs cover a wide span of frequencies, and utilize both 33- and 32- operating modes. Tuned bandwidths of up to 2.5 octaves of frequency, as well as source level increases of up to 15 dB at the band edges, will be demonstrated. t will also be shown that these technologies provide stable performance under high drive and duty cycle conditions. Finally, the impact of using these single crystal sources on the drive electronics and power consumption shall also be discussed.

Contributed Papers

11:30

4aEA5. Optical breakdown as a broadband underwater acoustic source. Athanasios G. Athanassiadis (Dept. of Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Bldg. 3-257c, Cambridge, MA 02139, thanasi@mit.edu)

Current underwater transducers provide little support for remote measurement of material properties. Such measurements are often limited by high impedance contrast, and target inhomogeneities that arise from biofouling and corrosion. In the presence of these limitations, a sensing system would benefit from broadband, spatially-localized target excitation. The high bandwidth would allow a return signal to encode broader information content about the target, while spatial localization would provide stronger target excitation and higher spatial resolution for measurements. Although these source properties are not achievable with traditional transducer designs, they can be produced through laser-induced optical breakdown in water. During optical breakdown, focused laser light excites a localized plasma, which expands and produces a mechanical shock wave. The shock properties are set by both the medium hydrodynamics and the laser parameters, allowing the acoustic source to be optically tuned. In laboratory experiments, breakdown from a 4 mJ/ 4ns laser pulse emits an exponential pressure wave with a decay constant under 250 ns and peak pressure near 600 kPa at 2 cm range. Here, I discuss the physics and fundamental limitations of acoustic transduction via underwater optical breakdown. Experimental results illustrate how the acoustic source properties can be tuned through optical control.

11:45

4aEA6. High power evaluation of textured piezoelectric ceramics for SONAR projectors. Arjun Shankar (Penn State Univ. / McKay Conant Hoover, Inc., 242 S Beck Ave, Tempe, AZ 85281, arjunkshankar@gmail.com) and Richard Meyer (Penn State Univ., State College, PA)

Textured Ceramics of the relaxor ferroelectric PMNT have shown exceptional piezoelectric properties suitable for high power SONAR projectors. The piezoelectric coefficient (d33>800pm/V), the electromechanical coupling coefficient (k33>0.80), and the mechanical quality factor (Qm>90) in various forms of PMNT showcase the potential for these textured ceramics to be implemented in a high power, broad bandwidth acoustic projector. This work attempted to devise a set of experiments to quickly characterize the electromechanical performance of these materials in relevant devices in order to provide feedback to both the material developers and the user community. In this effort, textured PMNT and Manganese doped PMNT (Mn:PMNT) were compared to the performance of conventional lead zirconate titanate (PZT) ceramics. Linearity and harmonic distortion measurements were made in a high pressure test facility to measure source levels of these SONAR projectors. In addition, isothermal measurements were made at 50 degrees Celsius to monitor source levels as a function of duty cycle. The results from these measurements were then compared to modeled results and appropriate recommendations on material composition improvements were made.

Session 4aID

Interdisciplinary, ASA Committee on Standards, Noise, Structural Acoustics and Vibration, and Architectural Acoustics: Acoustical Standards in Action: Realization, Applications, and Evolution

Christopher J. Struck, Chair
CJS Labs, 57 States Street, San Francisco, CA 94114

Invited Paper

8:00

4aID1. An overview of the Standards Program of the Acoustical Society of America. Christopher J. Struck (CJS Labs, 57 States St., San Francisco, CA 94114, cjs@cjs-labs.com)

An overview of the standards program of the Acoustical Society of America is presented. The American National Standards Institute accreditation and the U.S. voluntary consensus process are described. The structure of the ASA standards organization is explained and the roles of each standards committee are outlined. The standards development process from new work item proposal through final approval is detailed. The relationship between national and international standards bodies is also discussed. Examples of the practical benefits of acoustical standards for both participating organizations and end users are given. Information about how to participate in the standards process is also provided.

Contributed Paper

8:20

4aID2. Laboratory accreditation—Building confidence on acoustical testing- and calibration-lab quality. Richard J. Peppin (Engineers for Change, Inc., 5012 Macon Rd., Rockville, MD 20852, PeppinR@asme.org), George Anastasopoulos, and Prasanth S. Ramakrishnan (Int. Accreditation Service, Inc., Brea, CA)

Testing (and calibration) laboratories, to be good, must provide consistent accurate and precise measurements of the test objects based on their well-defined procedures. Lab accreditation is one way to assure the engineers, scientists, and technicians in the lab AND the lab customers and the public have confidence in the skills, facilities, and results of the lab's work. There is an international standard, ISO/IEC 17025-2005 "General requirements for the competence of testing and calibration laboratories." This standard is for use by laboratories in developing their management system, for quality, administrative, and technical operations Following this standard will help labs assure quality in their work. To assure the public (and the lab's customers) that the lab does follow this, an objective agency evaluates the lab's quality program. This process is called "accreditation." Accreditation is done under strict quality control by accrediting agencies, of which there are several available for worldwide labs. Most of the presentation will emphasize what is required and involved in the accreditation process This presentation will also discuss (1) why accreditation is useful, (2) who does the accreditation, (3) how are they qualified to do accreditation, (4) how costs are determined, and (5) how fast can it be accomplished.

Invited Paper

8:35

4aID3. Acoustical standards in the national parks. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CA 80525, kurt_fristrup@nps.gov)

U. S. National Parks began actively conserving park acoustic resources decades after U. S. community noise management practices were established. This presented opportunities to benefit from the knowledge acquired in community noise settings and challenges to adapt this knowledge and devise new methods as appropriate for national park settings. Participation in ANSI/ASA accredited standards development processes provided numerous benefits to the National Park Service (NPS): Access to a substantial fraction of the world's experts in acoustics, noise measurement, and noise reduction, many of whom were deeply involved in the development of community noise standards, Participation in an open, formal, coordinated, consensus-based process to develop voluntary standards, Engagement of a diverse range of stakeholders to develop standards through processes of cooperation and compromise, Conformance with the Office of Management and Budget's requirement to use voluntary consensus standards in lieu of government-unique standards (Circular A-119). NPS is an institutional member of two accredited standards committees and participates in the leadership of one committee and one working group. One standard has been published from this working group (ANSI S3-SC1.100 and ANSI S12.100: Methods to Define and Measure the Residual Sound in Protected Natural and Quiet Residential Areas). A second standard is in development.

Contributed Papers

8:55

4aID4. Comparing the similarity of sharpness predictions based on specific loudness from ANSI S3.4-2007 with those according to DIN 45692:2009 for several real-world sound classes. S. Hales Swift and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., N221 ESC, Provo, UT 84602, hales.swift@gmail.com)

A previous paper [Swift and Gee, Proc. Mtgs. Acoust., p. 030001, 2017] showed methods for transforming the standardized form of sharpness to accept inputs from the ANSI S3.4-2007 loudness standard. A further recent publication [Swift and Gee, JASA-EL, 2017] showed additional validation results comparing the results of the simplest version of the transformed metric for a set of 40 real-world sounds. In this paper, three versions of the ANSI-based sharpness metric are analyzed, in order to determine which of the three implementations gives the most similar results to the original DIN 45692:2009 standard. Finally, classes of sounds are identified for which predictions between the three ANSI-based sharpness metrics and DIN 45692:2009 may differ to a greater or lesser extent.

9:10

4aID5. Improving practices through analysis of long-term microphone calibration records. Gordon M. Ochi, Matthew G. Blevins, Gregory W. Lyons, and Edward T. Nykaza (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil)

A microphone's history of calibration is a useful source of information, as an independent analysis of the data can reveal more than simply the metrological traceability. The U.S. Army Engineer Research and Development Center maintains over 13 years of calibration records from an inventory of over 280 Class 1 microphones, which were examined for historical validity and consistency. Analyzing trends in the laboratory reported sensitivities showed that 12.9% of active microphones exhibited four possible trends that could indicate potential microphone failure: sensitivity drift, sensitivity drift followed by instability, jump in sensitivity, and general instability. Many of the microphones that displayed these trends had uncertainties greater than a ± 1.0 dB tolerance inferred from ANSI S1.14 Class 1 microphone specifications, and an additional 10.7% of microphones showed errors in the calibration laboratory's transcription process or testing procedures. Recommendations are put forth for best practices relative to laboratory independent microphone calibration record documentation and analysis, and a quantitative procedure for flagging microphones was developed. Specifically, it is recommended that microphones with a component type 'A' uncertainty due to the history of calibrations that exceeds ± 1.0 dB be removed for further inspection, in reference to the Class 1 microphone specifications in ANSI S1.14.

Invited Papers

9:25

4aID6. Pressure reciprocity calibrations of laboratory standard microphones at the National Institute of Standards and Technology. Randall Wagner (National Inst. of Standards and Technol., 100 Bureau Dr, MS 6833, Gaithersburg, MD 20899, randall.wagner@nist.gov)

The reciprocity technique has long served as a method for pressure calibration of microphones. It is a primary method, which determines microphone sensitivities from first principles and does not require a previously calibrated acoustic transfer standard. For calibrations of laboratory standard microphones, this method is standardized and utilized at national measurement institutes worldwide. Standard microphones calibrated by reciprocity are in turn used to calibrate additional microphones and sound calibrators, which apply known sound pressures to calibrate acoustical measuring devices and systems. Reciprocity calibrations done at the National Institute of Standards and Technology (NIST), which is the national measurement institute for the U.S., provide its customers with accurate results traceable to the International System of Units (SI). These customers and organizations that utilize their services perform large numbers of secondary and further calibrations and measurements concerned with hearing conservation and testing, aircraft noise, noise regulation enforcement, acoustical test and measurement equipment, and auditory research. An overview of the reciprocity technique is presented along with a few examples of how customers of the NIST acoustical calibration services make use of their calibrated devices and calibration data.

9:45

4aID7. Types of regulatory tools and their advantages, limitations, and disadvantages in a Model Noise Ordinance Standard. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

The American National Standards Institute is developing a model community noise ordinance. There are seven types of regulatory tools available to communities to regulate noise that can be used in a model noise ordinance. These include (1) nuisance based regulations, (2) disturbing the peace regulations, (3) decibel based regulations, (4) plainly audible regulations, (5) operational restrictions and prohibitions, (6) equipment requirements, and (7) quiet zones. In this paper, each is described, as well as the advantages, limitations, and disadvantages of each tool. A *Matrix of Advantages of Regulatory Tools Used in Community Noise Regulation* is presented and provides easy comparison between the regulatory tools. This matrix can provide the basis for a proposed Model Noise Ordinance standard.

10:05–10:20 Break

10:20

4aID8. Testing the limits of reverberation room qualification standard AHRI 220. Derrick P. Knight (Ingersoll Rand - Trane, 824 Olympic Dr., Onalaska, WI 54650, derrick.knight@irco.com)

Trane's acoustic lab recently upgraded measurement equipment which necessitated re-qualification of two reverberation chambers according to AHRI 220. We took this opportunity to test restrictions listed in the standard in relation to source-to-wall spacing and minimum number of fixed microphone positions. Measurements were simultaneously taken using rotating and fixed microphones. It will be shown that the wall proximity limit is overly restrictive, while the minimum fixed microphone requirement is appropriate.

10:40

4aID9. SII—Speech intelligibility index standard: ANSI S3.5 1997. Caslav Pavlovic (BatAndCat Corp., 602 Hawthorne Ave., Palo Alto, CA 94301, chas@batandcat.com)

The Speech Intelligibility Index Standard (SII) defines a method for computing a physical measure that is highly correlated with the intelligibility of speech. The SII is calculated from acoustical measurements of speech and noise. In this talk the activities of the Working Group for the Speech Intelligibility Standard will be reviewed including the issues to be resolved, the changes, and the additions under consideration. The general framework of the current standard (ANSI S3.5-1997) is such that new methodologies for specifying the effective ("equivalent") speech and noise can be easily incorporated as long as they have been tested and found valid for a well specified set of circumstances. The history of the Speech Intelligibility Index (SII) Standard and its relationship to the Articulation Index (AI) and the Speech Transmission Index(STI) will also reviewed. The general philosophy of Speech Intelligibility Index will also be discussed including the definition and specification of its fundamental variables: dynamic speech and noise spectra, information transmission by different bands, and their biologic/evolutionary compatibility with both the threshold of hearing, and the neural tuning.

11:00

4aID10. History of the sound level meter in standards. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

In 1932, the Acoustical Society of America started work on the first American Standards Association standard for sound level meters. The purpose was to create a set of standards for sound level meters "such that, if a given noise of a general character is measured with any meter designed in accordance with the standards, the result will be substantially the same as that which would be obtained with any other similarly designed meter." The standard that resulted, Z 24.3 1936 was the first of many that trace the development of the sound level meter over the past 86 years. This paper follows the history and development of the sound level meter in the original American Standards Association and subsequent American National Standards Institute standards: Z 24.3 1944, S1.4 1961, S1.4 1971, 1.4 1983, and S1.4 2014. During that time the sound level meter has shrunk from a more than 100 pound, vacuum tube machine to a modern the modern, lightweight meter. The standard, however, has grown from a mere 11 pages in 1936 to 50 pages in 2014.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics

Taffeta M. Elliott, Chair
Psychology Dept., New Mexico Institute of Mining and Technology, 801 Leroy Place, Socorro, NM 87801

Contributed Papers

8:30

4aMU1. Art in Acoustics. Peter Francis C. Gossweiler (Instituto de Artes, UFRGS, Trav. Nsa. Sra. de Lourdes, 230 apt. 503 D, Porto Alegre, Rio Grande do Sul 91920-040, Brazil, petergossweiler@gmail.com)

Since the first time the caveman shouted out loud to the heart of a cave, to listen for a kind of magical relationship, a huge spectrum of possibilities has grown to deal with this phenomenon of nature. In fact, the eco or even the reverberation has been fascinating the human being till nowadays, as we can see on noise control studies, acoustics studies, sound studies, etc. But we propose another approach, maybe not so straight related to science (as a demand centered on results) but with an undeniable interest with its capacities of creation through the study of the acoustics (with a demand focused on the process). From artistic works of John Cage, Alvin Lucier, Nicolas Collins, Charlemagne Palestine, Llorenç Barber, and Lukas Kühne, we engage an interdiciplinary relation between art and acoustic to clarify this type of magic in our life. A historical, theorical, and critical analysis is applied in this task.

8:45

4aMU2. Examining the effect of emerging audio formats and technology on acoustical space in film composition and mixing. Conor P. Cook (Design & Eng., Acoust. Surfaces, 123 Columbia Court N., Chaska, MN 55318, ccook@acousticalsurfaces.com)

Because film audio must balance the three basic soundtrack elements of dialogue, music, and effects, acoustical masking and professional disagreements during the post-production mixing process affect the importance given to the film score. Dialogue is customarily and logically given priority, causing film composers and music mixers to work around the acoustical space taken up by speech, typically through panning, equalization, avoidance of speech frequencies, and "density" in orchestration. Improvements in audio systems and formats, such as surround sound and now Dolby Atmos, are affording composers and mixers more acoustical space to avoid masking effects without a loss of dramatic support for the film, including the introduction of height cues and greater surround panning control, as well as sound objects, which control the localization of discrete sounds in 3D space. The new technologies also present acoustical design challenges as they interact in surprising ways with specific "acoustic signatures" of widely-varying media presentation spaces. Acoustical environment examples from earlier and recent film mixes as well as critical literature are given. Composers and mixers are encouraged to explore the range of expressive acoustical options made available by emerging technologies and the more widely-compatible mixing spaces designed for them.

9:00

4aMU3. Laryngeal height, modal registers, and modes of phonation contribute jointly to vocal timbre. Taffeta M. Elliott (Psych. Dept., New Mexico Inst. of Mining and Technol., 801 Leroy Pl., Socorro, NM 87801, taffeta.elliott@nmt.edu) and Lisa Popeil (Voiceworks, Sherman Oaks, CA)

Timbre encompasses every distinguishing quality of a sound other than its pitch, duration, loudness, and sound location. Musical sound makers exploit contrasts in timbre. The voice organ in particular is capable of an impressive array of vibrational patterns with acoustic consequences perceived as timbral changes. As a first step towards determining the dimensionality of perceptual timbre in a singing human voice, we assessed the relation between, on the one hand, production parameters, executed by a well-trained vocalist capable of the articulation control necessary for demonstrations, and on the other hand, the complex spectral and temporal modulations in the acoustic signal of the resulting 50 timbral variants. Production parameters on a high front vowel included 5 laryngeal heights (from the lowest possible to a mostly elevated position), 4 modes of phonation (clean, blowy, breathy, and pressed), and several changes to resonance (nasal, bright, and muted) as well as vocal register differences comparing vocal fry and modal registers. Raised and lowered larynx positions alter formants in a regular manner that overlaps with phonation patterns and other articulatory alterations of resonance.

9:15

4aMU4. Articulatory correlates of the acoustic transition during the second passaggio of sopranos. Richard C. Lissemore, Kevin Roon (Speech-Language-Hearing Sci., Graduate Center/The City Univ. of New York, 499 Fort Washington Ave., Apt. #4F, New York, NY 10033, rlissemore@gradcenter.cuny.edu), Christine H. Shadle (Haskins Labs., New Haven, CT), and D. H. Whalen (Speech-Language-Hearing Sci., Graduate Center/The City Univ. of New York, New Haven, Connecticut)

The present experiment examines articulatory correlates associated with the presence or absence of a change in A_1-A_2, the amplitude difference between the first two harmonics (f_o and $2f_o$),in the second passaggio transition (D5♮ to F5♮, or 587 to 698 Hz) of the soprano voice. An earlier investigation of this area in a classical, techniqued soprano singing on a low vowel revealed a change from negative to positive values in A_1-A_2 at a pivot point between E5♭ and E5♮. But when the same singer produced sounds without technique, the A_1-A_2 values remained negative. The present articulatory investigation uses ultrasound and optical tracking for analysis of head-corrected tongue contours, jaw opening, lip aperture, and lip protrusion that accompany the presence or absence of the acoustic change in this same singer. Preliminary results replicate the relationship between the acoustic change from negative to positive values of A_1-A_2 and show a substantial change in tongue dorsum height. In contrast, singing that exhibits only negative values of A_1-A_2 shows no such articulatory changes. The same experimental protocol was then used with sopranos who can and cannot make the A_1-A_2 acoustic change. Acoustic and articulatory results will be reported.
[Funded by NIH grant DC-002717.]

4a THU. AM

9:30

4aMU5. Comparing clarinet grunt and squeak notes to transitions in coupled oscillators. Cherise Cantrell, Joshua Vawdrey (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84058, cantrell.cherise@gmail.com), Jeffrey O'Flynn (Music, Utah Valley Univ., Orem, UT), and Bonnie Andersen (Phys., Utah Valley Univ., Orem, UT)

Under certain circumstance, a clarinet can cause an undesired squeak or grunt note. A squeak note is the clarinet shifting to a higher register—a higher set of harmonics—while a grunt note is the clarinet shifting to a lower register. One might hypothesize that such changes occur naturally in the clarinet over time. Analysis of such transitions with waterfall plots reveals a similarity to transitions made with thermoacoustically-driven coupled oscillators. For coupled oscillators, transitions can occur when the resonance of one oscillator passes through that of the other. This suggests that the cause of the undesired transitions in the clarinet are the result of one resonance (possibly the reed, which is affected by the embouchure) passing through that of the resonance of the other (likely the bore) rather than the clarinet itself relaxing to a different register as a function of time. Notes played on an artificially-blown clarinet in the second register were shown to be stable over several minutes. This suggests that changes in the embouchure drive the clarinet to grunt or squeak.

9:45–10:00 Break

10:00

4aMU6. Computational analysis of the Bundengan, an endangered musical instrument from Indonesia. Indraswari Kusumaningtyas (Dept. of Mech. and Industrial Eng., Universitas Gadjah Mada, Jalan Grafika 2, Kampus UGM, Yogyakarta, DIY 55281, Indonesia, i.kusumaningtyas@ugm.ac.id) and Gea O. Parikesit (Dept. of Nuclear Eng. and Eng. Phys., Universitas Gadjah Mada, Yogyakarta, DIY, Indonesia)

The Bundengan, an endangered Indonesian musical instrument, was first developed and played by duck herders. The dome-shaped resonator, which also protects the herders from adverse weather, is made from a specially woven grid of bamboo splits, covered with layers of bamboo sheaths, held in place with sugar palm fibre straps. Inside the resonator are a set of long, thin bamboo plates and a set of strings equipped with small bamboo clips, which vibrate together with the strings. When playing, the Bundengan player uses one hand to pluck the clipped strings and the other to pluck the bamboo plates. The clipped strings generate metal-like sounds, while the bamboo plates generate drum-like sounds. Hence, the Bundengan can imitate the sound of gamelan, a traditional Indonesian instrumental ensemble. Bundengan players report difficulties to tune this instrument and to keep it in tune for a long time. We develop simulations to help Bundengan musicians better understand this instrument. The first simulation quantitatively predicts how the bamboo clips influence the frequencies of the clipped strings. The second one quantitatively predicts how the shape of the resonator affects the soundscape of the Bundengan. These simulations also allow for a more precise procedure to build the instrument.

10:15

4aMU7. Switch of synchronization states of aeroacoustical coupled organ pipes. Jost L. Fischer and Rolf Bader (Inst. of Systematic Musicology, Univ. Hamburg, Neue Rabenstr. 13, Hamburg, Hamburg D-20354, Germany, jost.leonhardt.fischer@uni-hamburg.de)

The separation distance of two equally tuned, aeroacoustically coupled organ pipes is varied stepwise in a synchronization experiment. At distances of multiples of about half the wavelength of the fundamentals, abrupt changes of SPL and fundamental frequency of the coupled and synchronized system can be observed. Frequency jumps also occur at higher harmonics. The nonlinear coupling function of the system implicitly takes into account the time delay of the system caused by the separation distance and therefore plays an important role for the observed phenomenon. The effect can be explained as a switch of synchronization states from in-phase to anti-phase and vice versa.

10:30

4aMU8. A comparison of fractional-sized to full-sized cellos. Thomas Blanford (Graduate Program in Acoust., The Penn State Univ., 1623 S. Ashwicken Ct, State College, PA 16801, teb217@psu.edu) and Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., State College, PA)

Fractional-sized cellos (3/4, 1/2, etc.) are designed for the same musical playing range as a full-sized cello (4/4) but with scaled proportions for players for whom a full sized cello is too large. To compensate for the shorter string length of the smaller instruments, the strings are adjusted in order to obtain the correct tuning. The cello body vibration, which is strongly coupled to the internal air cavity, would not be expected to scale in the same manner as the strings. This causes the bridge impedance seen by the strings on the fractional-sized cellos to differ from the bridge impedance seen by the strings on a full-sized cello. In this talk, the physical dimensions of a $^1/_2$ and $^3/_4$ cello are compared with a full cello. Drive point measurements are also compared to illustrate how the strings couple differently with body of each size cello. The fractional-sized cellos are found to exhibit a slightly different sound due to the bridge impedance mismatch.

10:45

4aMU9. Correlation of string/body resonances on a cello. Samantha A. Young (Loyola Univ. Chicago, 1209 W Arthur Ave., Apt 209, Chicago, IL 60626, syoung11@luc.edu)

This project focuses on a 2001 Garavaglia full size acoustic cello, and investigates the resonance properties of a wooden body. Previous studies have estimated that the body resonance lies between the range of D_3-$F_3^{\#}$. The goals are to investigate the radiation patterns of the produced sound waves, to take high-speed video of a played string to physically observe the standing wave, and to simulate the body resonance of the cello. A mathematical model is formed to approach the situation where a musician plucks a string. Typical approaches assume the surface area of a finger to be a point source, but the surface area is actually parabolic, forming a bend in the string that is continuous to the following straight line. High speed video was taken to try out methods for tracking a standing wave at 1000 frames per second for a maximum number of data points per period. We will correlate the waveform fits from the high speed videos to the Fourier transformations from the acoustic data. These same methods will be applied in an anechoic chamber to test "ideal" conditions in order to isolate the body resonances. We will correlate the data from body resonances to our string data.

11:00

4aMU10. Impact of internal damping on forced vibrations of piano soundboards. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de), Niko Plath, and Florian Pfeifle (Inst. of Musicology, Univ. of Hamburg, Hamburg, Hamburg, Germany)

A Finite-Difference Model of a grand piano soundboard including internal damping is used to estimate the influence of the damping on the vibration of the soundboard. Theoretically, the increase of damping reduces eigenmodes and leads to forced oscillation patterns. Those patterns show considerable dependency of their vibrational shapes upon the driving point. In the extreme case of a heavy damping, the wave starting at the driving point does not reach the boundaries of the soundboard with much strength and is therefore not reflected. Then no eigenmodes exist anymore and the plate behaves like an infinite plate. In the other extreme of a soundboard not damped at all the eigenmodes of the soundboard clearly appear. The measured patterned of a soundboard are compared to the modeled ones and conclusions are derived in terms of the amount of damping in a real soundboard.

Session 4aNS

Noise, Psychological and Physiological Acoustics and ASA Committee on Standards: Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices, and Fit-Testing I

Cameron J. Fackler, Cochair
3M Personal Safety Division, 7911 Zionsville Road, Indianapolis, IN 46268

Elliott H. Berger, Cochair
Personal Safety Division, 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

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—8:00

Invited Papers

8:05

4aNS1. Total hearing health for preventing occupational hearing loss. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The National Institute for Occupational Safety and Health (NIOSH) developed the Total Worker Health program that encourages health and safety professionals to consider more than just a particular exposure. Workers can suffer hearing loss due to both workplace, but due to a wide range of activities outside of work. Training workers in the proper use and application hearing protection can extend the effectiveness of hearing loss prevention efforts and promote the concept of Total Hearing Health. Awareness of hazardous activities such as the use of power tools, firearms, and chemical exposures, can instill a safety culture beyond the workplace. Hearing protector fit testing gives workers an increased self-efficacy and assurance that they can safely enjoy noisy activities without the need to sacrifice their ability to hear. This presentation will describe the work that NIOSH has undertaken to promote hearing protector fit-testing for the workplace and describe ways in which the Centers for Disease Control and Prevention has promoted hearing health.

8:25

4aNS2. Fit-testing systems for hearing protectors: ANSI standards to the rescue. Cameron J. Fackler and Elliott H. Berger (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com)

To standardize the performance of fit-testing systems for hearing protection devices (HPDs), an ANSI/ASA standard has been in development since 2008. As of this writing, BSR/ASA S12.71 "Performance criteria for systems that estimate the attenuation of passive hearing protectors for individual users" has been submitted for public comment and ballot. This standard refers to fit-testing systems as field attenuation estimation systems (FAESs). In this talk, we introduce important FAES concepts, as handled in the draft S12.71. There are two types of FAESs: physical systems that take an objective measurement and psychophysical systems that rely on subjective responses from the user being tested. FAESs may produce a quantitative output as a personal attenuation rating (PAR) or simply indicate pass/fail for a given HPD fit. For all types of systems, S12.71 specifies methods to assess the quality of the attenuation estimates, by comparing the FAES results to a laboratory real-ear attenuation at threshold (REAT) measurement. Other important considerations covered in S12.71 include the maximum levels of ambient noise in which FAESs may be operated, the incorporation of uncertainties due to HPD fitting and spectral variability in users' noise exposures, calibration intervals, and information that must be provided to users.

8:45

4aNS3. A field microphone in real ear hearing protector attenuation measurement system. Kevin Michael (Michael & Assoc., Inc., 2766 W. College Ave., Ste. 1, State College, PA 16801, kevin@michaelassociates.com)

A Field-Microphone in Real Ear (F-MIRE) hearing protector Field Attenuation Estimation System (FAES) was developed for individual or group testing of any type or style of hearing protector device (HPD). Both occluded and unoccluded ear canal stimulus level measurements are made with a tiny Micro Electro-Mechanical System (MEMS) microphone mounted on a 2-mil thick flat flexible cable. The microphone is mounted inside of a conventional foam ear dam to protect the microphone and the ear canal and to facilitate insertion

into the ear canal. The presence of the cable in the ear canal was demonstrated to have a negligible effect on attenuation measurements for both muff and insert-type HPDs. The system measures HPD attenuation in 1/1 or 1/3 octave bands and it is capable of simultaneously measuring HPD attenuation on up to eight human test subjects. Stimuli can be presented via headphones or loudspeakers. Human subject testing using both the laboratory based ANSI S12.6-2016 method and the F-MIRE system was performed using muff- and insert- type HPDs. Regression equations were generated from these data. These equations are incorporated in the F-MIRE software to allow valid comparisons between F-MIRE data and labeled HPD attenuation data.

9:05

4aNS4. Fit testing of a hearing protection device with integrated in-ear noise exposure monitoring. Christopher J. Smalt, Shakti K. Davis (MIT Lincoln Lab., 244 Wood St, Lexington, MA 02420, Christopher.Smalt@ll.mit.edu), William J. Murphy, Chucri A. Kardous (Hearing Loss Prevention Team, CDC-NIOSH, Cincinnati, OH), Joe Lacirignola, and Paul Calamia (MIT Lincoln Lab., Lexington, MA)

In-the-ear hearing protectors are often used in high-noise environments, such as in military operations and during weapons training. The fit of such a hearing protection device in the ear canal can cause significant variability in the noise dose experienced, an important factor in characterizing the auditory health risk. Our approach for improved dose estimates under these conditions involves a portable noise recorder used to capture in-the-ear noise behind a hearing protector, and on-body noise, while assessing hearing protection fit throughout recording. In this presentation we describe evaluation of this system using ANSI S12.42 testing using a shock tube and an acoustic test fixture, to evaluate impulse peak reduction and for measurement validation. Also explored were angle dependent effects on the peak insertion loss and measurement accuracy of the on-body recorder. An exploratory study was conducted with a small sample of experimenters during a recent Navy-sponsored noise survey conducted at Marine Corps Base Quantico. Results will be presented from a shooter's ear and a nominal instructor's position as well as from bystanders observing at a firing range. Our results show good correspondence between behavioral fit-testing of insertion loss and estimated protection from the in-ear and on-body microphones. [Work supported by ONR.]

9:25

4aNS5. Acoustical corrections to be used for improved in-ear noise dosimetry measurements. Fabien Bonnet (École de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, fabien.bonnet@etsmtl.ca), Hugues Nélisse (Institut de recherche Robert-Sauvé en santé et en sécurité du travail, Montreal, QC, Canada), and Jeremie Voix (École de Technologie Supérieure, Montréal, QC, Canada)

Although much effort has been made to reduce sound at the source, hearing protection devices (HPDs) remain currently the most widely used solution against occupational noise exposure in an attempt to prevent noise-induced hearing loss. Such devices, though, often suffer from uncertainties when it comes to ensuring ideal protection for a given individual. These uncertainties stem mainly from three unknown variables: i) the actual attenuation provided by the HPD to the individual; ii) the ambient noise levels; iii) the wear time of the given HPD. In-ear dosimetry (IED) could overcome these issues, as it makes it possible to establish precise personal noise exposures through real-time in-ear noise monitoring. Furthermore, IEDs can be worn in the open, occluded (under earplugs), or cushioned (under earmuffs) ear. Nevertheless, current IEDs are weakened by two inaccuracies: the non-discrimination of self-induced noise, and the lack of acoustical corrections to convert the measured in-ear sound pressure levels to the eardrum and to the free-field. The latter issue is treated in this presentation, which describes methods to identify the appropriate acoustical corrections to be used for improved IED. Such methods aim at personal calibration procedures, so that the unique ear geometry of each individual is accounted for. Preliminary results will be presented as a comparison between theoretical and experimentally measured corrections on human test-subjects.

9:45–10:00 Break

10:00

4aNS6. Earplug comfort: From subjective assessment on the field to objective measurement and simulation using augmented artificial heads. Olivier Doutres (Mech. Eng., École de technologie supérieure, 1100 rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, olivier.doutres@etsmtl.ca), Franck C. Sgard (IRSST, Montreal, QC, Canada), Simon Benacchio (Mech. Eng., École de technologie supérieure, Montréal, QC, Canada), Jonathan Terroir (INRS, Vandoeuvre Les Nancy, France), Nellie Perrin, Nicolas Trompette (INRS, Vandoeuvre-lès-Nancy, France), Alessia Negrini (IRSST, Montreal, QC, Canada), Marc-André Gaudreau (Mech. Eng., Université du Québec à Trois-Rivières, Montréal, QC, Canada), Caroline Jolly (IRSST, Montreal, QC, Canada), Alain Berry, Philippe-Aubert Gauthier (Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC, Canada), Thomas Padois (Mech. Eng., École de technologie supérieure, Montréal, QC, Canada), and Chantal Gauvin (IRSST, Montreal, QC, Canada)

For several years, lack of comfort has been pointed out as a major reason of earplug poor efficiency as noise control solution. The great complexity of comfort makes it difficult to predict it in the earplugs design phase. It is rather considered in an empirical way by the manufacturers using trial-and-errors approaches based on subjective assessment over a panel of subjects. Furthermore, because comfort is not quantified, Occupational Health and Safety (OHS) practitioners cannot select earplugs ensuring wearer comfort. To address these issues, a major international research project funded by two OHS institutes (IRSST in Canada and INRS in France) and involving several Universities (in Canada and England) started in 2017. The main objectives are to: (1) improve the understanding of earplugs comfort as perceived by field workers with consideration of all comfort components, (2) develop laboratory tools (augmented experimental and virtual artificial heads) to measure physical design variables related to the auditory, physiological, and functional components of comfort and (3) design a battery of hybrid objective/subjective comfort indices to quantify / measure / predict the different components of comfort. The aim of this presentation will be to present the project and first results.

10:20

4aNS7. Application of a registration method on magnetic resonance images to evaluate the displacement field of a human subject ear canal due to various earplug insertions. Simon Benacchio, Olivier Doutres (École de technologie supérieure, 505 Boulevard de Maisonneuve O, Montréal, QC H3A 3C2, Canada, Simon.Benacchio@irsst.qc.ca), Arthur Varoquaux (Aix Marseille Université, CNRS, CRMBM/CEMEREM UMR 7339, Marseille, France), Eric Wagnac (École de technologie supérieure, Montréal, QC, Canada), Arnaud Le Troter, Virginie Callot (Aix Marseille Université, CNRS, CRMBM/CEMEREM UMR 7339, Marseille, France), and Franck C. Sgard (IRSST, Montreal, QC, Canada)

Earplugs are a usual way to protect workers subjected to noise exposure. However, the efficiency of these hearing protection devices is often affected by induced discomforts. A factor suspected to impact both acoustical and physiological comfort attributes of earplugs is the deformation they apply on the ear canal walls. As the geometry of both open and occluded ear canal is difficult to obtain, the ear canal deformation due to earplug insertion is not trivial to evaluate. Current medical imaging techniques and image post-processing methods are promising tools to investigate this deformation. In a previous study of the authors, an approach using registration methods on medical images had been proposed to estimate the ear canal displacement field induced by earplug insertion. This approach had been validated in the case of computed tomography scans of a human-like artificial ear occluded by a controlled-shape custom molded earplug. In the present study, this approach is used to evaluate the ear canal displacement of a human subject for various earplug insertions (foam, pre-molded and custom made) in the case of magnetic resonance images. The computed displacement field shows noteworthy differences between each earplug and gives information on how and where the occluded ear canal deforms.

10:40

4aNS8. Sound localization performance of sample hearing protectors using standardized measurement methods. Eric R. Thompson (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St, B441, Wright-Patterson AFB, OH 45433, eric.thompson.28@us.af.mil) and Richard L. McKinley (Oak Ridge Inst. for Sci. and Education, Wright-Patterson AFB, OH)

Sound localization is an important aspect in the performance of many tasks. These tasks frequently require the use of hearing protection to mitigate the potentially adverse effects of noise exposure. ANSI/ASA working group S3/WG94 has described four standard methods for measuring the effect of head-worn devices on sound localization. Data using three different hearing protectors and open ear and using three of the four methods described will be presented. The analysis techniques used for each of the three methods will be described in detail including the removal of front-back reversals and assessment of fine resolution location discrimination. The performance of the three hearing protectors and the open ear condition will be compared and contrasted. Data will be presented on proportion of localization discrimination and front/back reversals using Method 1 as well as localization error performance using Methods 2, and visual target acquisition and response time using Method 3. The performance will also be compared across the three methods. The results show that the measurement methods described are viable for the assessment of the effects of HPDs on performance associated with localization.

Contributed Paper

11:00

4aNS9. Standard methods to measure the effects of head-worn devices on sound localization. Richard L. McKinley (Oak Ridge Inst. for Sci. and Education, 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, rich3audio@aol.com) and Eric R. Thompson (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Four standard methods for measuring the effect of head-worn devices on sound localization in the horizontal plane have been developed by the working group S3/WG94. The objective was to establish methods that enable accurate, repeatable, and reliable measurement of sound localization performance for human listeners. The standard describes four measurement methods: (1) a low complexity, coarser method to estimate the proportion of location discrimination and front/back reversal errors using 8 loudspeakers; (2) a more complex, more robust method to measure localization error using 36 loudspeakers and a fine resolution response method; (3) a method to measure the functional impact of localization with degraded cues using 36 loudspeakers and an aurally guided visual search task; and (4) a method using 180 loudspeakers to precisely measure localization acuity. The methods will be described in detail.

Session 4aPA

Physical Acoustics, Noise, and ASA Committee on Standards: Sonic Boom I

Alexandra Loubeau, Cochair
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Chair's Introduction—8:40

Invited Papers

8:45

4aPA1. Atmospheric turbulence effects on sonic boom signatures. Christopher Hobbs and Kevin Bradley (KBRwyle, 8580 Cinder Bed Rd., Ste. 1700, Lorton, VA 22079, chris.hobbs@wyle.com)

Two measurement campaigns were conducted as part of a study to quantify the effect of atmospheric turbulence on sonic boom propagation. Supersonic overflights were conducted at the NASA Armstrong Flight Research Center and the Kennedy Space Center to provide different atmospheric conditions for study. Measurements were made of the strength of the turbulence along with the height of the atmospheric boundary layer. Multiple microphone arrays recorded the sonic booms from instrumented aircraft. This work will present statistics detailing the variation of measured booms as a function of the strength of the turbulence and the propagation distance through the atmospheric boundary layer. [Work supported by NASA.]

9:05

4aPA2. Validation of numerical predictions of sonic boom metric variability due to turbulence. Trevor A. Stout and Victor Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, stout.trevor.a@gmail.com)

Atmospheric turbulence scatters the sonic boom wavefront created by supersonic aircraft, causing random variations in the waveforms and spectra measured at the ground. To help assess the impact of turbulence, a time-domain algorithm based on the nonlinear KZK propagation equation has been developed which simulates turbulent fields via an approximate atmospheric theory. Two sets of recent supersonic overflight measurements at the NASA Neil A. Armstrong Flight Research Center and the Kennedy Space Center offer datasets which have been used to quantify the algorithm's utility to simultaneously model atmospheric turbulence, nonlinearity, and absorption. The field campaigns simultaneously recorded measurements of both sonic boom waveforms along linear microphone arrays and also turbulence quantities suitable for use in the algorithm. The algorithm's performance in predicting the measured variation of sonic boom metrics will be discussed. [Work supported by NASA via a subcontract from KBRwyle.]

9:25

4aPA3. Formulation of a Burgers' equation for shock waves in a turbulent atmosphere. Joel B. Lonzaga (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

The theoretical formulation of a Burgers' equation accounting for the attenuation and dispersion of shock waves due to scattering in a turbulent medium is presented. The physical effect treated here is in addition to the effects of geometric spreading, large-scale inhomogeneity, nonlinearity, viscous and thermal dissipation, and molecular vibrational relaxation. Although all Burgers' equations are inherently a one-way approximation, here the scattering effects are incorporated into the Burgers' equation through a correction to the small-amplitude sound speed. Such a correction is obtained from the full wave equation using a perturbative technique, and physically arises from multiple scattering of the signal due to small-scale inhomogeneities. The sound speed correction is complex, and is dependent on the frequency and the variance and length scale of the fluctuations. The real part of the correction gives rise to the dispersion of the signal while the imaginary part leads to signal attenuation. The latter is manifested by the increase in the rise time of the shock waves. The physical effect presented here is integrated into an existing sonic boom propagation code with minimal additional computational time. Numerical results will be compared with experimental data.

9:45

4aPA4. Experimentally derived turbulence filters applied to low amplitude sonic booms. Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., MS 463, Hampton, VA 23681, j.klos@nasa.gov)

Turbulence effects on low amplitude sonic booms are not currently well understood. In 2011, NASA funded an experiment to collect data using a large linear microphone array, consisting of 81 ground placed microphones spaced 125-feet apart, as a supersonic F-18B aircraft flew over the array. Most of the measurements were made during aircraft maneuvers designed to place a focus boom on the microphones. However, a few flights were straight and level passes of the aircraft over the array. Data from two of these level passes were used to estimate filter functions that describe the variation at each microphone relative to the mean response across the array. Two approaches were implemented, one is frequency domain and the second is time domain. The resulting filters can be applied to other non-turbulent boom waveforms to estimate the changes in either the spectrum or time domain signature due to the variations observed during these flights. Variation in loudness level both outdoors and indoors when these filters are applied to predicted signatures is presented. Both outdoors and indoors, the standard deviation of Perceived Level is only slightly smaller for low boom waveforms in comparison to N-waves.

10:05

4aPA5. Investigation of deturbing methods for sonic boom signatures. Janet L. Xu (Acoust., Penn State Univ., 1623 S. Ashwicken Ct., State College, PA 16801, jqx5082@psu.edu) and Victor Sparrow (Acoust., Penn State Univ., University Park, PA)

There is a desire to certify aircraft for supersonic flight using microphone measurements of the aircraft's sonic boom signature on the ground. If the signature propagates through the planetary boundary layer before it reaches the ground, the signature becomes distorted because of atmospheric turbulence, corrupting the measurement. Methods to remove the turbulence effect, or deturbing methods, currently exist but are restricted to typical N-wave shapes or knowledge of the turbulence structure beforehand. This work investigates methods of deturbing sonic boom signatures independent of the N-wave shape and not requiring prior knowledge of turbulence. Improvements are made to the averaging method and subtraction method of deturbing, and a new method is investigated which uses audio fingerprinting, an audio identification algorithm used by music identification apps such as Shazam. The effects of each deturbing method are described using a wide set of physical and subjective metrics. [Work supported by the FAA. The opinions, findings, conclusions, and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

10:25–10:40 Break

10:40

4aPA6. Sonic boom carpet widths for NASA's Low Boom Flight Demonstrator using realistic atmospheric profiles in the western hemisphere. William Doebler (The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, wfd5057@psu.edu), Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), and Victor Sparrow (The Penn State Univ., University Park, PA)

NASA is developing and building a Low Boom Flight Demonstrator (LBFD) within the next five years to study public reaction to low amplitude, shaped sonic booms. Understanding the potential land area ensonified by sonic booms is critical for the upcoming community tests, and is not well known outside of using the 1976 US Standard Atmosphere which assumes zero wind. Using atmospheric profiles from NOAA's Climate Forecast System Reanalysis (CFSR) for a span of many months, the sonic boom carpet widths and Stevens Mark VII Perceived Levels (PL) at the edge of the carpet were calculated using NASA's sBOOM augmented Burgers Equation propagation code. The LBFD's near field pressure cylinder calculated using NASA's LAVA CFD code was used as the input to the sBOOM code. Atmospheric profiles were sampled from several regions with varying climate in the western hemisphere. Preliminary results indicate that sonic boom carpets will be widest during winter months, and will have highest carpet edge PL values during summer months. [Work supported by the NASA Education AS&ASTAR Fellowship Program.]

11:00

4aPA7. An improved Mach cut-off Model based on a 3-D ray tracing method and realistic atmospheric data. Zhendong Huang and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, zfh5044@psu.edu)

This research aims to assess the possibilities for Mach cut-off flight over land. An improved Mach cut-off model has been developed using a 3-D ray tracing method. Based on the Integrated Global Radiosonde Archive (IGRA), a radiosonde dataset from the National Centers for Environmental Information (NCEI) consisting of radiosonde and pilot balloon observations, the influence of realistic atmospheric profiles and flight conditions on cut-off Mach numbers is examined. Examples are given for certain busy air routes in the United States. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

11:20

4aPA8. Mean flow atmospheric effects and their impact on sonic boom propagation. Sriram Rallabhandi (Aeronautics Systems Anal. Branch, NASA Langley, Rm. 190-25, Mailstop 442, NASA Langley Res. Ctr., Hampton, VA 23681, sriram.rallabhandi@nasa.gov)

One dimensional augmented Burgers equation has been generally used for nonlinear lossy sonic boom propagation. This equation models the effects of nonlinearities, loss mechanisms such as absorption and dispersion due to molecular relaxation and thermoviscous dissipation, and geometric spreading through ray tube areas including Blokhintzev scaling as well as atmospheric stratification.

However, in the presence of atmospheric winds, original terms in the augmented Burgers formulation do not account for the Doppler effects, where the observer is moving with the local flow rather than being fixed in space. Instead, wind is accounted for by updating the ray paths and effective speed of sound. Inclusion of mean flow wind effects in all terms of the augmented Burgers equation allows for an enhanced prediction capability that is closer to the underlying physics. This work will update sBOOM, an augmented Burgers' solver, to reflect the mean flow wind enhancements. The sonic boom ground signatures and other relevant data are compared against those obtained without using mean flow wind effects. The shock rise times, sonic boom duration and peak pressures are some variables that are expected to be different, resulting in differences in noise metrics. Such differences will be discussed and documented for cases which may include shaped low-boom as well as strong shock signatures.

11:40

4aPA9. Time-domain modeling of the wind effects on the nonlinearity and absorption of sonic booms. Joel B. Lonzaga (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

This paper discusses the derivation and solution, in the time domain, of a Burgers' equation for sonic boom propagation in a windy atmosphere. As the wind effects become important with increasing wind speed, the proper treatment of wind is necessary for sonic booms propagating through a jet stream as well as for those propagating to the stratosphere or thermosphere where stronger winds typically exist. The effects of wind are quantified in terms of the Doppler shift and of a parameter which measures the strength of the convection. The latter, referred to as the convective index, is a ratio of the local sound speed to the magnitude of the ray velocity and is unity in the absence of wind. Both Doppler shift and convective index are larger than unity if there is a component of the wind velocity opposite to the direction of the acoustic propagation. Consequently, the effective nonlinearity coefficient and effective absorption coefficients are larger for rays propagating opposite the wind direction than for rays propagating in the same direction. To demonstrate the significance of the derived Burgers' equation, the effect of a jet stream with a typical strength and profile on nominal sonic booms will be described.

THURSDAY MORNING, 10 MAY 2018 — NICOLLET D2, 8:00 A.M. TO 11:35 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Honoring Neal Viemeister's Contributions to Psychoacoustics

Andrew J. Oxenham, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

4aPP1. Temporal integration and multiple looks, revisited. Gregory H. Wakefield (Elec. Eng. and Comput. Sci. Dept., Univ. of Michigan, Ann Arbor, MI 48109, ghw@umich.edu)

Recent advances in Kullback-Leibler divergence and its generalizations have led to a renewed interest in the modeling of various detection and classification problems. Building from the concept of multiple looks, this paper considers adaptive detector and classifier structures that support stable, multi-stream perception. With respect to these structures, the phenomena of temporal integration and the distinctions between energetic and informational masking are discussed. [Work supported by ONR N00014-16-1-2560.]

8:25

4aPP2. Using psychoacoustics to examine a changing auditory system. Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, estrick@purdue.edu)

In the (long) time I was a student in Neal's lab, I learned a rigorous, programmatic approach to psychoacoustics that was grounded in signal detection theory. Much of the work at that time followed up on Neal's classic work in intensity discrimination and envelope processing, and how that might depend on processing within or across auditory channels. The research was tied to what was known

about physiological processing in the auditory system. Research from other laboratories at the time suggested that the cochlea might adjust in response to sound via efferent feedback, and a paper by Liberman used a physiological approach that could be easily translated to psychoacoustic experiments. This inspired a search for behavioral evidence of efferent feedback to the cochlea, and an array of experiments were performed looking for effects of contralateral sound on psychoacoustic measures. While those experiments did not provide compelling evidence of efferent feedback, they sparked an interest that has continued to be a focus of my research. This talk will cover the array of experiments done at that time, and consider the results in the context of more recent results from my laboratory and other laboratories. [Work supported by NIH(NIDCD)R01 DC008327.]

8:45

4aPP3. Effects of age and noise exposure on the representation of amplitude modulation. Christopher J. Plack (Dept. of Psych., Lancaster Univ., Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk) and Samuele Carcagno (Dept. of Psych., Lancaster Univ., Lancaster, United Kingdom)

Age and lifetime noise exposure were used as predictors for electrophysiological and psychophysical measures of amplitude modulation processing. Frequency-following responses (FFRs) were recorded from 61 listeners (20 young, 23 middle-aged, 18 elderly) to 0.6- and 2-kHz, 75-dB SPL, carrier tones amplitude modulated at 100 Hz. A pink noise highpass-filtered at 3 kHz was included to mask basal cochlear contributions to the FFR. Sinusoidal amplitude modulation detection (AMD) thresholds were measured for the same listeners for a 2-kHz carrier (40 and 80 dB SPL) and for three modulation frequencies (25, 50, and 100 Hz). The carrier was presented in notched pink noise to limit off-frequency listening. Lifetime noise exposure was estimated using a structured interview. A regression model was used to determine the independent contributions of age and noise exposure, while controlling for audiometric threshold. FFR amplitude for the 0.6 kHz carrier showed a marked age-related decline. Age was also associated with a significant increase in AMD thresholds. Lifetime noise exposure was not significantly associated with either FFR amplitude or AMD threshold. The results suggest that age is a stronger predictor of modulation processing deficits than lifetime noise exposure.

9:05

4aPP4. Psychoacoustic considerations in hearing aid design and fitting. Brent Edwards (National Acoust. Labs., Australian Hearing Hub, Level 4, 16 University Ave., Macquarie University, NSW 2109, Australia, brent.edwards@nal.gov.au)

Hearing aid technology has advanced tremendously over the past two decades, with multiple technologies introduced to improve the hearing of wearers with hearing loss. While hearing science has informed their development, the design of these technologies and their fitting have been approached primarily from a speech-centric perspective. This talk will reflect on hearing aid technology from the perspective of the type of fundamental psychoacoustic research conducted by Neal Viemeister throughout his career. Additional psychoacoustic research that is needed to inform future hearing aid designs will also be identified. A cookie will be bet that Neal is surprised by the relationship between psychoacoustics and hearing aid design.

9:25

4aPP5. Context effects in modulation masking. Magdalena Wojtczak (Psych., Univ. of Minnesota, 1237 Imperial Ln, New Brighton, MN 55112, wojtc001@umn.edu)

Neal Viemeister is especially known for his contributions to our understanding of perception of amplitude modulation. Modulation masking and signal-to-noise ratio in the modulation domain have been shown to be strong predictors of speech intelligibility in noise. This study uses 40-ms amplitude-modulated tones embedded in a simultaneous 100-ms noise masker to show that for a given type of simultaneous masker, modulation masking can vary significantly depending on the type of stimulus that precedes the masked target. Three types of precursor were used to investigate these context effects, a 2-octave noise centered on the signal frequency (the same as the masker), a 7-component inharmonic complex tone, and a pure tone with a frequency equal to that of the target. All precursors had a duration of 400 ms and were presented in close temporal proximity at levels equal to those of the noise masker. The noise precursor produced on average a 7-8 dB [20log(m)] release from modulation masking. In contrast no change in the amount of masking and a small release of masking were observed for the pure-tone and complex-tone precursors, respectively. Mechanisms that may underlie the observed context effects will be discussed. [Work supported by NIH grant R01 DC015462.]

9:45–10:00 Break

Contributed Papers

10:00

4aPP6. Neal Viemeister's contributions to understanding the dynamic range of the auditory system. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Most neurons in the auditory nerve show saturation of their firing rates in response to a tone with a moderate level presented at their characteristic frequency. However, humans can detect changes in intensity for levels up to 120 dB SPL. One explanation of this "dynamic range problem" is that intensity coding at high levels depends on the spread of cochlear excitation. To test this explanation, in 1974 Neal Viemeister measured the intensity discrimination of bursts of noise presented in a band-reject noise of fixed high intensity. He found that Weber's law held even at high levels. He concluded that spread of excitation is not necessary for the auditory system to maintain its large dynamic range. In 1983 Neal Viemeister showed that, over a wide range of intensities, subjects could detect small differences in the intensity of a high-frequency band of noise (6-14 kHz, a frequency range where phase locking is essentially absent) presented with a complementary band-reject noise, showing that phase locking is not essential for intensity discrimination at high levels. In 1988 Neal Viemeister showed theoretically that a localized rate-based intensity code using a small number of neurons can account for intensity discrimination over a wide range of levels.

10:15

4aPP7. Loudness of an auditory scene. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

When multiple sound sources, each of equal level, are added together to create an auditory scene, the overall level of the scene grows with the increased number of sources despite the fact that the level of the individual sound sources remains the same. This experiment addresses a simple question: how does the overall perceived loudness of an auditory scene change as the number of sound sources in the scene changes. Sixteen Consonant-Vowels (CVs), spoken by ten males and ten females, were the sound stimuli. The CVs could be concatenated to produce nonsense "words" of different lengths. Different number ("n") of concatenated CVs could be presented at one time from a single loudspeaker located on the azimuth plane or from different loudspeakers. Listeners determined if "n + delta n" sound sources were louder than "n" sound sources. The individual levels of the concatenated CVs was another independent variable. Preliminary data suggest that when "n" is small (<4), the level of the individual sounds determines loudness judgements. When n is larger ($n>4$) the overall level of the simultaneously presented sounds determines loudness judgements. A full study will be presented. (Work supported by grants from NIDCD and Oculus VR, LLC.)

10:30

4aPP8. The contribution of Neal Veimeister to the modulation theory of hearing. Christian Lorenzi (Institut d'Etude de la Cognition, Ecole normale superieure, 29 rue d'Ulm, Paris, Ile de France 75005, France, christian.lorenzi@ens.fr)

Neal Viemeister has dedicated his scientific career to one of the most influential research programs in hearing sciences. This program aims to apply the modulation theory to auditory perception. This theory, inspired by telecommunication sciences, relies on the following assumptions: (i) communication sounds including speech and animal vocalizations convey salient and slow temporal modulation cues; (ii) the auditory system of species using communication sounds has evolved to optimize its responses to these modulation cues; (iii) the transmission of information conveyed by communication signals is constrained by the 'modulation transfer function' (MTF) of the transmission path (a room, a hearing aid,...). We will show how the pioneering work of Neal Viemeister on the 'temporal MTF' and his rigorous investigation of modulation perception by human listeners contributed to an in-depth understanding of the demodulation processes implemented in the peripheral and central auditory system. We will show how his work yielded to models of modulation processing that account for a large range of listening situations including speech intelligibility and texture perception. Finally, we will show how his work contributed to a better understanding of the perceptual consequences of ageing and cochlear damage, and a better control of information transmission via rehabilitation systems.

10:45

4aPP9. Amplitude modulation detection and modulation masking in school-age children and adults. Emily Buss, John Grose (The Univ. of North Carolina at Chapel Hill, 170 Manning Dr., G190 Physicians, Chapel HIll, NC 27599, ebuss@med.unc.edu), and Lori Leibold (Boys Town National Res. Hospital, Omaha, NE)

Temporal resolution is often evaluated by measuring sensitivity to amplitude modulation (AM) as a function of modulation rate. Published data for AM detection on a noise carrier indicate that school-age children perform more poorly than adults, but thresholds increase with increasing AM rates above 50 Hz in a parallel fashion for both age groups. This result has been interpreted as reflecting adult-like temporal resolution in school-age children. However, this interpretation is complicated by the observation that inherent AM of the noise carrier can elevate AM detection thresholds in adults via modulation masking. Two studies with 5- to 11-year-olds and adults were carried out to better understand the development of AM detection with and without modulation masking. The first estimated thresholds for detecting 16-, 64- or 256-Hz sinusoidal AM on a 4.3-kHz carrier, a condition without modulation masking. Age effects were larger than previously observed with a noise carrier. The second study manipulated modulation masking by measuring detection of 16-Hz sinusoidal AM on a bandpass noise carrier (1.5—2.5 kHz), with and without additional masker AM at nominal rates of 6.3, 16, or 40.3 Hz. All listeners exhibited on-frequency modulation masking, with similar tuning to modulation rate for children and adults.

11:00

4aPP10. Predicting speech intelligibility based on the modulation spectrum and modulation frequency selectivity. Torsten Dau (Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, tdau@elektro.dtu.dk)

Speech intelligibility depends on factors related to the auditory processes involved in sound perception as well as on the acoustic properties of the sound entering the ear. A clear understanding of speech perception in complex acoustic conditions remains a challenge. Here, a computational modeling framework is presented that attempts to predict the speech intelligibility obtained by normal-hearing and hearing-impaired listeners in various adverse conditions. The model combines the concept of envelope frequency selectivity in the auditory processing of the sound with a decision metric that is based either on the signal-to-noise envelope power ratio or a correlation measure. The proposed model is able to account for the effects of stationary background noise, reverberation, nonlinear distortions and noise reduction processing on speech intelligibility. However, due to its simplified auditory preprocessing stages, the model fails to account for the consequences of individual hearing loss on intelligibility. To address this, physiologically inspired extensions of the auditory preprocessing in the model are combined with the modulation-frequency selective processing and the back-end processing that have been successful in the conditions tested with normal-hearing listeners. The goal is to disentangle the consequences of different types of hearing deficits on speech intelligibility in a given acoustic scenario.

11:15

4aPP11. Auditory enhancement and other context effects in normal, impaired, and electric hearing. Andrew J. Oxenham, Heather A. Kreft, and Lei Feng (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Neal Viemeister's contributions to our understanding of auditory enhancement stood out because they combined novel and intriguing empirical data with a testable theoretical framework to explain the results. Thirty-five years later, the resultant "adaptation of inhibition" hypothesis remains the default explanation for auditory enhancement effects, and it can be used to account for both basic auditory context effects as well as speech context effects. Recent work in our lab has focused on comparing speech and non-speech context effects in people with normal and impaired hearing, as well as cochlear implants, in an attempt to elucidate the underlying neural mechanisms and to work towards compensating for any loss of context effects in clinical populations via signal processing. This presentation will provide a survey of recent progress in this area with examples from simultaneous masking, forward masking, and speech discrimination experiments. [Work supported by NIH grant R01DC012262.]

11:30–11:35 Panel Discussion

Session 4aSA

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

Christina J. Naify, Cochair
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Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Invited Papers

9:00

4aSA1. Exploring heterogeneity and disorder In tunable elastic metamaterials. Paolo Celli, Behrooz Yousefzadeh, Chiara Daraio (California Inst. of Technol., Pasadena, CA), and Stefano Gonella (Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455-0116, sgonella@umn.edu)

The dominant paradigm in the design of elastic metamaterials revolves around periodic arrays of identical resonating elements, which are known to allow subwavelength locally-resonant bandgaps. In this work, we deliberately explore the wave manipulation capabilities of metamaterial configurations that are characterized by significant heterogeneity and disorder. Heterogeneity is here intended as the coexistence of multiple types of resonators tuned to resonate at distinct frequencies. Disorder refers to the arrangement of the resonators, which is randomized in space. The metamaterials of choice are thin elastic sheets endowed with heterogeneous populations of tunable telescopic pillars whose resonant characteristics can be agilely tuned through simple manual operations. The configurability of our experimental platform allows to seamlessly fabricate and compare a multitude of specimens, thus allowing to extract some empirical yet general rules. We first document how heterogeneity can lead to broadband mechanical filtering. More specifically, we illustrate how randomized spatial arrangements consistently outperform their functionally graded counterparts in terms of maximum achievable bandgap width. Our investigation shows that the design space of mechanical metamaterials can be stretched beyond the limits imposed by order and homogeneity, thus highlighting the engineering significance of the emerging concepts of organized disorder and design modularity.

9:20

4aSA2. Design, optimization, and fabrication of mechanical metamaterials for vibration control. Timothy F. Walsh, Chris Hammetter (Computational Solid Mech. and Structural Dynam., Sandia National Labs., PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov), Michael B. Sinclair, Harlan Shaklee-Brown (Electron., Optical, Nano, Sandia National Labs., Albuquerque, NM), Joe Bishop (Solid Mech., Sandia National Labs., Albuquerque, NM), and Wilkins Aquino (Civil Eng., Duke Univ., Durham, NC)

Harsh shock and vibration environments are common in engineering applications. Mechanical metamaterials are showing significant potential as candidates for controlling wave propagation and isolating sensitive structural components, but require proper design of their complex microstructures. In this talk we will present time and frequency-domain strategies for Partial Differential Constrained (PDE)-constrained design optimization of locally resonant elastic/acoustic metamaterials. We will present a variety of notch filter resonators and split ring cylinder/sphere geometries that can be easily optimized for wave control applications, along with fabrication details involving multi-material additive manufacturing. A frequency-domain approach will be presented for band-gap or notch filter materials, and a time-domain strategy for transient shock environments. As the metamaterial structures typically involve concentrated masses and elastic connections, an optimization strategy involving 3D simple shapes will be presented. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC., a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration. With main facilities in Albuquerque, N.M., and Livermore, C.A., Sandia has major R&D responsibilities in national security, energy and environmental technologies, and economic competitiveness.]

9:40

4aSA3. Extraordinary wave phenomena in active acoustic metamaterials. Romain Fleury (EPFL, EPFL - STI - LWE, ELB 033 - Station 11, Lausanne 1015, Switzerland, romain.fleury@epfl.ch)

Non-Hermitian sonic metamaterials are artifical acoustic media that exploit gain and loss as an extra degree of freedom to induce novel physical effects that cannot be obtained with lossless or trivial lossy systems. In this talk, we will review our recent theoretical and experimental results about one-dimensional non-Hermitian acoustics, highlighting the ability of active metamaterials to overcome limitations of passive metamaterials by engineering the acoustic power flow at the microscopic scale. We will discuss various experiments employing a set of electroacoustic resonators in a pipe, engineered to provide tailored distributions of acoustic gain or loss, and discuss the potentials of such systems in terms of sound manipulation. Altogether, our work points out the large potential and rich physics of non-Hermitian acoustic metamaterials.

Contributed Papers

10:00

4aSA4. Impact wave attenuation using Maxwell-type oscillators in dissipative elastic metamaterials. Sagr Alamri, Bing Li, and Kwek Tze Tan (Mech. Eng., Univ. of Akron, 302 E Buchtel Ave, Akron, OH 44325, sma114@zips.uakron.edu)

This work studies the development of a dissipative elastic metamaterial with single and dual-Maxwell type resonator for stress wave mitigation. Mass-spring-damper elements are used to model and analyze the mechanism of wave dissipation effect on the vibration characteristics. It is found that broadband wave attenuation region can be obtained and expanded to a wider range by properly utilizing interactions from resonant motions and viscoelastic effects of the Maxwell-type oscillators. In addition, numerical verifications are conducted for various cases, and excellent agreement between theoretical and numerical frequency response functions are obtained. The design of this dissipative metamaterial system is further applied for blast wave mitigation. By means of the bandgap merging effect that is induced by the Maxwell-type oscillator, the passing blast wave can be almost completely suppressed in the low-frequency range. A significantly improved performance of the proposed dissipative metamaterials for stress wave mitigation is verified in both time and frequency domains.

10:15–10:30 Break

10:30

4aSA5. Broadband asymmetric wave transmission in dissipative acoustic/elastic metamaterials with diatomic oscillators. Bing Li, Sagr Alamri, and Kwek Tze Tan (Dept. of Mech. Eng., The Univ. of Akron, The University of Akron, Akron, OH 44325, bingli@uakron.edu)

Asymmetric acoustic/elastic wave transmission have attracted increasingly attention recently. However, the propagation unidirectionality is always confined to a very narrow frequency band. This paper presents the development of a dissipative acoustic/elastic metamaterial with diatomic oscillators to realize "one-way" transmission in a broadband range. The effect of various types of dissipative oscillators on the asymmetric wave transmission is theoretically investigated and systematically discussed. By virtue of the surficial vibrational modes induced by the alternately arranged diatomic resonators and the merging effect of dissipative dashpot, the asymmetric transmission band is significantly broadened. Numerical results in frequency and time domains are verified using both lattice structures and continuum models. Preliminary experimental verification is further conducted. The broadband asymmetric transmission can be theoretically predicted and mathematically manipulated by careful design of the unit size parameters and deliberate selection of material properties, which could be potentially beneficial for applications in directional transducer and noise control.

10:45

4aSA6. Underwater acoustic ground cloak development and demonstration. Peter Kerrian, Amanda Hanford, Benjamin Beck, and Dean Capone (Appl. Res. Lab., Pennsylvania State Univ., PO Box 30, State College, PA 16804, ald227@psu.edu)

Acoustic ground cloaks, which conceal an object on a rigid surface, utilize a linear coordinate transformation to map the flat surface to a triangular void by compressing space into two triangular cloaking regions consisting of a homogeneous anisotropic acoustic metamaterial. Transformation acoustics allows for the realization of a coordinate transformation through a reinterpretation of the scale factors as a new material in the original coordinate system. An underwater acoustic ground cloak was constructed from perforated steel plates and experimentally tested to conceal an object on a pressure release surface. The perforated plate acoustic ground cloak successfully cloaked the scattered object. There was excellent agreement between the phase of the surface reflection and cloak reflection with a small amplitude difference. Above 15 [kHz], the cloaking performance decreased as the effective material parameters of the perforated plate metamaterial deviated from the required material parameters.

11:00

4aSA7. Comparative study of impurities in lithium tantalite and lithium niobate. Chandrima Chatterjee (Phys. and Astronomy, The Univ. of MS, University, Lewis Hall, Rm. 108, University, MS 38677), Daniel Miller (Phys. and Astronomy, The Univ. of MS, Oxford, MS), and Igor Ostrovskii (Phys. and Astronomy, The Univ. of MS, MS, iostrov@phy.olemiss.edu)

S 38677 The point defects in ferroelectrics may influence the physical properties of phononic crystals $LiTaO_3$ (LT) and $LiNbO_3$ (LN). The initial wafers, periodically poled (PP) structures of LT with 1-mm-domains, and PP-LN with 0.45-mm-domains are investigated. Photoluminescence (PL) is taken at room temperature 310-nm -excitation. The slit width in optical spectrometer is set to 1 nm. The PL spectra of LT reveal multiple impurities including Fe^+,W, Th, Cs, Kr, Cu, Xe. Unlike PL from LN, there are no lines of the F-center, Ar and some other elements. The intensities of PL-lines in both LT and LN demonstrate a saw-type periodicity along the direction normal to the Z-axis. For instance, the PL line of Fe^+ defect in LT varies $\pm$ 15% with periodicity about 200 $\pm$ 40 nm. In the case of PP LT phononic crystals, the difference in Fe^+ line amplitude remains about the same; however, the periodicity becomes equal to a ferroelectric domain length of 1 mm. There is a difference between the two periodicities of charged defects in PP-LT and PP-LN; the positions of maxima in PL-lines are close to the interdomain walls in PP-LN, but it is not the case for PP-LT. Possible practical applications are discussed.

11:15

4aSA8. On the behavior of acoustic/elastic metamaterials with anisotropic mass density. mehran jaberzadeh, Bing Li, and Kwek Tze Tan (Mech. Eng., The Univ. of Akron, Akron, OH 44325, mehran.jaberzadeh@gmail.com)

The effect of anisotropic mass density in acoustic/elastic metamaterials on wave propagation is presented in this research. The use of microstructures to achieve anisotropic physical properties in metamaterials is an intensive field of study. In this work, a two-dimensional (2D) 'mass-in-mass'-spring lattice system is utilized to achieve anisotropic effective mass density in two orthogonal principal directions.In each direction, the effective mass density is frequency-dependent and can be "effectively negative" if it falls within the frequency bandgap region. A 2D numerical continuum model,-based on a recently developed cantilever-in-mass model, is further presented and examined to study how wave input angles affect 2D wave propagation. Results show that wave attenuation is dependent on both input frequency and wave input angle in a 2D metamaterial. This study demonstrates a case whereby wave attenuation is achieved,by selecting wave input frequency in the bandgap region to enact "negative effective mass density". This study also illustrates another case when wave attenuation is obtained for positive anisotropic mass density. This behavior is achieved by directing wave propagation to transverse direction at a specific input wave angle of 60°. Both numerical calculations of mass-spring lattice model and continuum cantilever-in-mass model show good agreement.

11:30

4aSA9. Non-reciprocal wave phenomena through pump-signal wave interaction in discrete systems with asymmetric subwavelength geometry. Samuel P. Wallen, Benjamin M. Goldsberry, and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu)

Acoustic non-reciprocity has been shown to enable a plethora of effects analogous to phenomena seen in quantum physics and electromagnetics, such as immunity from back-scattering, unidirectional band gaps, and topologically protected states, which could lead to the design of direction-dependent acoustic devices. One class of material that holds promise as a means to achieve acoustic non-reciprocity is the "Willis medium," which exhibits strain-momentum coupling owing to asymmetry and non-local effects in the material microstructure. This work considers a method for obtaining non-reciprocal elastic wave propagation in a system with subwavelength asymmetry inspired by asymmetric Willis microstructures. Non-reciprocity is achieved through application of a slowly-varying pump wave that acts as a spatio-temporal modulation of the material, which is modeled as a discrete system with geometric nonlinearity. The modulation generates momentum bias for a fast-varying signal wave with pump-signal interaction enabled by weak nonlinearity associated with the variation of subwavelength asymmetric geometry. It is shown that momentum bias may be generated with a pump wave acting transverse to the direction of propagation, which may facilitate experimental realizations. [This work was supported by the National Science Foundation.]

THURSDAY MORNING, 10 MAY 2018 — NICOLLET A, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Perception and Processing of Voice and Speech (Poster Session)

Terrin N. Tamati, Chair

Department of Otorhinolaryngology / Head and Neck Surgery, University Medical Center Groningen, Hanzeplein 1, Groningen 9700RB, Netherlands

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 authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSC1. The impact of perceptual load on natural and synthetic speech perception. Adriana Ojeda, Ethan Kutlu, and Ratree Wayland (Dept. of Linguist, Univ. of Florida, P.O. Box 115454, Gainesville, FL 32611-5454, ojedaae13@ufl.edu)

Previous work has shown that the availability and the weighting of different types of information (e.g., lexical, segmental/subsegmental, and metrical prosody) for speech segmentation are gradiently affected by perceptual and cognitive load. Specifically, it has been shown that severe energetic masking increases reliance on the acoustic, sub-segmental, context-dependent coarticulation information and decreases reliance on lexical information (e.g., Mattys *et al.*, 2009). In this study, the effects of energetic masking on the intelligibility of synthetic and natural speech (controlled for lexical and metrical prosody) will be compared. Since fine acoustic detail linked to coarticulation is lacking in synthetic speech in comparison to naturally produced speech, energetic masking should result in greater reliance on lexical information than coarticulatory information on synthetic speech processing while the opposite will be true for natural speech. *Reference* Mattys, S. L., Brooks, J., & Cooke, M. (2009). Recognizing speech under a processing load: Dissociating energetic from informational factors. Cognitive Psychology, 59, 203-243.

4aSC2. Asymmetries in vowel perception arise from phonetic encoding strategies. Matthew Masapollo (Boston Univ., 677 Beacon St., Boston, MA 02215, mmasapol@bu.edu), Lauren Franklin, James Morgan (Cognit., Linguistic & Psychol. Sci., Brown Univ., Providence, Rhode Island), and Linda Polka (McGill Univ., Montreal, QC, Canada)

Directional asymmetries in vowel discrimination studies reveal that speech perceivers (both adult and infant) are biased toward extreme vocalic articulations, which lead to acoustic vowel signals with well-defined spectral prominences due to formant convergence. These directional effects occur with vowel stimuli presented in either the acoustic or the visual modality and are independent of specific linguistic experience. Current research is focused on elucidating the perceptual processes underlying this universal vowel bias. In the present investigation, the inter-stimulus interval (ISI) in AX discrimination tasks for unimodal acoustic and visual vowels was manipulated (500 ms vs. 1000 ms) in order to examine whether asymmetries are present under experimental conditions that reduce demands on attention and working memory. Subjects discriminated either video-only or audio-only tokens of naturally-spoken English [u] and French [u] which differ in their degree of visible lip-rounding and proximity between F1 and F2. We observed robust asymmetries with these stimuli with English- and French-speaking adults in earlier studies using a relatively long ISI (1500 ms). The present results demonstrated that asymmetries in both auditory and visual vowel discrimination are diminished in the short ISI conditions, suggesting that this vowel bias derives from phonetic encoding processes, rather than general psychophysical processes.

4aSC3. Formant pattern and spectral shape ambiguity in vowel synthesis: The role of fundamental frequency and formant amplitude. Thayabaran Kathiresan (Dept. of Computational Linguist, Univ. of Zurich, Andreasstrasse 15, Zurich, Zurich 8050, Switzerland, thayabaran.kathiresan@uzh.ch), Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zurich, Switzerland), Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zürich, Switzerland), and Volker Dellwo (Dept. of Computational Linguist, Univ. of Zurich, Zurich, Switzerland)

When investigating formant pattern and spectral shape ambiguity in Klatt synthesis, an earlier study showed that the perceived vowel quality of Standard German vowel sounds can be changed by varying fundamental frequency

only [Maurer *et al.* (2017). J. Acoust. Soc. Am. **141**(5):3469-3470]. In this follow-up study, the previous original synthesis experiment was repeated twice, firstly, with fundamental frequencies (f_o) of the corresponding sounds lowered by one octave, and secondly, with different ratios of the first and second formant amplitudes. Here, the role of the f_o range and the formant amplitudes for the investigation of formant pattern and spectral shape ambiguity in vowel synthesis was further examined. The same five phonetic expert listeners that participated in the previous experiment also identified all of the newly synthesised sounds in a multiple-choice identification task. Results revealed that the perceived vowel quality only changes for f_os above 200 Hz and that, for back vowels, the ratio of the formant amplitudes used in the Klatt synthesis also affects vowel recognition. Thus, the results of the experiments confirm earlier indications of a non-systematic relation between f_o or pitch and formant patterns or spectral envelopes for vowel recognition.

4aSC4. Sinewave vowel sounds: The role of vowel qualities, frequencies and harmonicity of sinusoids, and perceived pitch for vowel recognition. Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch), Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zürich, Switzerland), Thayabaran Kathiresan, and Volker Dellwo (Dept. of Computational Linguist, Univ. of Zurich, Zurich, Zurich, Switzerland)

In the literature, the recognition of sinewave vowels replicating statistical formant patterns is reported as impaired when compared to natural sounds. However, the corresponding formant simulating sinusoids were harmonically unrelated, with synthesised signals only accidentally being quasi-periodic, and vowel confusion was indicated to relate to vowel height. Involving five phonetic expert listeners, the present study tested vowel and pitch recognition of three sinusoid replicas based on statistical F_1-F_2-F_3 patterns of the Standard German closed and mid-open vowels /i-y-e-ø-o-u/ for women, "corrected" approximations of these patterns with harmonically related sinusoids, and harmonical patterns with fixed first and third sinusoids, yet varying only the second sinusoid so as to effect a change in harmonic relation. The results showed strong vowel confusions for mid-open but only limited confusions for closed vowels. Additional effects on vowel recognition were indicated to concern harmonicity and specific frequencies of the sinusoids, and perceived pitch (range recognised = 165-440Hz). Thus, sinewave replications of formant frequencies seem to represent perceived vowel qualities not per se but only in relation to specific vowel qualities, sinusoid configuration and pitch, supporting earlier claims of spectral representation of vowel quality as being non-systematic and pitch-related.

4aSC5. Trial-to-trial variability in talkers' fundamental frequencies restrains spectral context effects in vowel categorization. Ashley Assgari and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu)

Perception of a given sound is influenced by spectral properties of surrounding sounds. For example, listeners perceive /ɪ/ (low F1) more often when following sentences filtered to emphasize high-F1 frequencies, and perceive /ɛ/ (high F1) more often following sentences filtered to emphasize low-F1 frequencies. These biases in vowel categorization are known as *spectral contrast effects* (*SCEs*). When preceding sentences were spoken by acoustically similar talkers (low variability in mean f0), SCEs biased vowel categorization, but sentences spoken by acoustically different talkers (high variability in mean f0) biased vowel categorization significantly less (Assgari *et al.*, 2016 ASA). However, it was unclear whether these effects varied due to local (trial-to-trial) or global (across entire block) variability in mean f0. Here, the same sentences were arranged to increase/decrease monotonically in mean f0 across trials (low local variability) or vary substantially from trial-to-trial (high local variability) with equal global variability. On each trial, listeners heard a sentence filtered to add a low-F1 or high-F1 spectral peak to bias categorization of a subsequent vowel (/ɪ/-/ɛ/ continuum). Sentences with low local variability in mean f0 biased vowel categorization significantly more than sentences with high local variability. Relevance to studies of talker normalization will be discussed.

4aSC6. Listeners compensate for prosodic position when categorizing word-final obstruents. Jeremy Steffman (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jsteffman@g.ucla.edu)

This study explores how perceptual sensitivities to contextual variability extend to prosodically induced variation. In English, vowel duration is reliably longer preceding a voiced obstruent, as opposed to a voiceless obstruent, and listeners use preceding length as a cue to obstruent voicing (e.g. Raphael, 1972). Segmental duration also co-varies systematically with prosodic position, being longer phrase-finally (e.g. Turk & Shattuck-Hufnagel, 2007). In this study, we tested the extent to which listeners' categorization of word-final obstruents is altered by the prosodic position of the target. Participants heard a continuum that varied only in vowel length, and categorized stimuli as either *"coat"* or *"code"*. Prosodic position in a carrier phrase was manipulated by splicing the target word into either a phrase-final or phrase-medial context. Results show listener expectations about phrase final lengthening mediate categorization, with significantly longer vowel durations required in phrase-final position for a *"code"* response. The results are discussed in terms of other top-down influences on segmental categorization (e.g. lexical information), and implications for further work on the prosody-phonetics interface.

4aSC7. Evaluating mechanisms underlying nonspeech context effects in coarticulatory compensation. Navin Viswanathan (Speech-Language-Hearing, Univ. of Kansas, 1000 Sunnyside Ave, Lawrence, KS 66045, navin@ku.edu)

Human speech listeners overcome variability in the speech signal due to different speakers, rates, phonetic contexts etc., by demonstrating context-appropriate perceptual shifts. From a general auditory perspective, perceptual systems heighten contrastive spectral and temporal properties of the acoustic signal to help listeners cope with variability (Diehl *et al.*, 2004). In this study, I focus on a spectral contrast account of perceptual coarticulatory compensation and evaluate the claim that listeners cope with coarticulation by tracking spectral averages across multiple segments (Holt, 2005) while ignoring spectral variability (Holt, 2006). In Experiment 1, using tone analogue contexts (Lotto & Kluender, 1998) I created multi-tone contexts such that the global spectral average of each trial was pit against the frequency of the final tone that immediately preceded the target speech. Interestingly, nonspeech context effects were determined by the frequency of the final tone rather than the global trial average. A follow up experiment 2 indicated that contexts with identical overall spectral averages can produce different effects on following speech depending on the final tone. These findings taken together call into question the assumed mechanisms underlying spectral contrast.

4aSC8. The role of segment probability in perception of speech sounds. Seongjin Park, Maureen Hoffmann, Priscilla Z. Shin, and Natasha L. Warner (Dept. of Linguist, Univ. of Arizona, P.O. Box 210025, Tucson, AZ 85721, seongjinpark@email.arizona.edu)

This study investigates how listeners utilize segment probability in speech perception. In Warner *et al.* (2014), participants identified gated fragments of all combinations of two segments in English (e.g., diphones /hə/, /dt/, /ab/, /iɛ/). That data was further analyzed to study the influence of segment probability. Overall probability of the second segment (e.g., the probability of /t/ in English), and conditional probability of the second segment given the first segment (e.g., the probability of /b/ given preceding /a/) were calculated from the CMU pronunciation dictionary. There were significant correlations between accuracy of segment perception and both overall and conditional probability of second segments at early gates of the diphone, when few or no acoustic cues to the second segment were available (for example, perception of /b/ in /ab/ by one-third of the way through the /a/). However, when participants had enough acoustic information, the correlation was smaller or non-significant. The present study shows that English listeners utilize probability in identifying English segments when few acoustic cues are available, whereas Dutch listeners in a matched previous study on perception of Dutch (Warner *et al.* 2005) utilized probability only for a few unusual cases.

4aSC9. A replication of competition and prosodic effects on spoken word recognition. Natasha L. Warner, Genesis G. Hernandez, Seongjin Park (Dept. of Speech and Hering Sci., University of Arizona, Box 210071, Tucson, AZ 85721, genesishernandez@email.arizona.edu), and James M. McQueen (Radboud Univ., Nijmegen, Netherlands)

Despite the absence of clear and reliable word boundary markers, listeners recognize words in spoken sentences. A previous study tested a spoken-word recognition model, Shortlist, by asking speakers of British English to identify real words within nonwords (McQueen, Norris & Cutler, 1994). Some words were embedded within the onsets of longer words with either a weak-strong (WS) or strong-weak (SW) stress pattern (e.g., “mess” in /dəmɛs/, the onset of “domestic,” “sack” in /sækɹɪf/, the onset of “sacrifice), and other words were embedded without a real word onset with either a WS or SW pattern (e.g., /nəmɛs/ or /mɛstəm/ for “mess” and /sækɹək/ or /kləsæk/ for “sack”). The original study reported both competition effects (e.g. competition from “domestic” hindered recognition of “mess) and prosodic effects (e.g. the stress in “mess” facilitated segmenting it from the preceding context). This current study aimed to replicate these results in American-English. Despite the different listener population and dialect, pilot results for the current study confirm both types of effect. This data will be used to test a new American English version of the Shortlist-B model of spoken word recognition (Norris & McQueen, 2008).

4aSC10. Cortical entrainment to speech under competing-talker conditions: Effects of cognitive load due to presentation rate and task difficulty. Paul Iverson, Jieun Song, and Holly Bradley (Univ. College London, 2 Wakefield St, London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

Previous work has demonstrated that, when listeners attend to a target talker and ignore a distractor, their neural activity is more strongly entrained to the amplitude envelope of the target speech. Moreover, target-talker entrainment under these conditions can be higher for second-language listeners than native listeners, which might be due to the greater focused attention required to understand second-language speech. The present study manipulated the cognitive difficulty of the task in an attempt to make native-language listeners have entrainment results more like those of second-language listeners. Listeners heard simultaneous sentences spoken by male and female speakers, and had to detect semantically anomalous catch trials. The sentences had high or low semantic predictability (low-predictability sentences require more lexical processing to evaluate meaning) and were presented with short or long intervals between sentences (0.5 or 3.0 s, with shorter intervals forcing listeners to make a quicker evaluation of meaning). The preliminary results suggest that native-language listeners indeed have higher target-talker entrainment under more cognitively difficult conditions, suggesting that cognitive load may modulate focused attention to the target speech signal.

4aSC11. Electrophysiological measures of listening effort and comprehension: Speech in noise, vocoders, and competing talkers. Anna Exenberger and Paul Iverson (Univ. College London, 2 Wakefield St, London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

This study explored the relationship between signal degradation and electrophysiological measures of speech processing, in order to understand how these measures relate to intelligibility and listening effort. Subjects heard two talkers presented to different ears, with the target talker presented in quiet, three levels of speech-shaped noise (+3 dB, -0.5dB, -4dB SNR), or a 14-channel noise vocoder. The speech materials were simultaneous sentences that varied in final-word predictability; subjects were asked to monitor the sentences for catch trials (i.e., semantically anomalous final words). The results demonstrated that cortical entrainment to the amplitude envelope of the target was higher than for the distractor, as found in previous studies. Entrainment and lexical processing are robust with increasing noise, then fall when the stimulus becomes very poorly intelligible. Unexpectedly, results from the vocoded condition indicate no entrainment advantage for the target speaker, even when this condition is intelligible.

4aSC12. The time course of word recognition in younger and older adults: Signal degradation and lexical challenge. Kristin Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu), Avanti Dey (Psychol. and Brain Sci., Washington Univ. in St. Louis, New York, NY), Nichole Runge, Mitchell Sommers (Psychol. and Brain Sci., Washington Univ. in St. Louis, Saint Louis, MO), Brent Spehar (Otolaryngol., Washington Univ. in St. Louis, St. Louis, MO), and Jonathan E. Peelle (Otolaryngol., Washington Univ. in St. Louis, Saint Louis, MO)

As listeners age, speech recognition often becomes more challenging, especially in noisy environments. Older adults also appear to have more difficulty resolving lexical competition than do younger adults. Relatively little is known, however, about how such signal- and lexicon-related challenges interact to affect the timing of word recognition and how such effects might change across the lifespan. In this study, we use a visual world paradigm to investigate the effects noise, word frequency, and phonological neighborhood density on the time course of spoken word recognition in young and older adults. On each trial, listeners saw four items on a computer screen and heard “Click on the” followed by a target word that corresponded to one of the images. None of the other items on the display were phonologically or semantically related to the target. Eye-tracking results show that the time course of word recognition is predicted by age, noise, frequency, density, and several of their interactions. In general, high-frequency words and words from dense phonological neighborhoods were fixated more quickly, while noise and age slowed recognition. These results support a framework in which lexical and acoustic factors co-determine the cognitive challenges associated with speech perception.

4aSC13. Semantic context modulates intelligibility advantage of clear speech in temporally compressed sentences. Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), Rajka Smiljanic, and Sandie Keerstock (Linguist, Univ. of Texas at Austin, Austin, TX)

Sentences produced in a clear speech speaking style are typically more intelligible than sentences produced in a conversational style. However, factors that lead to the intelligibility difference are not fully understood. This study investigated the role of semantic context in intelligibility of temporally compressed clear and conversational speech. Grammatically correct English sentences, which were either semantically meaningful or semantically anomalous, were presented to 70 normal-hearing listeners in clear or conversational speaking style. The sentences were temporally compressed to produce seven different syllabic rates (6-12), matched across clear and conversational sentences to control for the original durational differences between clear and conversational speech. Results for semantically anomalous sentences indicate no differences in the rate of intelligibility decline across compression rates for clear and conversational speech. In contrast, for semantically meaningful sentences, the rate of intelligibility decline was smaller for the semantically meaningful clear-speech sentences over semantically anomalous conversational-speech sentences. At the highest compression rate of 12 syllables per second, intelligibility of semantically meaningful clear-speech sentences (~68% correct) was similar to that of semantically anomalous sentences at 9 syllables per second. These findings suggest that semantic context can modulate the effect of speaking style on the intelligibility of temporally compressed sentences.

4aSC14. Ganong shifts for noise- and sine-vocoded speech continua in normal-hearing listeners. Brian Roberts and Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

Speech perception involves interpreting incoming sensory information in the context of stored linguistic knowledge. Lexical bias is the tendency to perceive an ambiguous speech sound as a phoneme that completes a word rather than a non-word; the more ambiguous the auditory signal, the greater the reliance on lexical knowledge. For example, a speech sound ambiguous between [g] and [k] is more likely to be perceived as [g] when preceding

[Ift] and as [k] when preceding [Is]. The magnitude of this between-context difference—the Ganong shift—also increases when the listener experiences high cognitive load, limiting the resources available to process acoustic-phonetic information. In this study, an eight-step series between monotonized natural tokens of [gI] and [kI] was created using morphing software. Two continua—"gift"-"kift" and "giss"-"kiss"—were derived by adding [ft] or [s] to each step. High-resolution noise- (16-band) and sine-vocoded (32-band) versions were derived from these continua. Ganong shifts were substantially larger for sine- than for noise-vocoded stimuli, despite the greater number of channels for the former and similar slopes for the identification functions, perhaps reflecting the higher cognitive load associated with the relative spectral sparsity and less natural timbre of the sine-vocoded stimuli. [Work supported by ESRC.]

4aSC15. Across-ear integration of acoustic-phonetic information under competitive conditions: Effects of attenuation increase in the presence of an extraneous formant. Robert J. Summers and Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk)

Previous research using consonant-vowel syllables where one ear receives the first formant (F1) and the other receives the second and third has shown that dichotic release from masking allows F2 + F3 to remain effective speech cues even after substantial attenuation. This study used three-formant analogues of natural sentences and extended the approach to include competitive conditions. Target formants were presented dichotically (F1 + F3; F2), either alone or accompanied by an extraneous competitor for F2 (i.e., F1 ± F2C + F3; F2) that listeners must reject to optimize recognition. In experiment 1, F2C was absent and F2 was attenuated in 6-dB steps (range: -6-48 dB). Intelligibility was unaffected until attenuation >30 dB; F2 still provided useful information at 48-dB attenuation. In experiment 2, F2 was attenuated in 12-dB steps (range: 0-36 dB). F2C was created by inverting the F2 frequency contour and using the F2 amplitude contour without attenuation. When F2C was present, 12-dB attenuation was sufficient to cause some loss of intelligibility; the effect was large when attenuation ≥ 24 dB. This interaction suggests that some mandatory across-ear spectral integration occurs, such that informational masking arising from F2C rapidly swamps the acoustic-phonetic information carried by F2 as its relative level is attenuated. [Work supported by ESRC.]

4aSC16. Multimodal perception of interrupted speech and text by younger and older adults. Rachel E. Miller and Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1229 Marion St, Columbia, SC 29201, remiller@email.sc.edu)

The current study aimed to identify how partial auditory (speech) and visual (text) information are resolved for sentence recognition by younger and older adults. Interrupted speech was created by periodically replacing speech with silent intervals at various interruption rates. Likewise, interrupted text was created by periodically replacing printed text with white space. Speech was further processed by low-pass filtering the stimulus spectrum at 2000 Hz. Speech and text stimuli were interrupted at 0.5, 2, 8, and 32 Hz and presented in unimodal and multimodal conditions. Overall, older listeners demonstrated similar performance to younger listeners on the unimodal conditions. However, there was a marked decrease in performance by older listeners during multimodal testing, suggesting increased difficulty in integrating information across modalities. Performance for both groups improved with higher interruption rates for both text and speech in unimodal and multimodal conditions. Listeners were also able to gain benefit from most multimodal presentations, even when the rate of interruption was mismatched between modalities. Supplementing speech with incomplete visual cues can improve sentence intelligibility and compensate for degraded speech in adverse listening conditions. However, the magnitude of multimodal benefit appears mitigated by age-related changes in integrating information across modalities. [Work supported, in part, by NIH/NIDCD.]

4aSC17. Musical training and the perception of phonetic detail in a shadowing task. Grace Ji Yan Tsang (Communicative Sci. and Disord., New York Univ., 665 Broadway, Fl. 9, New York, NY 10012, gjt238@nyu.edu), Edmund L. Dana, Morwaread M. Farbood (Music Technol., New York Univ., New York, NY), and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Phonetic convergence, or shadowing, is the phenomenon in which people unintentionally and temporarily change phonetic details of their speech to sound more similar to another talker. Previous research has examined effects of the shadower and of the target speaker. In the current study, we examine listener contributions on the perception of phonetic convergence. Two hundred and sixty listeners completed an AXB perception task in which they were asked to determine whether the first or third stimulus was a better imitation of the middle stimulus. Listeners provided information about previous instrumental and vocal training. Preliminary examination of the results reveals that listeners with more musical training (in the form of instrumental or vocal experience) lead to better accuracy. Results from the study suggest that there is a link between musical training and detection of fine-grained phonetic details in speech.

4aSC18. Is the target of imitation directly acoustic or a pattern within a speaker's phonetic system? Rebecca Scarborough (Linguist, Univ. of Colorado, 295 UCB, Boulder, CO 80309, rebecca.scarborough@colorado.edu) and Kuniko Nielsen (Linguist, Oakland Univ., Rochester, MI)

Speakers may converge phonetically with another talker to whose speech they are exposed, resulting in reduced acoustic distance between interlocutors. However, it is unclear whether the target of imitation is raw acoustics or a linguistic pattern. Zellou *et al.* (2016) present a case that distinguishes these possibilities: after listening to a model talker whose speech was manipulated to reduce coarticulatory vowel nasality (measured acoustically in A1-P0), participants' nasality also decreased in post-test productions, relative to their baseline. However, the model speaker had naturally low A1-P0 (correlated with high nasality), and even after the manipulation to raise A1-P0, it was lower than all participants'. If the target of imitation is acoustic, participants *diverged* from the model (raised A1-P0). But if the target of imitation is a linguistic pattern (decreased coarticulatory nasality), participants *converged*. The current study uses an AXB perceptual task to determine whether it is acoustically more similar items (more nasal; speakers' baseline) or linguistically more similar items (less nasal; post-test) that listeners judge as "more similar" to the model talker's production. A generalized linear mixed model showed a significant preference for post-test items (55.4%, $z = 5.19$), indicating that linguistic similarity is the basis for listeners' similarity judgments.

4aSC19. Power and phonetic accommodation. Auburn Lutzross, Andrew Cheng, Alice Shen, Eric Wilbanks, and Azin Mirzaagha (Dept. of Linguist, Univ. of California Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94704, azshen@berkeley.edu)

Theories of speech accommodation claim that phonetic convergence to one's interlocutor is automatic (Goldinger, 1998; Shockley *et al.*, 2004). However, there is evidence that individual traits and social factors modulate accommodation so that speakers may even diverge from their interlocutor (Babel, 2010; Bourhis & Giles, 1977). This study investigates how talker's personal sense of power and power relative to the interlocutor ("interpersonal power") affect phonetic accommodation. Personal power was manipulated through a thought experiment in which participants described a time they felt powerful or powerless. Interpersonal power was manipulated through randomly assigned roles in a recorded interview with a confederate where the Inventor (powerless) pitched an entrepreneurial idea to the Investor (powerful). Participants were considered to have converged if the difference in pitch between the participant and confederate had decreased in post-interview versus pre-interview recordings. We found that though interpersonal power does not influence the direction or degree of accommodation, personal power does. Speakers with lower personal power diverged from their interlocutor ($t = -2.1389$, $p = 0.04$), indicating that an individual's self-perception has a stronger effect on accommodation than the established roles in the interaction.

4aSC20. The effects of lexical and phonological factors on talker processing. Sandy Abu El Adas and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, sandyabu@nyu.edu)

Previous research has shown that listeners are better at processing talker information in their native language compared to an unfamiliar language, a phenomenon known as the Language Familiarity Effect. Several studies have explored the cause of this effect. Some have argued that it is tied to the lexicon and the ability to comprehend, while others have suggested that it is the familiarity with the phonology. In the current study, we use an AX discrimination task to test these two factors simultaneously by manipulating lexical status (words/nonwords) and phonotactic probability (high/low). We also test individual differences in reading ability, as poor phonological awareness skills are linked to reading impairment. Twenty-four native speakers of American English completed the AX task and a battery of phonological awareness tasks. Reaction time results revealed an interaction between lexical status and phonotactic probability: words with high phonotactic probability were processed faster than words with low phonotactic probability, but no difference was found for the nonwords. Sensitivity (A') revealed no effects of lexical status or phonotactic probability, but did reveal that listeners with higher reading scores were more sensitive to talker differences.

4aSC21. Effects of training on the perception of talker information in degraded speech. Terrin N. Tamati (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen 9700RB, Netherlands, t.n.tamati@umcg.nl)

A fundamental property of human speech perception is its robustness in a wide range of listening conditions. Listeners are able to successfully recognize and understand speech under a wide range of adverse and challenging conditions, such as in noise, reverberation, or multi-talker babble, and with noise vocoding. Noise vocoding, a manipulation that reduces spectral resolution, has been shown to result in less accurate speech comprehension and also poorer perception of indexical information (e.g., talkers' voices). However, speech comprehension quickly improves with exposure, as listeners learn to better extract information from the systematically degraded speech. The extent to which the learning of noise-vocoded speech transfers to other talkers or tasks is still largely unknown. The current study investigated whether training, consisting of transcribing sentences or identifying talkers' voices from sentences, leads to improvements in the perception of talker information. Preliminary results suggest that both training tasks resulted in improved perception of talker information, as participants learned general information about the degradation. Larger improvements were observed after talker identification training, which focused attention on talker differences. Differences between training tasks, and implications for cochlear implant users, will be discussed. [Funding: VENI Grant (275.89.035) from the Netherlands Organization for Scientific Research (NWO).]

4aSC22. Perceptual similarity judgments of voices: Effects of talker and listener language, vocal source acoustics, and time-reversal. Kristina Furbeck, Emily J. Thurston, Jessica Tin, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu)

Listeners demonstrate reliable perceptual biases favoring voices speaking their native language versus those speaking a foreign language. This "language-familiarity effect" for voice processing has been found even for perceptual similarity judgments: Voices speaking listeners' native language sound less alike, even when those recordings have been rendered incomprehensible via time-reversal. Here, we sought to replicate and extend this finding of linguistic effects on voice similarity. Native English- and Mandarin-speaking listeners (both N = 40) rated the perceptual similarity of voices speaking English or Mandarin (both N = 20) for either time-reversed or forward speech. Both listener groups tended to find Mandarin-speaking voices more dissimilar, but this effect was reduced for English-speaking listeners, especially for forward speech. Perceptual similarity judgments of voices were always highly correlated between listener groups and between forward/time-reversed speech. Acoustic measurements (voice fundamental frequency mean and variance, local jitter, harmonics-to-noise ratio, speech rate, and formant dispersion) were also made to ascertain how listeners' perceptual similarity judgments were related to speech acoustics and whether these relationships differed under time-reversal or across listeners' or talkers' native language. Overall, these data suggest that, while native language may influence perceptual dissimilarity of voices, the magnitude of these effects tend to be very small.

4aSC23. Impact of talker adaptation on speech processing and working memory. Sung-Joo Lim, Jessica Tin (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 610 Commonwealth Ave, Auditory Neurosci. Lab., Boston, MA 02215-2422, sungj.m.lim@gmail.com), Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., Boston, MA), and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Talker adaptation is known to facilitate immediate speech recognition. However, it is unclear whether and how talker adaptation also facilitates working memory for speech. Using electroencephalography (EEG) during a delayed recall of digit span task, we investigated whether talker adaptation facilitates working memory performance. We also investigated whether neural alpha oscillatory power reflects facilitatory effects of talker adaptation. Listeners encoded sequences of seven randomly ordered digits, recalling them after a 5-s retention. Digit sequences were spoken by either a single talker or multiple talkers, and were presented at either faster or slower speech rates (0-ms vs. 500-ms inter-digit intervals). Overall, listeners responded faster and more accurately for single-talker sequences compared to multiple-talker sequences. Especially for the faster presentation rate, listeners were more efficient (faster and more accurate) in recalling sequences spoken by a single talker. For the faster presentation rate, processing digit sequences spoken by single vs. multiple talkers also elicited reduced alpha power during both speech encoding and working memory retention. These results suggest that talker adaptation reduces cognitive effort during both speech encoding and memory retention, thereby producing more efficient working memory for speech information, especially when listeners process speech rapidly.

4aSC24. Re-examining the effect of top down processing on voice perception. Ashley Quinto, Kylee Kim, and Susannah V. Levi (Communicative Sci. & Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, anq203@nyu.edu)

The current study aimed to replicate a recent study conducted by Narayan, Mak, & Bialystok (2016) that found effects of top-down linguistic information on a talker discrimination task by comparing four conditions: compounds (day-dream), rhymes (day-bay), reverse compounds (dream-day), and unrelated words (day-bee). The original study used both within- and across-gender pairs and same and different trials were analyzed separately, obscuring possible response biases. Narayan *et al.* found graded performance between the four conditions, but some results were likely to have been influenced by the use of across-gender trials in the different-talker condition. In the current study, only female speakers were used and results were analyzed with signal detection theory (sensitivity and bias measures). Results revealed that participants were faster to respond to rhyming pairs than the three other conditions. In addition, participants were significantly more sensitive to talker differences in rhyming pairs than unrelated pairs, but no other conditions differed. Participants were more biased to respond "same" in the rhyme and compound conditions than in the unrelated condition. These results demonstrate a partial replication of the Narayan, Mak, & Bialystok (2016) findings, suggesting an interaction between linguistic and talker information during speech perception.

4aSC25. Perception of femininity in normally phonated and whispered speech. Nichole Houle and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, New York, NY 10012, nh1473@nyu.edu)

Many transgender women seek out voice and communication therapy to support in their transition from their biological gender to presenting as their gender identity. This has led to an increased need to examine the perception of gender and femininity to develop evidence-based therapy practices. In this study, we explored perception of femininity in normally phonated and whispered speech. Transgender male-to-female, cismale, and cisfemale speakers

were recorded producing hVd words. Naïve listeners rated femininity using a visual-analog scale. The results revealed that listeners rated speakers more ambiguously in whispered speech than normally phonated speech. Within-group analyses were conducted to further examine how speaker characteristics (height, age, mean f0, duration) contributed to perceptions of femininity. While there was a significant effect of mean f0 within the normally phonated condition for all groups, none of the other speaker or acoustic cues consistently predicted listener ratings of femininity across speaker groups.

4aSC26. Investigating the influence of listener attitudes and expectations on the intelligibility of Hindi-influenced English. Veera S. Vasandani (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, vasan007@umn.edu), Molly E. Babel (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Phoneme recognition and speech intelligibility can be affected by purely social factors, such as beliefs about or attitudes toward speakers of varying races and ethnicities (e.g., Babel & Russell, J. Acoust. Soc. Am. **137** [2015]). The current study examined whether implicit and explicit attitudes about people from South Asian (SA) affect the intelligibility of Hindi-influenced English (HIE). Three measures were used: an Implicit Association Test (IAT) to measure general associative attitudes toward SA individuals; an intelligibility task in which listeners repeated HIE and North American English (NAE) sentences in noise, accompanied by pictures of SA or white individuals; and an ethnographic interview to acquire qualitative data about stereotypes, communicative practices with nonnative speakers, and other language attitudes. Data collection is ongoing in a more ethnically diverse location (Vancouver, BC) and a more homogenous one (Minneapolis, MN). Drawing from past studies on socioindexical influences on nonnative speech, we predict that individuals who harbor negative attitudes and stereotypes about SA people will perceive HIE as less intelligible than those without such beliefs. The results of this study will inform the growing body of literature regarding the ways in which socioindexical factors influence speech recognition across individuals with diverse ethnicities and language varieties.

4aSC27. The perception of sexual orientation through speech: Generational change. Lily Obeda and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Previous research has shown that listeners can perceive the sexual orientation of some lesbian, gay, and bisexual (LGB) people at a greater than chance levels based on the acoustic-perceptual characteristics of their speech (Munson *et al.*, J. Phonetics [2006], Pierrehumbert *et al.*, J. Acoust. Soc. Am. [2004]). The cues to these judgments were the pronunciation of specific sounds (i.e., /æ/, /ɛ/, /oʊ/, /u/, /s/), rather than global speech characteristics (like f0 range or formant-frequency scaling). In the years since those studies were conducted, societal attitudes toward LGB people have changed substantially. Moreover, ongoing sound changes have affected the pronunciation of many of the sounds that cued judgments of sexual orientation from speech. The current study examines whether these changes affect the perception of sexual orientation through speech. The perception experiments described in Munson *et al.* are being redone using two groups of listeners. One group is matched in age to subjects whose data were reported previously (i.e, currently 18-30 years old). The second group that is matched in birth year to the subjects from the earlier study (i.e., currently 33-45 years old). Ongoing analyses compare these new data are compared to those from Munson *et al.*

4aSC28. Gender typicality in children's speech, reconsidered: Effects of stimulus composition on listener judgments. McKalya Beaulieu, Emily Larson, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Listeners rate the speech of boys and girls as young as four years old as sounding gendered: boys are rated as sounding boy-like and girls as girl-like (Perry *et al.*, J. Acoust. Soc. Am. [2001]). Recent research found that the extent to which boys' speech sounds boy-like is correlated with measures of their gender identity and expression (Li *et al.*, J. Phonetics [2016], Munson *et al.*, J. Acoust. Soc. Am. [2015]). Munson *et al.* found that boys with a diagnosis of gender identity disorder [GID] were rated as sounding less boy-like than boys without GID. Munson *et al.*'s experiment used only a small number of girls' productions as filler items. The current study examined listeners' ratings of the gender typicality of speech of boys with GID and both boys and girls without GID. Significant differences in sex-typicality ratings were found between the two groups of boys. Boys with GID elicited ratings intermediate to those for boys and girls without GID. However, the differences between boys with and without GID were much smaller than those in Munson *et al.*, suggesting that the sex distribution in the stimulus set can affect ratings of the sex typicality of children's voices.

4aSC29. Children's ability to benefit from fundamental frequency and vocal tract length differences during speech-in-speech recognition. Mary M. Flaherty, Lori Leibold (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68106, maryflah@buffalo.edu), and Emily Buss (UNC Chapel Hill, Chapel Hill, NC)

The present study evaluated whether children's speech-in-speech recognition benefits from differences in fundamental frequency (F0) or vocal tract length (VTL) between the target and masker talkers' voices. Children, like adults, can benefit from a sex mismatch between competing talkers, but the relative contribution of individual voice characteristics to children's improved speech-in-speech understanding is unknown. In this study, we first tested children's ability to use differences in either F0 or VTL between target and masker speech to evaluate the independent influence of these cues on speech-in-speech recognition. Then F0 and VTL differences were combined to determine whether cue redundancy would reduce the child/adult differences observed in these contexts. Sentence recognition thresholds were measured in a two-talker speech masker. All stimuli were recorded by the same female talker. F0 and VTL of the target sentences were manipulated using the pitch-synchronous overlap-add method. Preliminary results suggest a prolonged developmental trajectory in the ability to use F0 or VTL in isolation, but indicate that the combination of these cues benefits children at an earlier age compared to when either cue is presented in isolation. Adults showed the greatest benefit from the F0-only manipulation, showing no additional benefit when F0 and VTL were combined.

4aSC30. Effects of speech competitors on encoding features of novel word-object pairs. Katherine M. Simeon and Tina M. Grieco-Calub (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Frances Searle Bldg. Rm 2-381, Evanston, IL 60208, ksimeon@u.northwestern.edu)

Fast-mapping is the ability to map novel word-object pairs with very few exposures (Carey & Bartlett, 1978), and is thought to underlie language acquisition. However, language learning often happens in natural environments that contain competing sounds. Previous studies examining whether competing sounds affect fast-mapping yielded inconsistent results, likely due to methodological differences. The present study adds to this body of work by testing the effect of competing sounds on the depth of encoding novel word-object pairs during a fast-mapping task. Three-to-four-year-old children performed a fast-mapping task in quiet and in the presence of a two-talker speech competitor presented at a +2 dB signal-to-noise ratio. In each condition, children were trained on three novel word-object pairs, whereby the object was verbally labeled and associated with related attributes while moving across a computer screen (e.g., "This *modi* can *bounce*"). Children were then tested on their recognition of each word-object pair. Children associated words to objects equally well in quiet and speech competitor conditions, but recalled fewer object attributes when speech competitors were present. Results suggest that speech competitors disrupt encoding of the semantic features of word-object pairs. This presentation will highlight how different methodologies demonstrate varying effects of competing sounds on fast-mapping.

Session 4aUW

Underwater Acoustics, Signal Processing in Acoustics, and Physical Acoustics: High Performance Computing Applications to Underwater Acoustics

Ying-Tsong Lin, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Megan S. Ballard, Cochair
Applied Research Laboratories at the University of Texas at Austin, P.O. Box 8029, Austin, TX 78758

Chair's Introduction—8:30

Invited Papers

8:35

4aUW1. High performance computing methods for nonlinear Bayesian uncertainty quantification. Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, jan.dettmer@ucalgary.ca), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Charles W. Holland (Appl. Res. Lab., Penn State Univ., State College, PA)

Bayesian uncertainty quantification (UQ) for nonlinear inverse problems requires the application of numerical integration (sampling) to estimate the posterior probability density since no closed-form expressions exist. Nonlinear problems are common in geophysics, in particular, in applications of studying the seabed with sound. The computational cost of applying UQ can be daunting and we present several approaches that improve efficiency for large inverse problems. Fundamentally, Bayesian sampling includes both fine-grained parallelism in the forward model (data prediction) as well as coarse grained parallelism in the sampling. Fine grained parallelism can be addressed efficiently by implementation on massively parallel accelerators, such as graphics processing units, and application to reflection coefficient computation is shown. The coarse grained parallelism of sampling is ideally addressed by traditional parallelization with message passing across a cluster of computers. We consider implementation of parallel sampling algorithms that scale efficiently to 10^3 computer cores. Finally, computational efficiency is closely tied to parametrization efficiency, which is addressed by self-adapting parametrizations of unknown complexity that lead to parsimonious representations of complex environments with few parameters. These three aspects of efficiency will be illustrated for several inverse and imaging problems in seabed acoustics and seismology. [Funded by the Natural Sciences and Engineering Research Council of Canada.]

8:55

4aUW2. Massively parallel structural acoustics in Sierra-SD. Gregory Bunting, Scott T. Miller, and Timothy F. Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., 14211 Vista CT NE, Albuquerque, NM 87108, gbuntin@sandia.gov)

Sierra-SD is a massively parallel finite element application for structural dynamics and acoustics. Problems on the order of 2 billion degrees of freedom and running on as up to one hundred thousand distributed memory cores have been solved. Sierra-SD offers a wide range of modern acoustic capabilities. Unbounded problems can be solved with either an absorbing boundary condition, infinite elements, or perfectly matched layers. Problems can be solved in the time domain, frequency domain, or as a linear or quadratic eigenvalue problem. Structural acoustics problems can be solved with monolithic strong coupling, or a Multiple Program Multiple Data (MPMD) coupling with the linear and nonlinear structural Sierra applications. One way handoff is available from other applications in the Sandia National Laboratories' Sierra Mechanics suite, enabling loading through the Lighthill tensor from incompressible fluid codes to calculate far field noise. Real world applications include ship shock loading and vibration of reentry bodies. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC., a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-NA-0003525.]

9:15

4aUW3. Some examples of the application of high-performance computing for time-domain full-wave simulations in underwater acoustics using a spectral-element method. Paul Cristini, Dimitri Komatitsch, and Alexis Bottero (CNRS-LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr)

Being able to perform accurate time-domain full-wave simulations in underwater acoustics is important within the context of complex environments. Among the numerical methods that can be used to perform such types of simulations, the spectral-element method is a method of choice because it can take advantage of the possibilities offered by high-performance computing. This capability is very important because accurate numerical simulations require a high computational power. In this presentation we will show some examples of the use a spectral-element method for underwater acoustics applications. 2D simulations as well as 3D simulations will be considered for various configurations, illustrating the interest of performing such simulations. Shallow water propagation, deep water propagation as well as diffraction by objects in complex environments will be analyzed.

9:35

4aUW4. Finite element modeling for ocean acoustics applications using high performance computing. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Finite element models provide solutions to the Helmholtz equation for ocean acoustics applications, which are exact on the order of the discretization. However, to achieve convergence the discretization must be on the order of several elements per acoustic wavelength. For large-scale ocean acoustics applications, and small (high-frequency) acoustic wavelengths, the number of elements required often exceeds millions. In addition, in each element, the basis set decomposition required for finite element modeling increases the number of degrees of freedom, and for problems involving elastic structures, additional degrees of freedom are required to describe the elastodynamic acoustics equation. Therefore, for a large-scale ocean acoustics problem, the number of degrees of freedom can reach into the tens of millions. This requires large memory caches on computing systems. The number of degrees of freedom can be reduced by using wavenumber decomposition techniques at the expense of running many models. In this talk, large-scale ocean acoustics finite element modeling will be reviewed from an historical perspective, the current state of modeling will be discussed from the in light of available computing resources and the future of finite element modeling will be considered. [Work supported by ONR, Ocean Acoustics.]

Contributed Papers

9:55

4aUW5. Numerical modeling with high performance computing of seismic waves for complex marine environments: Benchmarking with laboratory experiments. Bence Solymosi, Nathalie Favretto-cristini, Vadim Monteiller, Paul Cristini (CNRS-LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr), Borge Arntsen (NTNU, Trondheim, Norway), Dimitri Komatitsch (CNRS-LMA, Marseille, France), and Bjorn Ursin (NTNU, Trondheim, Norway)

Numerical simulations are widely used for forward and inverse problems in seismic exploration to investigate different wave propagation phenomena. However, the numerical results are hard to be compared to real seismic measurements as the subsurface is never exactly known. Using laboratory measurements for small-scale physical models can provide a valuable link between purely numerical and real seismic datasets. We present a case study for comparing ultrasonic data for a complex model with spectral-element and finite-difference synthetic results. The small-scale model was immersed in a water tank. Reflection data was recorded with piezoelectric transducers using a conventional pulse-echo technique. We paid special attention to the implementation of the real source signal—and radiation pattern—in the numerical domain. It involved a laboratory calibration measurement, followed by an inversion process. The model geometry was implemented using a non-structured mesh for the spectral-element simulations. The comparisons show a very good fit between synthetic and laboratory traces in general, and the small discrepancies can be assigned mostly to the noise present in the laboratory data.

10:10–10:25 Break

10:25

4aUW6. Integrated ocean acoustics and dynamics modeling with computer clusters. Ying-Tsong Lin, Timothy F. Duda, Weifeng G. Zhang, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation in the ocean can be influenced by a variety of physical oceanographic processes, including ocean currents, eddies, internal gravity waves, tides, fronts, instabilities, etc. Our group at the Woods Hole Oceanographic Institution has been developing several numerical simulation frameworks to integrate ocean acoustic and dynamical models and implement them on computer clusters. Most of the current dynamical ocean models run on clusters, and we will review our integration approaches in this talk and also point out the requirement for nesting model domains and interpolations. Different parallelization schemes will be introduced to speed up a variety of acoustic numerical models, for example, ray-tracing, normal modes, couple modes and Parabolic-Equation approximation models. Examples of 3D sound propagation in a number of ocean environments (canyons, slopes, continental shelves and basins) with different dominating ocean processes will be demonstrated. Model performance improvement and limitation will be discussed. [Work sponsored by the ONR.]

10:40

4aUW7. Broadband numerical simulations of scattering from rough surfaces. Derek R. Olson (Oceanogr., Naval Postgrad. School, 201 Appl. Sci. Bldg, University Park, PA 16802, olson.derek.r@gmail.com)

Numerical simulation of rough surface scattering via discretization of the Helmholtz-Kirchhoff integral equation (HKIE) is a powerful, yet computationally expensive method to study rough surface scattering. In this work, simulations are performed using Fourier synthesis of the numerical solution of the HKIE using the boundary element method (BEM). Each frequency component is solved using BEM employing a good approximation of a plane wave as an incident field. The surface length requirements on this incident field depend on angle, and all simulations are performed using plane waves whose angular width is less than one degree, and whose mean grazing angle is greater than 20 degrees. This requirement resulted in problems with over 22,000 degrees of freedom and approximately 3000 individual frequencies. This broadband technique is employed to study the dependence of the scattered intensity on surface parameters as well as the bandwidth of the transmitted pulse. Simulations are performed both on surfaces with truncated power law spectra, as well as rippled surfaces with power-law roughness superimposed.

10:55

4aUW8. Scattered field formulation applied to 3D models for buried targets. Aaron M. Gunderson, Anthony L. Bonomo, and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 99164-2814, aaron.gunderson01@gmail.com)

Calculation of the acoustic scattering from buried targets in a seafloor environment is a task ideally suited for finite element software. A complication arises though when the target's environment is modeled as two infinite half spaces of water and sediment, each surrounded by perfectly matched layers (PML's) to satisfy the Sommerfeld radiation condition. Each half space requires a surrounding PML tailored to its material properties, resulting in abrupt discontinuities in PML structure at the corners where the sand-water interface meets the edges of the physical domain. Acoustic energy incident on these corners results in nonphysical reflections directed back into the physical domain. It has previously been shown in a 2D axisymmetric geometry [Zampolli *et al.*, JASA **122**, 2007] that using a scattered field formulation, where the incident, reflected, and transmitted fields are applied as background fields, bypasses the discontinuity and eliminates these spurious reflections. Here, the theory is extended to full 3D geometries, allowing the scattering from 3D buried objects to be evaluated and tested against experimental and analytical benchmark results. Once complete, the 3D model template is easily modified to consider any target at any level of burial. [Work supported by Applied Research Laboratories IR&D and ONR, Ocean Acoustics.]

11:10

4aUW9. Sound radiation from a cylinder in shallow water by FEM and PE method. Zhiwen Qian, Dejiang Shang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University UnderWater Acoust. Fl., Harbin, Hei Longjiang 150001, China, 15846595689@163.com), and Yuanan He (Collage of underwater Acoust. Eng., Harbin Eng. Univ., Beijing, China)

There is no reliable research method for the acoustic radiation prediction of elastic structures in shallow water at present. Based on the finite element method coupled with the parabolic equation, the theoretical model for structure acoustic radiation in shallow water in low frequency is established in this paper. The vibro-acoustic characteristics of elastic cylinder influenced by with upper and lower fluid boundaries is calculated and analyzed. The study shows that when the cylindrical shell approaches the sea surface or bottom, the coupled frequency is higher or lower respectively than that of the shell immersed in the free field. When the diving depth reaches a certain distance range, the coupled frequency tends to be the same as that in free field. The acoustic field radiated from an elastic shell in Pekeris waveguide is similar to that from a point source in low frequency, but significant differences exist between them in high frequency. The structural source can't be equivalent to a point source. The research results also show that the method proposed in this paper has good adaptability to the type of structure geometry and shallow water waveguide, and the calculation efficient and accurate are also very high.

11:25–11:40 Panel Discussion

THURSDAY AFTERNOON, 10 MAY 2018 — GREENWAY F/G, 1:30 P.M. TO 3:15 P.M.

Session 4pBAa

Biomedical Acoustics: Therapeutic Ultrasound and Bioeffects

Emad S. Ebbini, Chair

Electrical and Computer Engineering, University of Minnesota, 200 Union St. SE, Minneapolis, MN 55455

Contributed Papers

1:30

4pBAa1. *In vivo* ultrasound thermal ablation controlled using echo decorrelation imaging. Mohamed A. Abbass, Neeraja Mahalingam, Krishna S. Krothapalli (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, abbassma@mail.uc.edu), Syed A. Ahmad (Surgery, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

The utility of echo decorrelation imaging feedback for real-time control of *in vivo* ultrasound thermal ablation was assessed in rabbit liver with VX2 tumor. High-intensity focused ultrasound (HIFU) and unfocused (bulk) ablation was performed using 5 MHz linear image-ablate arrays. Treatments comprised up to nine lower-power sonications, followed by up to nine higher-power sonications, ceasing when the average cumulative echo decorrelation within a control region of interest exceeded a predefined threshold (-2.3, $\log_{10}$-scaled echo decorrelation per millisecond, corresponding to 90% specificity for tumor ablation prediction in previous *in vivo* experiments [Fosnight *et al.*, Ultrasound Med. Biol. 43:176-186, 2017]). This threshold was exceeded in all cases for both HIFU ($N = 13$) and bulk ($N = 8$) ablation. Controlled HIFU trials achieved significantly smaller average lesion width compared to previous uncontrolled trials. Both controlled HIFU and bulk ablation trials required significantly less treatment time than uncontrolled trials. Prediction of local liver and VX2 tumor ablation using echo decorrelation was tested using receiver operator characteristic curve analysis, showing prediction capability comparing favorably with uncontrolled trials. These results indicate that control using echo decorrelation imaging may reduce treatment duration and increase treatment reliability for *in vivo* thermal ablation.

1:45

4pBAa2. Patient-specific large-volume hyperthermia in the liver for ultrasound-enhanced drug delivery from thermosensitive carriers. Brian Chu, Robin Cleveland, and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, St. Cross College, St. Giles, Oxford, Oxfordshire OX1 3LZ, United Kingdom, brianchu100@googlemail.com)

Ultrasound-mediated mild hyperthermia in the range of 39C to 42C is a promising technique for noninvasive targeted release of thermally sensitive carriers in cancer therapy. The need to sustain a temperature rise of a few degrees evenly throughout a large volume currently limits this therapy to regions with a large accessible acoustic window. Furthermore, the need to

continuously scan clinically available highly focused therapeutic ultrasound transducers typically used for ablation limits the volume that can be maintained at this temperature for tens of minutes, and greatly extends treatment time per unit volume. In addition, bone structures, such as the ribs, significantly restrict accessible target locations in areas such as the liver and play a significant role in preventing this therapy from entering into clinical practice. In order to overcome these limitations, we present a combined 3D full wave finite element modelling and experimental approach to design a sectored lens that can be placed on focused transducers to enable direct mild heating of a larger volume in the liver while accounting for patient-specific aberration in the prefocal path. [Work supported by the RCUK Digital Economy Programme, grant number EP/G036861/1 (Oxford Centre for Doctoral Training in Healthcare Innovation).]

2:00

4pBAa3. The effects of high intensity focused ultrasound on biofilms formed by *Pseudomonas aeruginosa*. Lakshmi D. Bharatula (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore), Enrico Marsili, Scott Rice (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)

Bacterial infections are increasingly difficult to treat due to their growing resistance to antibiotics. Most of these bacterial infections form a biofilm that limits the effectiveness of the antibiotic. Biofilms are microbial cells that are protected by a self-generated matrix of extracellular polymeric substances. In addition to their intrinsic antibiotic resistance, these biofilms are able to respond to the stresses from the antibiotic by inducing drug resistance mechanisms. Currently, the strategy to combat drug resistance is to develop novel drugs, however, the rate of drug development is being surpassed by the rate of drug resistance. There is therefore a need for alternative means in enhancing the efficacy of current drug therapeutics. We propose to use of high intensity focused ultrasound (HIFU) to disrupt the biofilm and promote drug penetration. However, the effects of HIFU on these bacterial communities remain unknown. Here we report on microstructural changes within biofilms formed by *Pseudomonas aeruginosa* due to exposure to HIFU at 500 kHz center frequency. Changes to the biofilm were nondestructively measured through impedance spectroscopy and confocal microscopy. Biofilms were shown to induce cavitation (as measured by a passive cavitation detector) at relatively low pressure amplitudes suggesting the presence of cavitation nuclei within the extracellular matrix.

2:15

4pBAa4. Clot stiffness is inversely correlated with rt-PA thrombolytic efficacy *in vitro*. Karla P. Mercado-Shekhar (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, Cardiovascular Ctr. 3944, Cincinnati, OH 45267-0586, karlapatricia.mercado@uc.edu), Robert Kleven (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Hermes Aponte Rivera, Ryden Lewis, Kunal B. Karani (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Hendrik J. Vos (Biomedical Eng., Erasmus Medical Ctr., Rotterdam, Netherlands), Todd A. Abruzzo (Neurosurgery, Univ. of Cincinnati, Cincinnati, OH), Kevin J. Haworth, and Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Predicting thrombolytic susceptibility to recombinant tissue plasminogen activator (rt-PA) *a priori* could help guide clinical decision-making during acute ischemic stroke treatment and avoid adverse off-target lytic effects. The composition and structure of clots impact their mechanical properties and rt-PA thrombolytic efficacy. The goal of this study was to determine the relationship between clot elasticity and rt-PA thrombolytic efficacy *in vitro*. Human and porcine retracted and unretracted clots were fabricated in glass pipettes. Clots were embedded in agar phantoms, and their Young's moduli were estimated using single-track-location shear wave elasticity imaging. The rt-PA thrombolytic efficacy was evaluated *in vitro* using the percent clot mass loss. The Young's moduli of unretracted porcine and human clots (1.68 ± 0.18 kPa and 0.72 ± 0.13 kPa, respectively) were significantly lower ($p<0.05$) than those of retracted porcine and human clots (4.96 ± 1.07 kPa and 3.38 ± 1.82 kPa, respectively). The percent mass loss of unretracted porcine and human clots ($28.9\pm6.1\%$ and $45.2\pm7.1\%$, respectively) were significantly higher ($p<0.05$) than those of retracted porcine and human clots ($10.9\pm2.1\%$ and $25.5\pm10.0\%$, respectively). The results revealed a linear inverse correlation between the Young's moduli and percent clot mass loss ($R^2=0.95$, $p=0.025$), suggesting that clot stiffness may serve as a surrogate metric for rt-PA thrombolytic susceptibility.

2:30

4pBAa5. *In situ* hydrogel formation for biomedical applications using acoustic cavitation from high intensity focused ultrasound. Umesh Jonnalagadda (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore), Feifei Li (Div. of Chemistry and Biological Chemistry, Nanyang Technolog. Univ., Singapore, Singapore), Jim Lee (Inst. of Chemical and Eng. Sci., Agency for Sci., Technol. and Res., Singapore, Singapore), Atsushi Goto (Div. of Chemistry and Biological Chemistry, Nanyang Technolog. Univ., Singapore, Singapore), Minh Nguyen (Inst. of Chemical and Eng. Sci., Agency for Sci., Technol. and Res., 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)

There is a growing interest in polymer mechanochemistry for their industrial applications. For example, stress-induced crosslinking gel formation from polymer networks is a rapidly growing field of study. Recent work utilizes a variety of different polymer structures and crosslinking mechanisms. However, these polymers are typically soluble in only organic solvents and require the use of a sonicating probe or bath at frequencies below 100 kHz. These requirements limit their use in biomedical applications that require *in situ* gel formation within a patient (e.g., blocking of varicose veins, internal wound healing, etc.). Here we report on the development of a water soluble block copolymer that forms a hydrogel in the presence of acoustic cavitation from high intensity focused ultrasound. These block copolymers are comprised of hydrophilic polyethylene glycol methyl methacrylate units and hydrophobic tridentate crosslinkers. The tridentate crosslinker forms bonds with free metal ions in solution only in the presence of acoustic cavitation induced mechanical stress. We show that the block copolymer is capable of forming a hydrogel in under 90 seconds and will also block a liquid channel formed in an agarose cylinder.

2:45

4pBAa6. Blood coagulation monitoring using acoustic levitation. Vahideh Ansari (Mech. Eng., Boston Univ., 110 Cummington Mall, Rm 414, Boston, MA 02215, vansari@bu.edu), Carol Brugnara (Dept. of Lab. Medicine, Boston Children's Hospital, Boston, MA), and R. G. Holt (Mech. Eng., Boston Univ., Boston, MA)

Impaired blood coagulation can result from a variety of conditions including severe trauma, illness or surgery, and can cause life-threatening bleeding or thrombotic disorders. As a result, instruments which measure functional changes in blood properties upon activation of the clotting cascade are crucial to assess coagulopathies in various disease states. Thromboelastography (TEG) is the gold standard for whole blood coagulation monitoring. TEG requires the blood to be in contact with the sample holder and sensor, leading to a variety of potential artifacts in the results. Acoustic levitation provides non-contact containment and manipulation. We have levitated microliter drops of blood, and employed a drop eigenmodal oscillation technique in order to infer viscoelastic material properties of blood as it coagulates. By comparing our results with TEG results on the same samples, we are able to illustrate certain advantages of the levitation technique. [Work supported by NSF grant # 1438569.]

3:00

4pBAa7. Disposable device for human plateletpheresis. Yuyang Gu, Chuyi Chen, Zeyu Wang, Po-Hsun Huang (Mech. Eng. and Mater. Sci., Duke Univ., B342 Levine Sci. Res., Duke University, Durham, NC 27708, yg90@duke.edu), Hai Fu (Dept. of Fluid Control and Automation, School of Mechatronics Eng., Harbin Inst. of Technol., Durham, North Carolina), and Tony Jun Huang (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Plateletpheresis is a crucial step for the blood donation and clinical diagnosis. Herein we demonstrate a method to conduct plateletpheresis using

acoustofluidic technology in a disposable device that is fabricated solely by a plastic material with precise simulated modeling using low-cost technic. In this device, a quarter wavelength resonator is utilized in order to serve the top layer as the 'tracking wall.' With this device, over 87% RBC removal and platelet recovery rate are achieved at 20 mL/min flow rate with undiluted whole human blood while the function of the platelets is still retained and better than conventional plateletpheresis method. Conclusively, this disposable device synthesizes the high efficiency, high throughput and high biocompatibility which is suitable as the substitute of the current plateletpheresis equipment.

THURSDAY AFTERNOON, 10 MAY 2018 GREENWAY F/G, 3:30 P.M. TO 5:30 P.M.

Session 4pBAb

Biomedical Acoustics: Imaging

Jeffrey A. Ketterling, Chair
Riverside Research, 156 William St, New York, NY 10038

Contributed Papers

3:30

4pBAb1. Simulation and analysis of three-dimensional echo decorrelation imaging. Michael T. Cox, Mohamed A. Abbass, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH 45267-0586, cox2mt@mail.uc.edu)

A numerical model for pulse-echo ultrasound imaging [Mast, JASA 128:EL99-EL104, 2010] is extended to three-dimensional (3D) imaging of linear, weakly scattering continuum media by matrix array transducers. In this model, beam patterns for steered transmit and receive subapertures, calculated analytically under the Fresnel approximation, are combined with a 3D numerical tissue model to yield beamformed scan lines in a rectilinear or pyramidal configuration. Simulated scan lines are processed to yield both volumetric B-mode images and echo decorrelation images, comprising spatial maps of the normalized decorrelation between sequential pulse-echo image volumes. A method is demonstrated for construction and scan conversion of quantitative 3D echo decorrelation images such that, as for 2D echo decorrelation imaging [Hooi *et al.*, JASA 137:585-597, 2015], the mapped echo decorrelation consistently estimates the local decoherence spectrum of the tissue reflectivity. Simulated 3D echo decorrelation images of random media are shown to accurately estimate known decoherence distributions that simulate heating effects in thermal ablation. Effects of factors including transducer frequency, scan configuration, and simulated lesion size on accuracy of decoherence estimates are tested. Applications to 3D monitoring of radiofrequency and microwave ablation are discussed.

3:45

4pBAb2. Exploiting Ballou's rule for better tissue classification. Johannes Kvam, Bjørn A. J. Angelsen (Dept. of Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Det medisinske fakultet, Institutt for sirkulasjon og bildediagnostikk, Postboks 8905, Trondheim, Sør-Trøndelag 7491, Norway, johannes.kvam@gmail.com), and Sverre Holm (Dept. of Informatics, Univ. in Oslo, Oslo, Norway)

Tissue characterisation using the parameter of nonlinearity B/A has shown promising diagnostic value in medical imaging. Its estimation usually infers an assumption of constant sound speed and density. However, an empirical result known as Ballou's rule shows a significant correlation between the reciprocal sound speed and the parameter of nonlinearity for liquid metals. Since the 60's, this result has been found also to be applicable for soft tissues as well with a high correlation ($r = 0.83$). The correlation between B/A and compressibility, the inverse bulk modulus, is just as strong ($r \geq 0.83$) indicating that the product of the two can be beneficial for two main reasons. First, as the variables are highly correlated the variation in the product is larger than that of B/A by itself. This allows for a higher sensitivity as the separation between tissue types will be larger. Second, as methods for estimating B/A often assume constant sound speed and density, variations in these parameters result in uncertainty in the estimated B/A values. By estimating the product of B/A and compressibility instead, the accuracy of the estimate improves. There are multiple methods for estimating this product including the dual frequency SURF method (Hansen *et al.*, JASA 2010).

4:00

4pBAb3. High-frequency (10-100 MHz) ultrasound instrumented forceps for precision cancer surgery. Timothy E. Doyle (Phys., Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Michael J. Salisbury (Manufacturing Eng., Utah Valley Univ., Orem, UT), Michael J. Bennett (Phys., Utah Valley Univ., Orem, UT), Garrett Wagner (Comput. Eng., Utah Valley Univ., Orem, UT), Caitlin Carter (BioTechnol., Utah Valley Univ., Draper, UT), Tobee P. Gunter, Jessica E. Carlson (Biology, Utah Valley Univ., Orem, UT), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), Johnathon S. Gay (Phys., Utah Valley Univ., Orem, UT), and Cindy B. Matsen (Surgery, Univ. of Utah, Salt Lake City, UT)

A critical issue for surgery of soft tissue cancers is ensuring removal of all malignant tissue while conserving as much unaffected tissue as possible. A minimally invasive, *in vivo* technique for detecting residual cancer in surgical margins or metastatic cancer in lymph nodes would significantly improve the precision and efficacy of cancer surgery. The objective of this project was to develop ultrasound instrumented "smart" forceps to guide surgeons during operations, to provide instant diagnostic information, and to enable more precise and complete resection of malignant tissue. The device was based on Martin forceps, modified to include two ultrasonic sensors (polyvinylidene difluoride) at the tips of the forceps, a tissue thickness sensor (rotary potentiometer), sensor wires, and a spring to control the manual stiffness of the forceps. A high-frequency pulser-receiver was used to excite the transmitting sensor at 50 MHz and amplify through-transmission signals from the receiving sensor. The forceps were tested with agarose phantoms embedded with polyethylene microspheres of different diameters (58-550 μm) to vary attenuation. Testing included standard 50-MHz immersion transducers for comparison. The results showed that the forceps' performance was equal to standard transducers, with comparable sensitivity for attenuation and superior accuracy for wave speed.

4:15

4pBAb4. Gas-generating nanoparticles for molecular ultrasound imaging. In-Cheol Sun and Stanislav Emelianov (ECE and BME, Georgia Inst. of Technol., 777 Atlanti Dr., Atlanta, GA 30332-0250, stas@gatech.edu)

Diagnostic ultrasound imaging often relies on contrast agents. Most common contrast agents are microbubbles that are confined to vascular compartments and molecular targets because micrometer size particles cannot escape through endothelial barriers. Therefore, a desired ultrasound contrast agent should consist of nanometer scale particles that are capable of escaping from vasculature, penetrating into tissue, and then generating sufficient contrast once they reach the target site. We developed a novel contrast agent—plasmonic nanoparticles covered by azide compounds (AzNPs), capable of on-demand laser-induced gas generation via photolysis of azide functional groups. The 50 nm diameter AzNPs were small enough to penetrate the endothelial barrier. Upon laser activation, the AzNPs generated nitrogen gas which served as ultrasound contrast agent. The feasibility of the AzNPs to produce ultrasound contrast was tested within polyethylene tube under laminar flow. Upon laser irradiation, significant ultrasound signal enhancement was observed due to the N2 gas bubbles generated by the photolysis of AzNPs. These and other studies suggest that the AzNPs may enable the ultrasound diagnosis of various diseases that conventional microbubbles cannot detect. Furthermore, these optically absorbing particles can also be used for imaging and therapeutic applications of light and sound such as ultrasound-guided photoacoustic imaging.

4:30

4pBAb5. Plane-wave vector-flow imaging of adult mouse heart. Jeffrey A. Ketterling (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St, New York, NY 10038, jketterling@riversideresearch.org), Akshay Shekhar, Orlando Aristizabal (Skirball Inst. of Biomolecular Medicine and the Dept. of Radiology, New York Univ. School of Medicine, New York, NY), Anthony Podkowa (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), Billy Y. Yiu, and Alfred Yu (Dept. of Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

High-frequency ultrasound Doppler modes have been used extensively for murine cardiovascular (CV) studies, but traditional linear-array imaging modes are limited in terms of spatial and temporal resolution. Plane-wave imaging methods allow for high-speed vector-flow information to be obtained throughout a full image frame. Plane-wave imaging has been demonstrated in human CV studies, but its use in mouse models has received minimal attention. A Verasonics Vantage with an 18-MHz linear array was used to acquire plane-wave data at a frame rate of 30 kHz from the left ventricle of adult mice. Batches of 3 transmissions spanning $\pm$ 5 degrees were sent out. The mouse was placed supine on a heated imaging platform and then 2D + time data sequences. The data were beamformed using standard delay-and-sum methods and vector-flow estimates were obtained at each pixel location using a least-squares, multi-angle Doppler analysis approach. Vortex patterns in the left ventricle were visualized over several heart cycles showing the complex flow in the mouse heart.

4:45

4pBAb6. Using mean frequency estimation algorithms for unambiguous identification and visualization of an acoustically active catheter. Viksit Kumar, Bae Hyeong Kim (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine and Sci., 200 First St. SW, Rochester, MN 55905, kumar.viksit@mayo.edu), Azra Alizad (Radiology, Mayo Clinic College of Medicine and Sci., Rochester, MN), Marek Belohlavek (Cardiology, Mayo Clinic College of Medicine and Sci., Scottsdale, AZ), and Mostafa Fatemi (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine and Sci., Rochester, MN)

Minimally invasive procedures using ultrasound are becoming prominent, especially in applications like biopsy, catheter guidance, and surgeries. In our previous work a catheter tip is fitted with a miniature omnidirectional piezoelectric crystal; the ultrasonic field from the imaging array and the piezoelectric crystal interact resulting in a symmetric Doppler shift. Previously, a frequency domain algorithm was used to identify pixels having the symmetric Doppler shift. The identified pixels can then be visualized on B-mode and color map with a unique color thus representing the location of the catheter tip. However, frequency domain algorithms are computationally expensive and require changes to data workflow. Using mean frequency estimators with symmetric Doppler shift results in the catheter tip being displayed as zero mean noise making it hard to distinguish from color Doppler noise. To overcome this limitation, complex demodulation can be used. Modulating the time domain signal with sine and cosine by the piezoelectric excitation frequency and summing them up results in cancellation of one of the symmetric Doppler components. With only one frequency component remaining mean frequency estimators can be used to identify the location of the catheter. The presented algorithm has lower computational complexity and requires minimal change in software design.

5:00

4pBAb7. Detection of lymph node metastasis using photoacoustic imaging and glycol-chitosan-coated gold nanoparticles. Diego Dumani, In-Cheol Sun, and Stanislav Emelianov (School of Elec. and Comput. Eng. and Dept. of Biomedical Eng., Georgia Inst. of Technol. and Emory Univ. School of Medicine, 777 Atlanti Dr., Atlanta, GA 30332-0250, stas@gatech.edu)

Metastases, rather than primary tumors, are responsible for majority of deaths in cancer patients. We developed a non-invasive method to detect sentinel lymph node (SLN) metastasis using combined ultrasound and photoacoustic (US/PA) imaging with glycol-chitosan-coated gold nanoparticles (GC-AuNP). The spatio-temporal distribution of GC-AuNP is affected by presence of metastasis, which is detected by US/PA. GC-AuNPs (100 μl, 0.1 mg Au/ml) were injected peritumorally in breast tumor-bearing mice, and allowed to drain for 24 hours into the SLN. The SLN was located using b-mode ultrasound, and multi-wavelength PA was used to detect GC-AuNPs. Spectroscopic analysis allowed to isolate the signal from GC-AuNPs and quantify their accumulation in the SLN after cellular uptake and transport by immune cells. Results show that distribution of GC-AuNPs in the SLN was affected by presence of metastasis. The overall accumulation was also reduced, with more than 20% signal decrease compared to non-metastatic controls. Histological analysis confirmed that distribution of GC-AuNP-containing immune cells is reduced due to presence of metastatic cells. Our method successfully distinguishes metastatic from non-metastatic lymph nodes using biocompatible nanoparticles and could aid physicians in detection of micrometastasis thus guiding and avoiding unnecessary SLN biopsy.

5:15

4pBAb8. Modeling element directivity in minimum variance beamforming for medical ultrasound. Brian H. Tracey (ECE, Tufts Univ., 196 Boston Ave., Medford, MA 02155, btracey@eecs.tufts.edu), David Lemmerhirt (Sonetics, Inc., Ann Arbor, MI), Dominique Penninck (Clinical Sci., Tufts Cummings School of Veterinary Medicine, Grafton, MA), and Joseph F. Polak (Radiology, Tufts School of Medicine, Boston, MA)

Minimum variance (MV) adaptive beamforming, which adaptively estimates and suppresses interfering signals, has attracted increased attention for ultrasound imaging in recent years. While many MV algorithms have been applied to ultrasound, these algorithms generally rely on signal models that neglect amplitude differences across the array due to element directivity or other factors, introducing mismatch for near-array sources. This paper proposes a beamspace method using randomly sampled subarrays which allows the use of higher-fidelity signal models. Using experimental data, we demonstrate that this algorithm yields noticeable gains in signal contrast by accounting for element directivity.

Session 4pMU

Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Sound Effects and Perception

Jonas Braasch, Chair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Chair's Introduction—1:00

Invited Papers

1:05

4pMU1. Travelling by microphone—The origami of time- and level-based effects. M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com)

The introduction of the microphone brought radical transformations to vocal technique—consider the transition from Bel Canto to crooning. This radically changed the relation between the singer and the backing band or orchestra. More importantly, the singer could now appeal intimately to the listener with all the directness and art of a whispering lover, but with the full backing of the band for sympathy, illustration, and reverie. With the introduction of mutitrack recording and sound reinforcement, artists could bring their listeners closer, even behind their eyes, or hold a space for listeners to relate to themselves, by presenting multiple spaces with multiple actors at multiple distances simultaneously. The simple use of microphones, delays, reverbs (modulated and not), compression, and other effects to say what cannot be said with time and space-bound forms where the relations between musicians, singers and listeners are defined and constrained by the single space they share will be considered. The manifestation of the psychedelic/deep listening experience as a transform, via simple time- and level-based effects, of music into experience grounded in actual, unknowable reality will be considered and auditioned.

1:25

4pMU2. Repurposing the algorithm: How digital signal processing brought on the re-use of reverb. Nikhil Deshpande and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 220 3rd St., Troy, NY 12180, deshpn@rpi.edu)

The earliest digital reverberation algorithms were designed to provide sound with a wider and more immersive sense of physical time-based space. These algorithms successfully replicated the perceptual attributes of realistic rooms, but there was also an interesting and perhaps unforeseen secondary outcome. The ubiquity of digital reverberation made the creative misuse of these algorithms to construct physically impossible spaces readily accessible to musicians. By providing users with access to algorithm controls, reverb grew from a tool for constructing realistic perceptions of rooms into the creation of sound design—and often, the centerpiece of composition. Rather than provide a perceptually accurate presentation of sounds in a stereo field, artists instead began to approach reverb as a smaller part of larger timbral systems. Examples span from reverse reverb to the more complex Eno-based shimmer system, and reverb's place in production and perception will be considered and discussed.

1:45

4pMU3. Audio-haptic perception in immersive improvisational environments. Doug Van Nort (York Univ., 4700 Keele St, Toronto, ON M3J 1P3, Canada, dvnt.sea@gmail.com)

This presentation describes a recent study that explored audio-haptic perception in group performance contexts. Vocalizing and listening activities were structured using methods from the Deep Listening practice developed by Pauline Oliveros, with augmentations to the immersive environment made using a multi-channel audio system, in combination with under-floor and body-worn vibrotactile transducers. Bio-signals from participants were captured and qualitative feedback solicited in order to understand the perceptual and physiological effects of introducing signals to these structured performative activities that are dynamically spatialized in audio as well as haptic sense modalities. The study compares the differences in response when these signals are driven by an external source in comparison to being driven by the bio-signals of participants, including measures focused on convergence and divergence of bio-phyiscal activity. The presentation will compare these scenarios and discuss the larger implications for developing immersive interactive environments that contribute to a sense of connection, sociality and play.

2:05

4pMU4. Guitar amplifier modeling—Perceptual evaluation of audio similarity. Felix Eichas and Udo Zoelzer (Elec. Eng., Helmut Schmidt Univ., Holstenhofweg 85, Hamburg 22043, Germany, udo.zoelzer@hsu-hamburg.de)

Virtual analog modeling of guitar equipment, especially tube-based guitar amplifiers, is a hot topic in sound effect research. However, there are no established standards on the evaluation of the accuracy of such modeling processes. This work presents a first approach on finding a metric for evaluating the similarity between the output of an analog reference amplifier and a digitized version of the same device. A gray-box modeling procedure is used where only assumptions about the general structure of a guitar amplifier are made. The necessary information about the reference device is obtained by input and output measurements. First a digital model is constructed and the error between digital model and analog system is minimized by the Levenberg-Marquardt parameter optimization algorithm. The error is expressed as the result of a cost function. To be able to produce good results, this cost function needs to consider psycho-acoustic aspects like e.g. the perceived frequency resolution. The results of a spectrogram based cost function are compared to the results of a listening test, where the test subjects should rate the similarity of the two audio signals (reference and digital simulation) and the findings are discussed.

2:25

4pMU5. The use of physical and artificial room reverberation to create transparent or fused ensemble sounds. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Room Reverberation has shaped the sound of musical instruments from the beginning. The oldest known musical instrument, a 40,000-year-old bone flute has been found in a cave, the Hohle-Fels Hoehle. Impulse response measurements taken at this cave reveal that the cave had a reverberation time, T_{30}, at mid frequencies of 1.8 seconds, a value within the range of that of a modern concert hall. Room reverberation has continued to play an important role in defining our cultural sonic environments. During the Reformation, for example, there was a clear call for transparency so that the words of the reverends could be understood by the people. This sound ideal also affected the music of protestant composers like Johann Sebastian Bach an ideal that culminated in the interpretations of Glenn Gould, who preferred the natural sound of the dry sound recording studio. While a concert hall is always bound to its sound, most modern recording studios offer a flexible design where the room can be created artificially around the artist to achieve a certain sound ideal. Unlike in a physical hall that embeds all musicians in a homogenous sound, artificial reverb is often used to provide a unique environment for each instrument to meet specific goals.

2:45

4pMU6. Correlation between perception and acoustical parameters in five musical genres. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

To investigate the influence of sound effects on the perception of musical genres, excerpts from 150 musical pieces of Rock, Jazz, Classical, Ethnographic and Electronic music were compared. In a listening test subjects were asked to judge them in terms of 12 adjectives related to spaciousness, timbre and semantic attributes. Additionally the pieces were analyzed with three methods. To judge the spaciousness the Interaural Cross Correlation (IACC) was calculated. To estimate the audio effect strength an echodensity was defined as the amount of phase fluctuations. To estimate the density of tonal texture a fractal correlation dimension was used. The musical genres clearly sorted along their IACC values with Rock being most 'mono' and Classical music most spatial. The echodensity was about the same for all genres except for electronic music where it was considerably larger. The mean fractal correlation dimensions were about four with all genres pointing to a mean textual density for all genres. In terms of perception, the higher echodensity correlated positively with the adjective "artificial" for Electronic music. Additional findings were made.

3:05–3:20 Break

3:20

4pMU7. Sound effects without side effects—Suppression of artifacts in audio signal processing. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St, Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Sound effects processes intending one production outcome can too often be accompanied by unwanted side effects. Signal processing strategies have evolved in popular music production that preserve a producer's sonic goals while suppressing these artifacts. The solutions found can be surprising and frequently have no analog in all-acoustic music creation and presentation. Sound effects with these built-in defense mechanisms become studio-only creations, effects for loudspeaker mediated art only.

3:40

4pMU8. Preference, localization, attention, and the limit of localization distance (LLD). David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Recent work by Lokki *et al.* has found that a perception he calls "proximity" is a major component of preference in concert halls. In this talk we propose that proximity is perceived when sources are close enough or clear enough that they can be sharply localized and understood. In halls and rooms we find this ability abruptly disappears at a critical distance from the source, the Limit of Localization Distance, or LLD. We find that proximity attracts and holds attention, which is a major reason cinemas and drama theaters have dry acoustics and strong direct sound paths. In this talk we will describe the physics behind the perception of proximity, why it has not been previously recognized, how to predict where the LLD will occur from a binaural impulse response, and how to optimize proximity in halls, theaters, and classrooms.

4:00

4pMU9. Effects created by time-variant spectral processing of sounds. James W. Beauchamp (Elec. & Comput. Eng. and School of Music, Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr, Urbana, IL 61801-6824, jwbeauch@illinois.edu)

Various processes can be used to modify a sound to produce special effects. One approach is to first compute a time-frequency representation of a sound signal, follow that by altering the representation in various ways, and finally perform resynthesis to produce modified versions of the signal. Example modifications are (1) duration modification (time-scaling), (2) pitch change, (3) noise content manipulation, (4) spectral envelope modification, and (5) temporal envelope modification. The time-frequency representation can also be used to develop simplified synthesis models that depend on only a few time-varying parameters, such as overall amplitude, pitch, and spectral centroid and shape. In the special case of musical sounds with vibrato, spectral amplitudes and frequencies can be parameterized in terms of time-variant vibrato depth, rate, and mean envelope. These parameters can then be controlled independently of pitch and duration. Also, timbral parameters of instruments can be interchanged or blended to create hybrid instruments with novel timbral qualities. Several of these effects will be described and demonstrated.

4:20

4pMU10. Allpass decorrelating filter design and evaluation. Elliot K. Canfield-Dafilou and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Dr, Stanford, CA 94305, kermit@ccrma.stanford.edu)

By breaking up the phase coherence of a signal broadcast from multiple loudspeakers, it is possible to control the perceived spatial extent and location of a sound source. This so-called signal decorrelation process is commonly achieved using a set of linear filters, and finds applications in audio upmixing, spatialization, and auralization. It is important that any individual decorrelating filter does not perceivably alter the sound of its input, i.e., that it does not color the signal spectrum or smear transients over time. In that way, the process will only affect the spatial properties of the sound. Allpass filters make ideal decorrelation filters since they have unit magnitude spectra, and therefore can be perceptually transparent. By manipulating their phase, allpass filters can be leveraged to achieve a degree of decorrelation. Here, we present a method for designing allpass decorrelation filters by specifying group delay trajectories in a way that allows for control of the amount of correlation as a function of frequency. This design is efficiently implemented as a cascade of biquad allpass filters. We present statistical and perceptual methods for evaluating the amount of decorrelation and audible distortion.

Contributed Papers

4:40

4pMU11. Pitch and timbre discrimination at wave-to-spike transition in the cochlea. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

A new definition of musical pitch is proposed. A Finite-Difference Time Domain (FDTM) model of the cochlea is used to calculate spike trains caused by tone complexes and by a recorded classical guitar tone. All harmonic tone complexes, musical notes, show a narrow-band Interspike Interval (ISI) pattern at the respective fundamental frequency of the tone complex. Still this fundamental frequency is not only present at the bark band holding the respective best frequency of this fundamental frequency, but rather at all bark bands driven by the tone complex partials. This is caused by drop-outs in the basically regular, periodic spike train in the respective bands. These drop-outs are caused by the energy distribution in the wave form, where time spans of low energy are not able to drive spikes. The presence of the fundamental periodicity in all bark bands can be interpreted as pitch. Contrary to pitch, timbre is represented as a wide distribution of different ISIs over bark bands. The definition of pitch is shown to also works with residue pitches. The spike drop-outs in times of low energy of the wave form also cause undertones, integer multiple subdivisions in periodicity, but in no case overtones can appear. This might explain the musical minor scale, which was proposed to be caused by undertones already in 1880 by Hugo Riemann, still until now without knowledge about any physical realization of such undertones.

4:55

4pMU12. The time and place of cochlear implant pitch perception. Susan R. Bissmeyer (Biomedical Eng., Univ. of Southern California, 1042 Downey Way DRB 140, Los Angeles, CA 90089-1111, ssubrahm@usc.edu), Shaikat Hossain, and Raymond L. Goldsworthy (Otolaryngol., Univ. of Southern California, Los Angeles, CA)

It is well established that cochlear implant users can perceive pitch associated with pulse train stimulation rate, but that rate sensitivity diminishes above 300 pps. However, there have been multiple reported cases of individuals sensitive to stimulation rates well above 300 pps. Furthermore, Goldsworthy and Shannon (2014) demonstrated that sensitivity to stimulation rate could be improved through psychophysical training. The goal of the present study is to identify factors affecting rate sensitivity in cochlear implant users, including age, etiology, and duration of deafness, electrode location, stimulation mode, neural health as measured by multipulse integration, and spatial tuning as measured by forward masked thresholds. Adult cochlear implant users serve as subjects in a 3-week protocol that includes 4 hours per week of electrode psychophysics conducted in the laboratory and 30 minute daily auditory listening exercises conducted at home. Results indicate that cochlear implant users with minimal psychophysical training consistently discriminate rate differences at least as high as 800 pps. Emerging trends show a group effect with a monopolar stimulation mode and basal cochlear location benefit. We will report on the effect of the factors affecting rate sensitivity for this ongoing study.

5:10

4pMU13. Examining the auditory cortex response to musical stimuli with differences in timbre and reverberation. Martin S. Lawless and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu)

In a previous functional magnetic resonance imaging (fMRI) study by the authors, the addition of reverberation was shown to degrade the cortical auditory response to musical stimuli. This degradation may occur since reverberation blurs the frequency content of the stimuli in time, making it more difficult for the brain to distinguish between distinct auditory objects. The current study aims to further establish the auditory response to reverberant stimuli by investigating if higher-order frequency content (timbre) affects the auditory cortex's sensitivity to reverberation. Using an anechoic solo-instrumental trumpet motif, a sine-trumpet musical stimulus was generated by removing the harmonic content from the played notes while maintaining the attack of the instrument and rhythm of the musical passage. Auralizations of both motifs were created from eight simulated room conditions with reverberation times ranging from 0.0-7.2 s. Participants were recruited to rate the auralizations in terms of overall preference, perceived reverberance, and perceived clarity. Subsequently, the participants listened to a subset of the solo-instrumental and sine-trumpet auralizations in an MRI machine. Contrasts between the motifs and room acoustic conditions were analyzed to confirm the sensitivity of the auditory cortex to reverberation, as well as investigate the possible interaction effect between reverberation and timbre.

Session 4pNS

Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards: Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices II

Cameron J. Fackler, Cochair
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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

Invited Papers

1:30

4pNS1. Observation on the variability of fitting by two experimenters at REAT and the agreement with objective measurement on three premolded earplugs. Yufei Liu (Person Safety Div., 3M, No.9, Nanxiang Er Rd., Sci. City, Guangzhou, Guangdong 510663, China, sliu9@mmm.com)

Real ear attenuation at threshold (REAT) has been recognized as the gold standard method to evaluate the attenuation of hearing protection devices (HPDs). It has been standardized in ANSI, ISO/EN and AS/NZS documents. The fitting of HPDs is one of the key factors of REAT measurement. The experimenter fit protocol of ANSI S3.19 is considered one precise approach to obtain the "best fit" of HPDs. A study of the attenuation obtained between the three fits for two experimenters on the same ten subjects on three premolded three-flange earplugs was conducted at 3M E-A-RCAL laboratory, including two models of Chinese brands earplugs and 3M™ E-A-R™ UltraFit earplug. Good agreement was shown between the REAT results that were obtained by the two experimenters. The REAT result of one Chinese brand earplug shows much higher variability within subjects. Comparing with the insertion loss (IL) measured by G.R.A.S 45CB, which has been corrected for bone conduction, occlusion effects and physiological noise, produced IL for two of the tested earplugs that did not showed good agreement with REAT results at most, but not frequencies. The discrepancies will be discussed.

1:50

4pNS2. Quantifying the insertion loss of hearing protection using a compressed gas shock tube. Theodore F. Argo (Appl. Res. Assoc., Inc., 7921 Shaffer Parkway, Littleton, CO 80127, targo@ara.com), Nate Greene (Dept. of Otolaryngol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), James Easter (Cochlear Boulder LLC, Boulder, CO), Daniel J. Tollin (Dept. of Otolaryngol. and Dept. of Physiol. and Biophys., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), and Timothy J. Walilko (Appl. Res. Assoc., Inc., Littleton, CO)

Compressed gas shock tubes provide a method of generating repeatable and consistent blast overpressure waves similar to those observed in combat and military training. Shock tubes of different diameters and lengths have been used to evaluate the level dependence of various types of hearing protection devices (HPDs) for sound pressure levels ranging from 132 to 192 dB. Insertion loss data was generated using mechanical test fixtures, post mortem human surrogates, and animal models in order to compare the respective auditory responses at these high pressures. The inclusion of animal models was possible since gas driven shock tubes can be housed in mobile laboratories and trailered to vivarium for physiological and behavioral observations. Measurements were conducted on a variety of HPDs and indicate that the responses of many advanced hearing protection devices are nonlinear at these sound pressure levels. Models of hearing protection and auditory injury have been updated and improved based upon quantification of HPD level-dependence. [Work supported by DOD grant W81XWH-15-2-0002.]

2:10

4pNS3. Occlusion effects of hearing protection device on ear canal transfer function and cochlear injury from impulse noise exposure. Brissi Zagadou (L3 Technologies, 10180 Barnes Canyon Rd, San Diego, CA 92121, Brissi.Zagadou@L3T.com)

The objective of this paper is to present a new finding on the effects of hearing protection devices (HPDs) on the ear canal (EC) transfer function (TF_{EC2ED}). Specifically, we show that when an earmuff is worn, the TF_{EC2ED} is not the same as that for the bare head, leading to a significant difference in cochlear injury prediction. We used shock tube blast with an acoustical test fixture (ATF) fitted with the recovered earmuffs from the historic blast overpressure project (BOP) to measure the TF_{EC2ED}. A microphone was placed at the

EC entrance under the earmuff (undermuff) to measure the pressure at the EC entrance, while the built-in ATF's microphone was used to measure the eardrum (ED) pressure. The TF_{EC2ED} is defined as the ratio of the pressure at the ED to the pressure at the EC entrance. We compared the cochlear integrated energy (ICE) values, when the TF_{EC2ED} obtained from bare head (Case-1) and that measured using the ATF fitted with an earmuff (Case-2), respectively are used in the ICE-model, to estimate the effects of the difference between the TF_{EC2ED}s on injury. The human BOP undermuff pressure data were used as model input. For Case-1, the input was at the EC. The ED pressure, reconstructed using the undermuff pressure and the TF_{EC2ED} measured using the ATF, was used for Case-2. We found significant differences in the ICE-values as the result of the difference between the TF_{EC2ED}s. We conclude that hearing protector models must correctly model the occlusion effects of HPDs by incorporating the change in the TF_{EC2ED}.

2:30

4pNS4. Anticipatory middle ear muscle contractions in damage-risk criteria. Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com), Stephen M. Tasko (SASRAC, Kalamazoo, MI), Kristy Deiters (SASRAC, Portage, MI), William J. Murphy (National Inst. for Occupational Safety and Health, Cincinnati, OH), Heath Jones, William A. Ahroon (U.S. Army Aeromedical Res. Lab., Fort Rucker, AL), and Nate Greene (U.S. Army Aeromedical Res. Lab., Aurora, CO)

Anticipatory middle ear muscle contractions (MEMC) have been implemented as protective components of Damage-Risk Criteria (DRC) for impulsive noises. However, no studies have shown that anticipatory MEMC are pervasive among humans. This presentation describes a series of studies of the viability of assumed anticipatory MEMC obtained either through classical conditioning or while operating a model gun. Participants were adults with normal hearing, and the conditioning tasks varied on sensory modality and attention. Both between- and within-subjects designs were used. A conditioned response was defined as an MEMC occurring prior to the unconditioned stimulus and when only the conditioning stimulus was presented. These studies do not suggest that anticipatory MEMC should be included in DRC for impulsive noises.

Contributed Papers

2:50

4pNS5. Developing a method to assess noise reduction of firearm suppressors for small-caliber weapons. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Gregory Flamme (SASRAC, Forest Grove, OR), Adam R. Campbell, Edward L. Zechmann (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH), Michael Stewart (Dept. Commun. Disord., Central Michigan Univ., Mount Pleasant, MI), Stephen M. Tasko (Dept of Speech, Lang. and Hearing Sci., Western Michigan Univ., Kalamazoo, MI), James E. Lankford (Allied Health and Commun. Disord., Northern Illinois Univ., Dekalb, IL), Deanna K. Meinke, and Donald Finan (Audiol. and Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO)

Firearm suppressors have the potential to reduce the muzzle blast by diffusing the initial shock wave of a gunshot. Currently, the American National Standards Institute does not have any standards that specifically address suppressor measurements. A recent NATO standard, AEP 4875 has been proposed to characterize suppressor performance, but the scope of this standard does not include suppressor effects at the shooter's ear. Additionally, the standard requires firing the weapon from an elevated platform 4 meters above the ground with microphones positioned with regular spacing of about 18 degrees at 5 meters from the muzzle. This study evaluated fourteen different firearms with and without a suppressor. Different loads of ammunition were used to vary the speed of the projectile. For ten of the guns, both supersonic and subsonic conditions were measured. Twelve microphones were positioned at 30-degree spacing in 3-meter ring at 1.5 meters above the ground. One microphone was positioned at 1 meter to the left of the muzzle and two microphones were positioned at 15 centimeters from the right and left ears. The suppressors were effective in reducing the peak sound pressure levels between 3 and 28 dB and 8-hour equivalent energy (LAeq8) between 2 and 24 dB.

3:05

4pNS6. Impulse noise measurements of M16 rifles at Quantico Marine Base. Reese D. Rasband (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU N283 ESC, Provo, UT 84602, r.rasband18@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Kent L. Gee, S. Hales Swift (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Caleb M. Wagner (Human Systems Program Office, Air Force Res. Lab., Dayton, OH), William J. Murphy, and Chucri A. Kardous (Hearing Loss Prevention Team, National Inst. for Occupational Safety and Health, Cincinnati, OH)

This paper describes a study conducted at U.S. Marine Corps Base Quantico to determine firing range impulse noise levels and assess noise exposures. Measurements were performed with M16 rifles at an outdoor firing range using a 113-channel array of 6.35 and 3.18 mm microphones that spanned potential locations for both shooters and instructors. Data were acquired using 24-bit cards at a sampling rate of 204.8 kHz. Single weapon measurements were made with and without an occupied range, with a shooter and with a remotely triggered gun stand. In addition, measurements were made with multiple shooters to simulate exposures for a realistic range environment. Results are shown for the various range configurations as a function of angle and distance. Analyses include waveforms, spectra, and peak levels, as well as the 100 ms A-weighted equivalent levels required by military standard MIL-STD-1474E. [Sponsored by US Office of Naval Research.]

3:20

4pNS7. Rating the perception of jet noise crackle. Paul B. Russavage, Tracianne B. Neilsen, Kent L. Gee, and S. Hales Swift (Dept. of Phys. & Astronomy, BYU, N283 ESC, Provo, UT 84602, pau616@gmail.com)

Crackle is a perceptual aspect of noise caused by impulsive acoustic shocks and observed in supersonic jets, including those from military aircraft and rockets. Overall and long-term spectral noise metrics do not account for the unique perception of crackle. Listening tests were designed to better understand perception of crackle and examine its relationship to physical noise metrics, such as skewness of the first time derivative of the pressure waveform, hereafter derivative skewness. It is hypothesized that as derivative skewness increases, the perception of crackle tends to increase. Two listening tests were conducted with 30 subjects to examine their perception of crackle. In the first test, subjects compared and ordered crackle-containing sounds. In the second test, category scaling was employed with subjects rating the crackle content with category labels: (1) smooth noise with no crackle, (2) rough noise with no crackle, (3) sporadic crackle, (4)

continuous crackle, and (5) intense crackle. There is a high correlation between perception of crackle and derivative skewness. This five-point classification scheme appears to be an effective tool to measure crackle perception. These insights will help inform community noise models, allowing them to incorporate annoyance due to jet crackle. [Data courtesy of F-35 JPO.]

THURSDAY AFTERNOON, 10 MAY 2018 GREENWAY J, 1:40 P.M. TO 4:15 P.M.

Session 4pPA

Physical Acoustics, Noise and ASA Committee on Standards: Sonic Boom II

Alexandra Loubeau, Cochair
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Joel B. Lonzaga, Cochair
Structural Acoustics Branch, NASA Langley Research Center, 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681

Invited Papers

1:40

4pPA1. Sonic boom induced window rattle in indoor environments. Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., MS 463, Hampton, VA 23681, j.klos@nasa.gov)

The perceptual environment inside homes ensonified by sonic booms consists mainly of the transmitted indoor boom sound, rattle from items in loose contact with vibrating structures, and structural vibrations that may be felt or seen. In this presentation, measurements of window rattle from a recent laboratory study conducted inside the Interior Effects Room at NASA Langley Research Center will be summarized. Low amplitude sonic booms were reproduced at the facility exterior, which induced rattle in several distressed windows that were interchanged in the facility wall. The indoor sound field, the combined transmitted boom and window rattle sound, was measured at seven microphones placed inside the test room. The "rattle only" sound was recovered by subtracting the "boom only" sound, which was computed by convolving a measured room impulse response with the exterior excitation waveform, from each microphone measurement. In total, predicted low booms from fourteen different vehicles, each at three loudness levels, ensonified forty-two different window rattle conditions. Each measurement was repeated six times in both a lively and damped room configuration. This database of 148,176 rattle measurements enabled objective comparisons of rattle produced by different aircraft, at different booms levels and for different window rattle conditions.

2:00

4pPA2. Evaluation of the effect of aircraft size on indoor annoyance caused by sonic booms and rattle noise. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

NASA plans to use a low-boom flight demonstration supersonic aircraft for community annoyance studies of sonic booms. As previously reported, laboratory studies were conducted in NASA's Interior Effects Room (IER) to help determine the relevance of using a sub-scale aircraft. Indoor annoyance was evaluated for a variety of sonic booms predicted for several different sizes of vehicles, whose levels were adjusted to the same range of loudness levels. Although no significant effect of aircraft size was found for equivalent loudness levels, a new laboratory study was conducted in the IER to extend this investigation to include the effect of secondary rattle noises. The rattle noise playback was determined from window rattle measurements performed in the IER that resulted in a database of rattle noises matched with sonic booms predicted for different aircraft, at different boom levels. The main objective was to test whether aircraft size is still not significant when realistic window rattles are included in the simulated indoor sound field. The data have also been used to evaluate a variety of noise metrics. Results using metrics computed on exterior boom levels show greater variation in subjective annoyance than in previous studies conducted without rattle.

2:20

4pPA3. Perceptual characterization of Mach cut-off sonic booms. Nicholas D. Ortega, Michelle C. Vigeant, and Victor Sparrow (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, njo5068@psu.edu)

Mach cut-off refers to a set of flight conditions wherein the direct sound of a sonic boom does not reach the ground. This phenomenon occurs in the idealized lower atmosphere, which naturally refracts sound upwards. The current work, subdivided into two perceptual studies, investigates the perception of the evanescent sound field below the direct path of the boom. Both studies used Mach-cutoff ground signature recordings from NASA's "Farfield Investigation of No-boom Thresholds" (FaINT). In the first study, subjects provided

descriptors for Mach-cutoff ground signatures in order to develop a vocabulary useful for describing these sounds. In the second study, subjects rated the stimuli on multiple scales, comprised of descriptors from the first task and annoyance, to identify which perceptual attributes factor into annoyance. These findings were used to obtain an overall perceptual characterization of the Mach-cutoff ground signatures, and these studies may inform metrics for use in predicting reactions to Mach-cutoff flyovers. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

2:40

4pPA4. How ordering effects change with cursor placements on scales when evaluating annoyance levels of transient environmental sounds. Yiyun Zhang and Patricia Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907, zhan1728@purdue.edu)

Sound ordering effects are the impacts of the last heard sound on the rating of the current sound when evaluating a group of signals in sequence. Previously, it was shown that such effects are more significant in evaluations of transient sounds than in evaluations of steady-state sounds. The goal of this study was to investigate whether initial cursor placement on the rating scale could strengthen or attenuate ordering effects. In the previous study, the cursor was placed in the middle for the first signal, and then was left at the place from the last rating. In this follow-up test with the transient sounds, the effect of three different cursor setting strategies was examined: (1) always at the left (indicating the least annoyance level), (2) at the place of the last rating (same as the previous test), and (3) randomly placed on the rating scale. Preliminary results show that the overall trends of sound annoyance ratings with the three cursor setting strategies are similar for most subjects, and results are similar to those in the previous test. However, for some subjects, left cursor placement tends to reduce those subjects' annoyance ratings.

3:00–3:15 Break

3:15

4pPA5. A wide-area system for tracking supersonic aircraft overflights and capturing their shock waves. Gonzalo Sanchez (Sanchez Industrial Design Inc., 4319 Twin Valley Rd., Ste. 3, Middleton, WI 53562, gsanchez@sid-inc.com), Richard Horonjeff (None, Boxborough, MA), Vincent Mestre (Landrum & Brown, Irvine, CA), and Sanford Fidell (Fidell Assoc. Inc, Woodland Hills, CA)

An Internet-based, geographically-distributed prototype system for capturing sonic booms produced by NASA's low boom flight demonstrator (LBFD) aircraft has been successfully field tested. The system is designed to track supersonic LBFD flights over distances of hundreds of miles so that interviewing of overflown populations can be synchronized with the time of arrival of their sonic booms in multiple, widely separated communities. A central server archives reports of reception of the demonstrator aircraft's ADS-B signals from multiple field receivers, calculates shock wave arrival times at interviewing sites, captures and stores the acoustic waveform produced at the predicted time of arrival of the boom, coordinates interview start and stop time with sonic boom arrival times, uploads the captured waveforms on demand, and provides multiple, geographically-distributed analysts with password protected, remote access to the flight tracking and acoustic information in near-real time.

3:35

4pPA6. Multilevel analysis of recent noise social survey data including noise sensitivity. Jonathan Rathsam (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov), Maurice E. Hayward (None, Newport News, VA), Laure-Anne Gille (None, Vaulx-en-Velin, France), Edward T. Nykaza (ERDC-CERL, Champaign, IL), and Nicole M. Wayant (ERDC-GRL, Alexandria, VA)

Noise social surveys are typified by large variation in annoyance ratings for equivalent noise exposure. Some of this variation has previously been linked to individual attitudes, including noise sensitivity. A framework provided by multilevel statistical analysis can be used to study the relationships between noise annoyance and individual attitudes. The current effort applies multilevel analysis to two recent community surveys of impulsive noise for which noise-sensitivity was assessed via a baseline survey. In the results, the sizable annoyance variation among respondents is quantified. Multilevel models with random effect terms for slope and or intercept describe more variation than multiple regression models with only fixed effects for noise exposure and noise sensitivity.

3:55

4pPA7. Determining a sufficient range of sound levels for single event analysis in quiet sonic boom community surveys. Jasme Lee (Dept. of Statistics, North Carolina State Univ., NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jasme.lee@nasa.gov), Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA), and Alyson G. Wilson (Dept. of Statistics, North Carolina State Univ., Raleigh, NC)

In order to enable quiet supersonic flight overland, NASA is establishing a quiet sonic boom database to present to national and international noise regulators. The database will contain dose-response models to provide evidence of a cause-and-effect relationship between boom sound levels and annoyance. In order to build the dose-response models, NASA is designing surveys for data collection. An important aspect in planning for future community surveys is the range of sound levels necessary for single event analysis. The dose-response models must be characterized at sound levels both above and below a notional threshold. The trade-offs for testing at high sound levels are the increased risk of negative community reaction and misconception of quiet sonic booms. On the other hand, testing only at low sound levels will result in unstable estimates in the dose-response models. To find a compromise between the two, analysis of a pilot community survey from 2011 explores how excluding high level booms will affect the dose-response analysis since both low and high level booms were present in the study. The data analysis will also provide insight into the appropriate range of sound levels for single event analysis in future community surveys.

Session 4pPPa

Psychological and Physiological Acoustics: Sound Localization, Spatial Release from Masking, and Binaural Hearing with Devices

Nirmal Kumar Srinivasan, Chair
Audiology, Speech-Language Pathology, and Deaf Studies, Towson University, 8000 York Road, Towson, MD 21252

Contributed Papers

1:00

4pPPa1. Improvements in transaural synthesis with the Moore-Penrose pseudoinverse matrix. Aimee Shore and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., 567 Wilson Rd, East Lansing, MI 48824, shoreaim@msu.edu)

Transaural synthesis using loudspeakers is a powerful technique for conducting binaural hearing experiments while avoiding myriad problems caused by headphones. It is the most promising method to test binaural hearing for listeners using cochlear implants. Transaural synthesis is typically implemented using crosstalk cancellation, a technique whereby the left-channel signal and right-channel signal are adjusted so that reproduction by two synthesis loudspeakers results in near perfect transmission of the target left and right signals to the left and right ears (or processor microphones). Crosstalk cancellation has enabled precise experimental control over the stimulus delivered to each ear. Unfortunately, the 2×2 matrix inversion required for crosstalk cancellation can occasionally introduce spuriously large amplitudes for specific frequencies into the adjusted signals. These large amplitudes lead to perceptually salient tones. We demonstrate through simulation and experiment in a real room that adding a third loudspeaker and solving the resulting 2×3 problem using the Moore-Penrose pseudoinverse matrix yields dramatically fewer large amplitudes in the resulting adjusted waveforms. We have also conducted experiments to investigate the 2×3 system's robustness to inadvertent listener motions.

1:15

4pPPa2. Threshold interaural time differences under optimal conditions. Mathias Dietz and Sinthiya Thavam (National Ctr. for Audiol., Western Univ., 1201 Western Rd., London, ON N6G 1H1, Canada, mdietz@uwo.ca)

Klumpp and Eady (1956, J Acoust Soc Am 28, p.859-860) reported preliminary data on human sensitivity to interaural time differences (ITD) with various stimuli. At 10 μs ITD the best discrimination of 79% correct was reported for band-pass filtered (150-1700 Hz) noise. Despite the preliminary nature, and presentation methods different from todays, the above is still the best available reference for optimal ITD discrimination. The goal of the current study is to systematically determine the stimulus and the experimental paradigm that results in the smallest threshold ITD and to provide an accurate reference value. We varied seven stimulus and procedure parameters: stimulus waveform, stimulation level, stimulus duration, adaptive versus constant stimulus procedure, alternative-forced-choice (AFC) procedure, inter-stimulus pause duration, and complete waveform versus ongoing ITD. The condition yielding the lowest threshold ITD was Gaussian noise band-pass filtered from 20 to 1400 Hz, presented at 70 dB SPL, with a short inter-stimulus pause of 50 ms, and an interval duration of 0.5 s. Averaged across 8 trained subjects, the threshold ITD for this condition at the 79% correct level was 7 μs. The influence of each parameter will be discussed together with the obstacles of accurately determining this value.

1:30

4pPPa3. Importance of low frequency hearing for spatial release from masking. Nirmal Kumar Srinivasan, Maxwell Schmidt, Alexis Staudenmeier, and Kelli Clark (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu)

Listeners tend to achieve better speech understanding when the masking talker(s) are spatially separated from the target talker; an effect known as spatial release from masking. Here, we present data on spatial release from masking experiment where the target and maskers are high pass filtered with a three frequency audiometric pure-tone threshold equal to 20 dB HL. Young normal hearing listeners were presented with CRM sentences in anechoic and reverberant listening environments. The target speech was presented simultaneously with two maskers that were either colocated (0° azimuth) or symmetrically separated from the target in azimuth (±15°). Reverberant listening environments were simulated using techniques described in Zahorik (2009) and reverberation times (T60) of 1s and 2s were used. Preliminary results indicate that the amount of release from masking reduced for filtered speech and this reduction was greater for reverberant listening environments as compared to anechoic environment.

1:45

4pPPa4. Is there spatial release from listening effort in noise and reverberation? Jan Rennies and Gerald Kidd (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, jrennies@bu.edu)

Spatial unmasking of speech in the presence of interfering sounds has been investigated mostly by assessing speech recognition performance (e.g, SRTs or percent correct). Typically, these experiments measure recognition under very difficult listening conditions in which optimal performance (i.e., obtained in quiet) is reduced. For normal-hearing listeners this usually means presenting the signal at low/negative SNRs to avoid ceiling effects. However, such conditions may not be representative of everyday listening and hence the generalization of the results to real-world environments is limited. However, several recent studies have concluded that *listening effort* may be more meaningful than intelligibility because it can vary across a wider range of SNRs including positive SNRs where intelligibility is at ceiling. Despite this advantage, relatively few studies have examined listening effort in realistic listening conditions. In particular, binaural effects in listening effort have not been systematically investigated. The goal of this study is to measure listening effort in normal-hearing listeners in conditions with systematically varying binaural parameters, involving one or more interferers and different degrees of reverberation. Listening effort is assessed using categorical listening effort scaling, and experimental data are compared to predictions of a binaural listening effort model derived from a binaural speech intelligibility model.

2:00

4pPPa5. Binaural perceptual weighting of reverberation level in normal hearing listeners. Gregory M. Ellis (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, g.ellis@louisville.edu) and Pavel Zahorik (Otolaryngol. and Communicative Disord., Univ. of Louisville and Heuser Hearing Inst., Louisville, KY)

In general, perceived reverberation strength is related to physical reverberation level present at the two ears; however, this relation depends on listening condition. Pervious work using virtual auditory space techniques has demonstrated that when physical reverberation is reduced in one ear while leaving the other ear unchanged, listeners do not report a change in reverberation strength. Reducing physical reverberation equally in both ears under binaural listening or in the signal ear under monaural listening elicits a decrease in perceived reverberation strength. To better understand the relation between physical and perceived reverberation in different listening conditions, a perceptual weighting experiment was performed. Listeners reported perceived reverberation strength in a test stimulus using a magnitude estimate paradigm relative to a standard while listening over headphones. Test stimuli were generated by jittering reverberation level independently in the two channels of a binaural room impulse response (BRIR) before convolution with a speech token. BRIRs with reverberation levels of -12, -6, and 0 dB at each ear relative to the standard were tested. Results of a perceptual weighting analysis suggest that listeners use different weighting strategies in different listening conditions, and are generally consistent with a "better ear" hypothesis for reverberation listening.

2:15–2:30 Break

2:30

4pPPa6. Using binaural beat sensitivity to explore mechanisms of bimodal temporal envelope beat sensitivity. Coral Dirks, Peggy Nelson, and Andrew J. Oxenham (Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, hans3675@umn.edu)

Current cochlear implant (CI) fitting strategies aim to maximize speech perception through the CI by allocating all spectral information across the electrode array without regard to tonotopic placement of each electrode along the basilar membrane. For patients with considerable residual hearing in the non-implanted ear, this approach may not be optimal for binaural hearing. This study aims to explore fitting procedures in which CI maps better complement information from the acoustic ear by reducing the frequency mismatch between them. We investigate the mechanisms of binaural temporal-envelope beat sensitivity in normal-hearing listeners using bandpass filtered pulse trains with parameters including stimulus level, filter bandwidth, filter slope, and spectral overlap using bandpass filtered pulse trains. We find the minimum baseline interaural timing difference and spectral mismatch that normal-hearing listeners can tolerate while maintaining their ability to detect interaural timing differences. Initial results consistently demonstrate maximum sensitivity to binaural beats when place of stimulation is matched across ears. The outcomes of this study will provide new information on binaural interactions in normal-hearing listeners and guide methodology for incoming single-sided-deafness patients as we adjust their CI maps in an effort to reduce the frequency-mismatch. [Work supported by NIH grant F32DC016815-01.]

2:45

4pPPa7. Impact of the interaural stimulation timing mismatch on localization performance in bimodal HA/CI users. Stefan Zirn, Julian M. Angermeier (Elec. Eng. and Information Eng., Univ. of Appl. Sci. Offenburg, Badstraße 24, Offenburg 77652, Germany, stefan.zirn@hs-offenburg.de), and Thomas Wesarg (ENT, Medical Ctr. of the Univ., Freiburg, Germany)

The normal-hearing human auditory system is able to perceive interaural time differences (ITD) as small as 10 μs. The largest ITD occurring physiologically is about 700 μs. In bimodal users differences in processing latencies of digital hearing aids (HA) and cochlear implants (CI) up to 9 ms superpose these tiny ITD resulting in an interaural stimulation timing mismatch. Our hypothesis in the present study is that this interaural stimulation timing mismatch impairs sound localization in bimodal HA/CI users. To investigate, we conducted localization tests in bimodal users with and without differences in processing latencies. To compensate for individual processing latency differences we designed a wearable programmable delay line based on a circular buffer implemented on a microcontroller in order to delay the faster device (the CI system). In each subject, an initial localization test with the delay line deactivated was conducted. Afterwards the delay line was activated followed by a 1 hour familiarization period. Finally, the localization test was repeated. Results showed an improvement in sound localization in the horizontal plane of 10 % averaged across 8 bimodal users after compensation. The effect was significant ($p<.05$) using a Wilcoxon signed rank test.

3:00

4pPPa8. Binaural speech unmasking and interference in adult bilateral cochlear-implant users. Matthew Goupell, Olga Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland, College Park, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu), and Joshua G. Bernstein (Walter Reed National Military Medical Ctr., Bethesda, MD)

Bilateral cochlear implants (BICIs) provide improved speech perception in noise, primarily derived from the head-shadow benefit. BICI listeners can also demonstrate binaural speech unmasking or squelch, particularly if the interferer is a single other speaker. However, under the same conditions, some BICI listeners can experience interference. This study tested 21 adult BICI listeners on speech understanding using a range of masker conditions (1, 2, 3, or 4 same-sex talkers, speech-modulated noise, or speech-shaped stationary noise). For listeners that showed binaural unmasking, the amount of unmasking was highest for the one-talker condition and least for the stationary noise, decreasing as the number of interfering talkers increased and the envelope modulation depth decreased. For listeners that showed interference, there were mainly two types of interference patterns: (1) where the amount of interference varied idiosyncratically across masker type, and (2) any stimulation in one ear essentially produced chance performance in the other ear for all masker conditions. The latter interference pattern occurred for the listeners with the largest asymmetries in speech understanding. This suggests, in these extreme cases, a possible central cortical processing bias for one ear and a more equal speech understanding between the ears may diminish interference.

3:15

4pPPa9. Evaluating sound localization cues for premium hearing aids across multiple manufacturers. Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., 516 High St., MS 9171, Bellingham, WA 98225, anna.diedesch@wwu.edu), G. Christopher Stecker (Vanderbilt Univ., Nashville, TN), and Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., Portland VA Health Care System, Portland, OR)

Modern hearing aids employ complex signal processing to improve signal-to-noise ratios for people with impaired hearing. Some of this processing, however, can reduce interaural level differences (ILD), which could reduce sound localization abilities and thus interfere with communication in complex acoustic scenes. It has been speculated that interaural time differences (ITD) are also distorted by amplification, perhaps due to even very brief signal processing delays or multi-path acoustics. Previously we found little ITD distortion for a single set of hearing aids, except when "strong" directional microphones were enabled. To extend and replicate these results, premium hearing aids from four major manufacturers were compared. Aids were set to low-gain amplification and a range of different signal processing algorithms were systematically activated in a manner consistent with clinical practice. Binaural acoustical recordings were measured using an acoustic manikin on which the hearing aids were attached as would be the case for a typical patient. This presentation will describe recordings collected in anechoic and simulated rooms for which frequency-specific ITD and ILD have been extracted and compared to unaided recordings. [Work supported by the F.V. Hunt Postdoctoral Research Fellowship, NIH R01-DC011548 (GCS), R01-DC011828 (FJG), and the VA RR&D NCRAR.]

Session 4pPPb

Psychological and Physiological Acoustics: Clinical Topics and Devices (Poster Session)

Tanvi D. Thakkar, Chair
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All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

4pPPb1. The optimal noise-rejection threshold for normal and impaired hearing. Jordan Vasko, Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm 110, 1070 Carmack Rd, Columbus, OH 43210, healy.66@osu.edu), and DeLiang Wang (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH)

Binary masking represents a powerful tool for increasing speech intelligibility in noise. An essential aspect involves the local criterion (LC), which defines the signal-to-noise ratio below which time-frequency units are discarded. But binary masking is a victim of its own success in one regard—it produces ceiling sentence intelligibility across a broad range of LC values, making the exact optimal LC value difficult to determine. Further, the optimal value for hearing-impaired (HI) listeners is largely unknown. In the current study, the optimal LC was determined in normal-hearing (NH) and HI listeners using speech materials less likely to produce ceiling effects. The CID W22 words were mixed with noise consisting of recordings from a busy hospital cafeteria, then subjected to ideal binary masking. LC values ranged from -20 to + 5 dB relative to the overall SNR of -8 dB. NH subjects were tested at 65 dBA and HI subjects were tested at 65 dBA plus NAL-RP hearing-aid gains. Preliminary results suggest that the optimal LC is similar for NH and HI listeners. Additional conditions involving different speech materials and noise types suggest that the optimal LC can vary as a function of speech and/or noise type. [Work supported by NIH.]

4pPPb2. Effects of self-generated noise on quiet threshold by transducer type in school-age children and adults. Heather Porter, Lori Leibold (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, heather.porter@boystown.org), and Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel HIll, NC)

This study tested the hypothesis that higher levels of self-generated noise in children results in larger child-adult differences for the detection of low-frequency sounds when testing is conducted using transducers associated with a pronounced occlusion effect. Detection thresholds were measured at 125, 250, 500, and 1000 Hz using standard clinical procedures with supra-aural headphones, insert earphones, or a loudspeaker. Probe microphone recordings were made during testing with each transducer. Listeners were 4.5- to 11-year-olds, and adults. Preliminary results are consistent with the a priori hypothesis. For all listeners, thresholds at 125 and 250 Hz were highest with supra-aural headphones and lowest with free-field presentation. This transducer effect was most pronounced for younger children. Child-adult differences were smaller at 500 and 1000 Hz, an observation consistent with reduced effects of self-generated noise with increasing frequency. Trial-by-trial analysis of probe microphone recordings will be presented.

4pPPb3. Integrating remote microphone signals with hearing-aid processing. James Kates and Kathryn Arehart (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

Remote microphones (RM) are used with hearing aids to provide speech signals free from noise and room reverberation. However, because the RM signal bypasses the acoustics of the head and pinna, it can provide an unnatural signal that is lateralized at the center of the head rather than localized at the talker. A structural binaural model is combined with synthesized early room reflections to provide the missing externalization cues. The effectiveness of this simulation has been demonstrated over headphones using linear amplification to compensate for the hearing loss. In a hearing aid, however, the RM processed signal is combined with the acoustic inputs at the hearing-aid microphones, and the processed hearing-aid output is combined with the reverberated speech that is transmitted through the vent or open fitting. The hearing-aid processing implemented in the experiments described in this presentation includes both linear amplification and wide dynamic-range compression. The externalization, spatial soundfield characteristics, and sound quality provided by the entire signal-processing system are evaluated for normal-hearing and hearing-impaired listeners.

4pPPb4. Assessing perceived listening difficulty using behavioral gaze patterns for audiovisual speech. Sterling W. Sheffield and Joshua G. Bernstein (Walter Reed National Military Medical Ctr., 4954 North Palmer Rd, Bethesda, MD 20889, sterling.sheffield.ctr@mail.mil)

Listeners alternate their gaze between the talker's eyes, nose, and mouth during conversation, spending more time gazing directly at the mouth in more difficult listening conditions. Perceived listening difficulty and speech perception performance vary independently, demonstrated with indirect measures of perceived listening difficulty/effort such as pupil dilation and dual task paradigms. This study took a more direct approach, evaluating whether perceived listening difficulty could be assessed by observing a listener's natural gaze pattern in a challenging speech-perception task. The proportion of time spent gazing at the talker's mouth was measured for auditory-visual speech perception in conditions that produced equal percentage-correct scores, but where one condition was designed to produce greater perceived difficulty. Experiment 1 compared gaze proportion for a speech-shaped noise versus two same-gender interfering talkers. Experiment 2 examined the effect of sentence context using low and high-probability sentences. The results are discussed in terms of the potential application of the eye-gaze metric to evaluate rehabilitative interventions in cases where subjectively reported benefits do not manifest in improved speech-understanding scores. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

4pPPb5. Alternative methods for faster and more accurate estimation of hearing threshold. Amitava Biswas and Jennifer Goshorn (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Determination of audiologic threshold is often necessary for variety of diagnoses and treatments. The usual method involves "10 dB down for response and 5 dB up for miss". This common procedure lacks sufficient accuracy to identify smaller changes due to temporary threshold shift and for early diagnosis of degradation of hearing threshold due to ototoxicity or chemotherapy etc. This presentation will explore several alternative methods for faster and more accurate estimation of the threshold.

4pPPb6. Exploring some questions on occlusion or the Bing effect during speech production with a microphone coupled to the speaker's external ear canal. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Sometimes some speakers prefer to use their palm or other objects to cover or occlude at least one ear during public address. This practice may be helpful to enhance their self monitoring of the sound production using the occlusion effect. The basic occlusion effect in the human auditory system has been explored and reported in the literature by several individuals. According to those reports, the speaker can hear his or her own voice significantly louder when the ear canal is blocked at the outer end. Many clinicians routinely utilize detectability of vibrations from a tuning fork when placed on the mastoid process and the ear canal is occluded. This is popularly known as the Bing effect. These empirical data suggest existence of an efficient acoustic connectivity from the vocal tract to the ear canal for healthy individuals. Therefore, this study will explore quantified effects of the classic Bing effect for several speakers across the audio spectrum with a microphone coupled to the speaker's external ear canal.

4pPPb7. Online training for perceptual rehabilitation in cochlear implant users. Damian Almaraz, Elizabeth Een, Elaine Grafelman (Psych., St. Olaf College, Northfield, MN), Justin Pacholec (Comput. Sci., St. Olaf College, Northfield, MN), Yadi Quintanilla, Paula Rodriguez, and Jeremy Loebach (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, loebach@stolaf.edu)

Although cochlear implants have been demonstrated to be effective surgical treatments for deafness, new implant users must undergo an intense period of perceptual learning and adaptation to learn to hear with their prosthesis. Few adult cochlear implant users receive any formal training following implantation, and as a result, the perceptual skills that they develop are highly variable. This study assessed the efficacy of the high variability online training program for adult cochlear implant users dubbed the HiVOlT-CI in 31 normal hearing individuals listening to cochlear implant simulations and three adult cochlear implant users. The goal of our interactive, adaptive, high-variability training program is to provide empirically-based, adaptive and interactive perceptual training to new cochlear implant users to help them adjust to their prosthesis. We also aim to develop a common set of robust and adaptive cognitive auditory skills in cochlear implant users that will help them to hear in real world situations. We also hope to establish baseline levels of performance for experienced cochlear implant users to evaluate successful implant use by new users.

4pPPb8. The development of sentence recognition in noise by cochlear implant users. Chris J. James, Chadlia Karoui, Mathieu Marx, Marie-Laurence Laborde, Marjorie Tartayre, Olivier Deguine, and Bernard Fraysse (ORL, Hôpital Purpan, Pl. du Dr Baylac, Toulouse 31300, France, chris.j.james@wanadoo.fr)

We previously established that under certain physiological conditions the sentence recognition scores in noise of cochlear implant (CI) users one month after activation are similar to those for normal listeners listening to 10–12 channel acoustic (vocoder) simulations (James *et al.*, CIAP, Lake Tahoe, July 2017). We further analysed sentence recognition scores in noise over time in order to determine whether short term, 1-month scores predict long-term outcomes for sentence recognition in noise. Long-term sentence-in-noise data for fixed-SNR-level testing (10, 5, 2.5, 0 dB) were reduced to SNR50 values using a validated analytic sigmoid fit. Data were analysed for 106 adult CI users. 12-month SNR50s were highly correlated with 1-month sentence recognition scores in quiet and 10 dB SNR ($r^2 = 0.29$ and $r^2 = 0.22$, both $p<0.001$): CI users with initially poorer scores could catch up to some extent with early, better performers. However, early high performance (>80% correct score in quiet) was generally associated with 12-months SNR50s $\leq$2 dB. Early sentence recognition scores for CI users may predict their long-term recognition of speech in noise. The number of effective channels available to CI users may evolve to greater than 10-12 in the long term. Disclosure: C.J. is also an employee of Cochlear France. C.K. received partial doctoral funding from Cochlear France as part of the program "Conventions industrielles de formation par la recherche" (CIFRE).

4pPPb9. Binaural intelligibility level difference with a mixed-rate strategy simulation. Thibaud Leclère, Alan Kan, and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Waisman Ctr., Madison, WI 53705, leclere2@wisc.edu)

In noisy situations, bilateral cochlear implant (BiCI) listeners demonstrate poorer spatial release from masking compared to normal hearing (NH) listeners. One reason for this difference could be the limited access for BiCI listeners to fine structure interaural time differences (ITDs) which are crucial for binaural unmasking. Clinical processors convey speech envelopes using pulsatile high-rate (~1000 Hz) carriers, and discard the original temporal fine structure. At high rates, BiCI listeners are not sensitive to ITDs, whereas low rates (< 300 Hz) result in better ITD sensitivity. To preserve both ITD sensitivity and well-sampled speech envelopes, high and low rates are needed. The present study examines whether a mixed rate strategy will enable binaural unmasking in NH listeners using a 16-channel mixed-rate vocoder. Binaural intelligibility level differences were calculated from speech reception thresholds measurements in five mixed-rate configurations. Targets had a 0-μs ITD and masking speech-shaped noises were co-located or separated by applying a 400-μs ITD to the low-rate channels. Results from this experiment will help determine the potential benefits of a mixed-rate strategy to restore spatial unmasking abilities of CI listeners in noisy situations.

4pPPb10. The role of onset cues for lateralization of an auditory object in bilateral cochlear implant listeners. Tanvi D. Thakkar, Alan Kan, and Ruth Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 934B Eagle Heights Dr., Madison, WI 53705, tthakkar@wisc.edu)

In bilateral cochlear implant (BiCI) listeners, a common strategy for improving sensitivity to interaural timing differences (ITDs) is to introduce low rate stimulation in some channels. Previous work showed that BiCI listeners can lateralize a single sound with a constant ITD at every pulse of a low rate stimulus. However, in reverberant environments, ITDs for a target sound can change over a short period of time. Previously we have shown that BiCI listeners had difficulty using the ITD in the first pulse of a low-rate pulse train to lateralize a stimulus when the subsequent pulses had ITDs that varied over time. Hence, we hypothesized that a more salient onset is needed in order to restore ITD sensitivity in BiCI listeners when using low rate stimulation. In this study, we measured the number of pulses with a consistent ITD that is needed at the onset in order for BiCI listeners to reliably perceive one sound in the correct location. Results suggest that BiCI listeners need at least five pulses with consistent ITDs in order to report a single auditory object that is correctly lateralized. These results have important implications for low rate strategies aimed at restoring ITD sensitivity to BiCI listeners.

4pPPb11. The effect of movement, compression, and cochlear-implant processing on the output of cochlear implants. Alan Archer-Boyd and Robert P. Carlyon (Univ. of Cambridge, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, alan.archer-boyd@mrc-cbu.cam.ac.uk)

Dynamic-range compression is used in cochlear implants (CIs) to compensate for the greatly reduced dynamic range of CI listeners. It has been found that unlinked bilateral compression in hearing aids (HAs) can affect the spatial location of sounds, reducing interaural level differences (ILDs) and causing sound sources to appear closer to 0° azimuth and/or more diffuse (Wiggins and Seeber, 2011). The compressors in CI processors are often much stronger than hearing aids (e.g., CI compression ratios of 12:1 vs. ≤3:1 in HAs). We investigated the effect of compression on monaural and binaural level cues during simulated rotational movements in the horizontal plane. Moving signals were created by combining static behind-the-ear hearing aid microphone impulse responses at successive angles. For sound levels above the compressor threshold, fast movements over wide angles (120°/s) altered the size and extent of monaural level changes and ILDs considerably, relative to those from those that would occur naturally and without compression. The relatively slow attack and release times of the compressor also had the effect of producing a "lag" in the level changes after the movement had stopped. Microphone directionality was also found to have an effect on the output of the compressor during movement.

4pPPb12. The segregation of sequentially interleaved sentences by normal-hearing and cochlear-implant listeners using monaural level cues and the effect of dynamic-range compression. Alan Archer-Boyd and Robert P. Carlyon (Univ. of Cambridge, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, alan.archer-boyd@mrc-cbu.cam.ac.uk)

Normal-hearing (NH) listeners can use level differences to segregate pairs of vocoded sentences that are presented to interleaved frequency channels; performance is better at both positive and negative target-to-masker ratios (TMRs) compared to a TMR of 0 dB (Ihlefeld and Shinn-Cunningham, 2008). The present study investigated whether unilateral cochlear-implant (CI) users could also use level differences alone to segregate two sentences. Due to the greatly reduced frequency resolution and increased TMR thresholds of CI users, the sentences were interleaved in time and presented sequentially. The Boston University (Kidd *et al*., 2008) corpus was used to construct interleaved, five-word sentences, consisting of name, verb, number, adjective, and object. The presentation order of the words alternated randomly for each word type. Each sentence began with the same two words and listeners were required to select the last three words in the target sentence. The level differences ranged from -12 to 12 dB. A monaural vocoder study showed that the task was possible for both with younger and older NH listeners, with performance increasing as level difference increased. The results of the CI user study will be presented, and the effects of dynamic-range compression on their performance discussed.

4pPPb13. Effects of broad binaural fusion and hearing loss ondichotic concurrent vowel identification. Lina Reiss (Otolaryngol., Oregon Health & Sci. Univ., 3181 SW Sam Jackson Park Rd., Mailcode NRC04, Portland, OR 97239, reiss@ohsu.edu), Sara Simmons, Larissa Anderson (School of Audiol., Pacific Univ., Portland, OR), and Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., Portland Veterans Administration, Portland, OR)

Many hearing-impaired (HI) individuals have abnormally broad binaural pitch fusion, such that tones differing in pitch by up to 3-4 octaves are fused between ears (Reiss *et al*., 2017). Broad binaural fusion leads to averaging of the monaural pitches (Oh and Reiss, 2017), and may similarly lead to fusion and averaging of speech streams across ears. In this study, we examined the relationship between binaural fusion range and dichotic vowel identification in normal-hearing (NH) and HI listeners. Synthetic vowels /i/,/u/,/a/, and /ae/ were generated with three fundamental frequencies (F_0) of 106.9, 151.2, and 201.8 Hz, and presented dichotically through headphones. For HI listeners, stimuli were shaped according to NAL-NL2 prescriptive targets. Listeners identified 1 vowel or 2 different vowels on each trial. Preliminary results indicate that NH listeners improved their percentage of both vowels identified as the ΔF_0 increased, but HI listeners did not. HI listeners were more likely to fuse two vowels together even with large ΔF_0. NH listeners with broad fusion also had poorer overall performance than those with sharp fusion. The findings suggest that broad binaural fusion can impede separation of voices and speech streams in both NH and HI listeners. [Work supported by NIH-NIDCD grant R01 DC013307.]

4pPPb14. Mechanisms underlying speech masking release in hybrid cochlear implant users. Viral Tejani (Dept. of Otolaryngology-Head & Neck Surgery, Univ. of Iowa Hospitals and Clinics, 200 Hawkins Dr., 21003 PFP, Iowa City, IA 52242, viral-tejani@uiowa.edu) and Carolyn Brown (Commun. Sci. & Disord., Univ. of Iowa, Iowa City, IA)

Hybrid cochlear implant (CI) users utilize low-frequency acoustic and high-frequency electric stimulation in the implanted ear, outperform traditional CI users in speech perception in noise (Gantz *et al*., 2017), and show "masking release," an improvement in speech recognition in fluctuating noise relative to steady noise (Turner *et al*., 2004). Improved performance in noise has been attributed (but not proven) to the spectral resolution and temporal fine structure (TFS) provided by acoustic hearing. Spondee recognition in two-talker and ten-talker babble was assessed in Hybrid CI users, and masking release was calculated as the percent difference in performance in two-talker (easy condition) and ten-talker (hard condition) babble. Spectral ripple density discrimination thresholds and fundamental frequency discrimination limens was also assessed, as these psychophysical measures reflect underlying spectral resolution and TFS and correlate with speech perception in noise in traditional CI users (Won *et al*., 2007; Goldsworthy *et al*., 2013; Goldsworthy, 2015; Jeon *et al*., 2015). Preliminary data suggest masking release is possible in Hybrid CI users, and that psychophysical measures and masking release may be correlated, suggesting a role of spectral resolution and TFS in masking release. The extent of residual hearing also appears to correlate with psychophysical and masking release performance.

4pPPb15. Modulation interference in cochlear implants. Monita Chatterjee and Aditya M. Kulkarni (Boys Town National Res. Hospital, 555 N 30th St, Omaha, NE 68131, monita.chatterjee@boystown.org)

Modulation interference, in which the temporal envelope of a signal is rendered less salient due to the presence of a competing temporal envelope of a masker, has been modeled as the result of modulation-tuned filters operating in the midbrain. In listeners with cochlear implants (CIs) who rely more heavily on the speech temporal envelope, such interference may be particularly problematic for speech perception in competing, fluctuating backgrounds. Here, we present evidence for modulation interference in CI users from our laboratory, using both non-speech and speech targets. Maskers were presented either in the same ear or in the contralateral ear relative to the target, and either concurrently with the target or preceding the target (forward masking). Tasks include the detection of the temporal envelope on the target as well as the discrimination of temporal envelopes between targets. Results suggest that modulation interference plays an important role in CI users, reducing the salience of the speech temporal envelope under a variety of masking conditions.

4pPPb16. Improving temporal modulation sensitivity in cochlear implant users. Shaikat Hossain (Otolaryngol., Univ. of Southern California, 7220 Mccallum Blvd #307, #307, Dallas, TX 75252, shossa3@gmail.com), Susan R. Bissmeyer (Biomedical Eng., Univ. of Southern California, Newhall, CA), and Raymond L. Goldsworthy (Otolaryngol., Univ. of Southern California, Los Angeles, CA)

The present study investigated the limitations of temporal modulation sensitivity in cochlear implant (CI) users. Adult users of Cochlear Nucleus devices were tested by presenting sounds through their clinical processors using an auxiliary cable and using direct electrical stimulation provided through a custom research processor. Threshold and comfort current levels were determined by adjusting the phase duration of 10 Hz amplitude modulated pulse trains of 400 ms duration at a given stimulation rate. After mapping the participants, modulation detection thresholds were measured by means of a 3 alternative forced choice procedure which adaptively adjusted the modulation depth to determine the threshold in dB relative to full modulation. Participants completed the task using a graphical user interface to indicate which interval was different. Modulation frequencies from 10 to 800 Hz were tested at various stimulation rates on basal and apical electrodes in both psuedomonopolar and bipolar modes of stimulation. Preliminary findings indicate that modulation sensitivity drops off with increases in modulation frequency. Performance was significantly better with direct stimulation as compared to through the clinical processor and better at higher modulation frequencies than previous findings from CI users. Outcomes will have implications for the development of next generation CI devices.

4pPPb17. Comparison of different implementations of the ACE cochlear implant signal processing strategy with an objective metric. Bernardo Murta (Federal Univ. of Santa Catarina, Rua da Passagem, 111, Belo Horizonte, Minas Gerais 30220-390, Brazil, be.murta@gmail.com), Rafael Chiea, Gustavo Mourão (Federal Univ. of Santa Catarina, Florianopolis, Brazil), Stephan Paul (Federal Univ. of Santa Catarina, Joinville, Brazil), and Julio A. Cordioli (Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

Research that seeks to compare outcomes of Cochlear Implant (CI) processing strategies must ensure that different implementations of a given strategy are safe, and means for their comparison are required. This work aims to analyze the sensitivity of an objective metric to the effects introduced by different sets of filter bank models, compressing functions, envelope detectors and other aspects in implementations of the same CI signal processing strategy. To this end a subset of signals containing pure tones, complex harmonics, exponential sweeps and speech signals is defined and processed by means of the same CI strategy (ACE) in two different implementations: a standard version from the Cochlear Nucleus Toolbox and another version implemented by the authors using published data (FSC). As expected, variations in the output are due to different aspects in the signal processing chain of each implementation. While the proposed metric is sensitive to this differences, its magnitude also depends heavily on the test signals used and specificity is very low. Therefore a more accurate, specific and sensitive protocol was developed using UTD's CCi-Mobile Research Interface to analyze the root of the differences. This includes a metric and performance criteria that are specific for certain signal processing parameters.

4pPPb18. Mitigating the effects of reverberation and noise in cochlear implants. Kevin M. Chu (Biomedical Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27708, kmc100@duke.edu), Chandra S. Throckmorton, Leslie M. Collins, and Boyla O. Mainsah (Elec. and Comput. Eng., Duke Univ., Durham, NC)

In listening environments with room reverberation and background noise, cochlear implant (CI) users experience substantial difficulties in understanding speech. Because everyday environments have different combinations of reverberation and noise, there is a need to develop algorithms that can mitigate both effects to improve speech intelligibility. Desmond *et al.* (2014) developed a machine learning approach to mitigate the adverse effects of late reverberant reflections of speech signals by using a classifier to detect and remove affected segments in CI pulse trains. In this study, we investigate the robustness of the reverberation mitigation algorithm in environments with both reverberation and noise. We conducted sentence recognition tests in normal hearing listeners using vocoded speech with unmitigated and mitigated reverberant-only or noisy reverberant speech signals, across different reverberation times and noise types. Improvements in speech intelligibility were observed in mitigated reverberant-only conditions. However, mixed results were obtained in the mitigated noisy reverberant conditions as a reduction in speech intelligibility was observed for noise types whose spectra were similar to that of anechoic speech. Based on these results, the focus of future work will be to develop a context-dependent approach that activates different mitigation strategies for different acoustic environments. [Research supported by NIH grant R01DC014290-03.]

4pPPb19. Band importance functions of listeners with sensorineural hearing impairment. Sarah E. Yoho (Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84321-6746, sarah.leopold@usu.edu) and Adam K. Bosen (Boys Town National Res. Hospital, Omaha, NE)

The Speech Intelligibility Index (SII) includes a series of band importance functions for calculating the estimated intelligibility of speech under various conditions. Band importance functions describe the relative contribution of discrete spectral regions of speech to overall speech intelligibility. They are generally measured by filtering the spectrum into a series of frequency bands and determining intelligibility when each band is present, relative to intelligibility when other bands are present. The functions found in the SII are commonly used to provide estimations of intelligibility for listeners with hearing loss, even though they were developed with the use of normal-hearing listeners. In the current study, band importance functions were derived for individual listeners with sensorineural hearing impairment and a group of normal-hearing control participants. Functions were measured by filtering sentences to contain only random subsets of bands on each trial, and regressing speech recognition against the presence or absence of each band across trials. Preliminary results indicate that the shape of hearing loss plays an important role in the shape of band importance functions, with some listeners who have high-frequency sloping losses displaying decreased and even negative importance of high-frequency speech bands.

4pPPb20. Utilizing cross-channel features for reverberation mitigation in cochlear implants. Lidea K. Shahidi, Chandra S. Throckmorton, Leslie M. Collins, and Boyla O. Mainsah (Elec. and Comput. Eng., Duke Univ., 100 Sci. Dr., Durham, NC 27710, lidea.shahidi@duke.edu)

Cochlear implant (CI) users experience difficulties in understanding speech in reverberant environments as both active speech and non-informative speech segments of the reverberant signal are incorporated into the CI stimulus pattern. The non-informative speech segments are due to late reverberant signal reflections and are referred to here *as overlap-masking* segments. Desmond *et al.* (JASA, 2014) showed that removal of all overlap-masking segments from the CI stimulus pattern significantly improved the intelligibility of reverberant speech in CI users. Desmond (2014, Ph.D. dissertation) proposed a reverberation mitigation strategy based on training a machine learning classifier to identify and delete overlap-masking segments in CI pulse trains. The mitigation strategy employed *within-channel features* to mitigate overlap-masking effects in a given electrode channel. However, since adjacent electrode channels tend to enter the overlap-masking state concurrently, additional information about the state of other channels might provide useful information for classification. In this work, we explore the use of *cross-channel features* to improve classification performance. We find significant improvements in classification performance in low-frequency electrode channels after the addition of cross-channel features. Preliminary results from normal-hearing listeners performing speech recognition tests using vocoded speech under simulated reverberation will be reported. [Research supported by NIH grant R01DC014290-03.]

4pPPb21. Estimating the relative interaural place mismatch for bilateral cochlear-implant listeners. Olga A. Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland, College Park, 4954 North Palmer Rd., Bldg. 19, R. 5607, Bethesda, MD 20889, olga.stakhovskaya.ctr@mail.mil), Joshua G. Bernstein (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Kenneth K. Jensen (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Eden Prairie, MN), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD)

Frequency-matched bilateral input is important for the optimum encoding of binaural cues to facilitate sound localization, auditory scene analysis, and speech understanding in noise. Current bilateral cochlear-implant (CI) programming procedures do not account for potential interaural place-of-stimulation mismatch. This study investigated whether a perceptual test of interaural-time-difference (ITD) sensitivity can effectively estimate relative interaural mismatch for bilateral CI listeners. Ten bilateral CI listeners were tested on a two-interval left-right ITD discrimination task. Loudness balanced 300-ms, 100 pulse-per-second constant-amplitude pulse trains were delivered to single-electrode pairs using direct stimulation. Measurements were made for five reference electrodes evenly distributed along the array in one ear, and for at least five closely spaced comparison electrodes in the other ear for each reference electrode. The pair with the smallest thresholds was assumed to be the most closely place matched. ITD sensitivity was variable across listeners, with thresholds ranging from 80-2000 μs. ITD tuning curves varied across and within listeners, ranging from widths of approximately one (sharp tuning) to eight electrodes (broad tuning). While cases with narrower tuning provide clear suggestions for frequency programming to minimize interaural mismatch, estimates of mismatch were difficult to ascertain for cases with broad tuning.

Session 4pPPc

Psychological and Physiological Acoustics: Sensitivity to Spectral and Temporal Information

Christian Stilp, Chair
Psychological and Brain Sciences, University of Louisville, 308 Life Sciences Building, Louisville, KY 40292

Contributed Papers

3:45

4pPPc1. Infants' use of amplitude modulation cues in the segregation of concurrent vowels. Monika-Maria Oster and Lynne A. Werner (Speech and Hearing Sci., Univ. of Washington, 1417 North East 42nd St, Seattle, WA 98105, mmoster@uw.edu)

Infants have difficulties separating speech from competing sounds. One explanation is that infants do not use acoustic cues to sound source segregation as adults do. This study investigated 3- and 7-month-old infants' and adults' ability to use AM cues in concurrent vowel segregation. Seven American-English vowels were produced by a male and by a female talker. One male and one female vowel were randomly chosen and superimposed. A train of such vowel pairs was presented to listeners, who were trained to respond to the male target vowel /i:/ or /u:/. Target-to-nontarget ratios were chosen to equate performance across age groups. In the baseline condition, all vowels had flat envelopes. In the cue condition, male vowels were modulated with three envelopes from infant-directed vowels. The proportion of 3-month-old infants achieving an 80% correct criterion at a ratio of +15dB was higher in the cue than in the baseline condition. For 7-month olds and adults d' could be estimated based on 15 target and 15 nontarget trials in each condition. Preliminary results indicate that both 7-month-old infants and adults have a higher d' in the cue than in the baseline condition. Thus, infants appear to use AM to segregate concurrent vowels.

4:00

4pPPc2. Effects of age on the discrimination of modulation type for 2- and 10-Hz rates. Brian C. Moore, Sashi Mariathasan (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk), and Aleksander Sek (Inst. of Acoust., Adam Mickiewicz Univ., Poznan, Poland)

Frequency modulation (FM) at a rate of 10 Hz may be detected via conversion of FM to amplitude modulation (AM) in the cochlea, while 2-Hz FM may be detected partly using temporal fine structure (TFS) information. Greater age may impair the processing of TFS information while sparing the processing of AM information. To test these ideas, while controlling for the effects of detection efficiency, a two-stage experiment was conducted. In stage 1, psychometric functions were measured for the detection of AM alone and FM alone imposed on a 1000-Hz carrier, using 2- and 10-Hz rates. In stage two, the task was to discriminate AM from FM at the same modulation rate when the detectability of the AM alone and FM alone was equated. For young normal-hearing subjects, discrimination was markedly better for the 2-Hz than for the 10-Hz rate, consistent with the idea that FM is coded in a different way from AM at 2 Hz. For older subjects with normal hearing at 1000 Hz, a similar pattern was obtained, but discrimination of AM from FM at 2 Hz was somewhat poorer than for the young normal-hearing subjects, consistent with an age-related reduction in ability to use TFS cues.

4:15

4pPPc3. Natural signal statistics and the timecourse of spectral context effects in consonant categorization. Christian Stilp and Ashley Assgari (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Speech perception is heavily influenced by surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects* (*SCEs*) that bias categorization of later sounds. For example, when context sounds have a low-F3 bias, listeners report more high-F3 responses to the target consonant (/d/); conversely, a high-F3 bias in context sounds produces more low-F3 responses (/g/). SCEs have been demonstrated using a variety of approaches, but most often, the context was a single sentence filtered two ways (*e.g.*, low-F3 bias, high-F3 bias) to introduce long-term spectral properties that biased speech categorization. Here, consonant categorization (/d/-/g/) was examined following context sentences that naturally possessed desired long-term spectral properties without any filtering. Filtered sentences with equivalent spectral peaks were included as controls. For filtered context sentences, as average spectral peak magnitudes (*i.e.*, filter gain) increased, SCE magnitudes (*i.e.,* category boundary shifts) increased linearly. However, unfiltered context sentences showed no relationship between long-term average spectral energy and subsequent SCEs. Instead, spectral energy in the last 500 ms of unfiltered sentences predicted SCE magnitudes. Results highlight important considerations about how long-term versus short-term spectral energy in preceding sounds affects categorization of later sounds.

4:30

4pPPc4. Complex frequency modulation detection and discrimination at low and high frequencies. Kelly L. Whiteford and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Whether frequency modulation (FM) is represented by place (tonotopic) or temporal (phase-locking) information in the peripheral system may depend on the carrier frequency (f_c) and the modulation rate (f_m), with only f_cs below 4 kHz and f_ms below 10 Hz thought to involve temporal coding. This study tested the role of temporal coding in harmonic complexes by measuring FM detection and discrimination for two F0s (200 Hz and 1400 Hz), modulated at slow (1 Hz) and fast (20 Hz) rates, for tones with lower (2-5) or upper (6-9) harmonics embedded in threshold equalizing noise. Pure-tone FM detection was measured for f_cs between 200 and 12000 Hz at the same f_ms. In detection tasks, participants selected which of two intervals contained FM. In discrimination tasks, participants selected which of three FM complex tones was incoherently modulated. Preliminary results suggest better FM detection for slow than fast rates, even when all tones are above 4 kHz. However, this effect was stronger at lower frequencies, where phase-locking cues may be viable. Most participants were only able to discriminate inharmonicity for slow f_ms in conditions with components below 8000 Hz. [Work supped by NIH grant R01DC005216 and an Eva O. Miller Fellowship.]

4:45

4pPPc5. Frequency modulation improves sensitivity to interaural timing differences at high frequencies in the presence of noise. Alan Kan (Univ. of Wisconsin-Madison, 1500 Highland Ave, Madison, WI 53705, ahkan@waisman.wisc.edu)

Sensitivity to interaural timing differences (ITDs) at high frequencies has largely been attributed to slow amplitude modulations (AM) in the envelope. However, in noisy situations, high-frequency signal envelopes can easily be corrupted making AM less useful for sound localization. Previous work has shown that frequency modulation (FM) introduced into a high frequency pure tone can also yield sensitivity to ITDs. This suggests that FM may be an additional cue used by the auditory system to resolve high-frequency ITDs, which may be useful in noisy situations. This experiment tests the hypothesis that in-phase AM and FM are complementary cues that improve ITD sensitivity at high frequencies in the presence of noise. A 4-kHz tone stimulus with a fixed ITD of 500 μs was used, and had either 200 Hz AM, FM or in-phase AM + FM imposed. The signal-to-noise ratio (SNR) needed to achieve 70.7% correct in a left-right discrimination task was measured. Results showed that the SNR needed for FM was typically lower than AM, and AM + FM was significantly lower than either AM or FM alone. The results suggest that in-phase AM and FM may provide complementary cues that aid in discriminating ITDs at high frequencies in the presence of noise.

5:00

4pPPc6. The gammawarp filterbank: A frequency-warped implementation of the dynamic compressive gammachirp filter. James Kates (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

This paper presents an alternative method for implementing a dynamic compressive gammachirp (dcGC) auditory filterbank. Instead of using a cascade of second-order sections, the new approach uses digital frequency warping to give the gammawarp filterbank. In the warped implementation, the unit delays in a conventional finite impulse-response (FIR) filter are replaced by first-order allpass filters. The set of warped FIR filter coefficients is constrained to be symmetrical, which results in a filter phase response that is identical for all filters in the filterbank. The dynamic variation in filter response is provided by interpolating the outputs of three linear filters adjusted for low-, medium-, and high-input signal levels. The gammawarp filterbank offers a substantial improvement in execution speed compared to previous dynamic gammachirp filter implementations. For a linear filterbank, the gammawarp execution time is comparable to that of a gammatone or linear second-order section gammachirp implementation for 32 bands, but there is a processing time advantage that increases as the number of bands is increased. For a dynamic compressive gammachirp filterbank, the gammawarp implementation is 27 to 47 times faster than the dcGC MATLAB code of Irino.

5:15

4pPPc7. Musical instrument categorization is highly sensitive to spectral properties of earlier sounds. Ashley Assgari, Jonathan Frazier, and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu)

Auditory perception is shaped by spectral properties of surrounding sounds. For example, when spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects* (*SCEs*; *i.e.*, categorization boundary shifts) that bias perception of later sounds. SCEs influence perception of speech and nonspeech sounds alike. When categorizing vowels or consonants, SCE magnitudes increased linearly with greater spectral differences between contexts and target speech sounds [Stilp *et al.* (2015) *JASA*; Stilp & Alexander (2016) *POMA*; Stilp & Assgari (2017) *JASA*]. Here, we tested whether this linear relationship between context spectra and SCEs generalizes to nonspeech categorization. Listeners categorized musical instrument targets that varied from French horn to tenor saxophone. Before each target, listeners heard a one-second music sample processed by spectral envelope difference filters that amplified / attenuated frequencies to reflect the difference between horn and saxophone spectra. By varying filter gain, filters reflected part of (25%, 50%, 75%) or the full (100%) difference between instrument spectra. As filter gains increased to reflect more of the difference between instrument spectra, SCE magnitudes increased linearly, parallel to speech categorization. Thus, a close relationship between context spectra and biases in target categorization appears to be fundamental to auditory perception.

Session 4pSA

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

Christina J. Naify, Cochair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr, Pasadena, CA 91109

Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Invited Papers

1:30

4pSA1. Bianisotropic acoustic metasurfaces for wavefront manipulation. Benjamin M. Goldsberry, Andrew J. Lawrence, and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758, haberman@arlut.utexas.edu)

Materials with sub-wavelength asymmetry and long-range order have recently been shown to respond to acoustic waves in a manner that is analogous to electromagnetic biansiotropy [Sieck *et al*. Phys. Rev. B **96**, 104303, (2017)]. A characteristic of these materials is the generation of dipolar scattering when subjected to a uniform time-varying pressure field. This behavior leads to a characteristic acoustic plane-wave impedance that has a preferential direction, known as a polarization, which results in direction-dependent magnitude and phase of an acoustic field scattered from bianisotropic acoustic media. These materials can therefore be used as acoustic metasurfaces to control the reflected or transmitted acoustic fields. This work presents the use of bianisotropic acoustic media for wavefront manipulation through analytical and finite element modeling and discusses potential acoustic analogues to existing bianisotropic electromagnetic metasurfaces. [This work was supported by ONR and the National Science Foundation.]

1:50

4pSA2. Wave control with “Time Materials”. Mathias Fink (Langevin Inst., 1 rue Jussieu, Paris 75005, France, mathias.fink@espci.fr)

Phononic crystals and Metamaterials are made from assemblies of multiple elements usually arranged in repeating patterns at scales of the order or smaller than the wavelengths of the phenomena they influence. Because time and space play a similar role in wave propagation, wave propagation is affected by spatial modulation or by time modulation of the refractive index. Here we emphasize the role of time modulation. We show that sudden changes of the medium properties generate instant wave sources that emerge instantaneously from the entire wavefield and can be used to control wavefield and to revisit the way to create time-reversed waves. Experimental demonstrations of this approach will be presented More sophisticated time manipulations can also be studied in order to extend the concept of phononic crystals in the time domain.

Contributed Papers

2:10

4pSA3. Reciprocity theorems and perception of non-reciprocal behavior in vibrating systems, imetamaterials, acoustic devices, and electroacoustic devices. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

Equations governing physical systems often admit a reciprocity corollary. Such systems include mechanical, acoustical, vibrational, and electroacoustic systems. There is no universally applicable succinct definition of reciprocity, although it is sometimes loosely stated that reciprocity means that, “if you see it, then it can see you.” There have been a considerable number of derivations of reciprocity theorems for different types of acoustical and mechanical systems published over the years — notable names are Helmholtz, Rayleigh, Lamb, Lyamshev, Schottky, Foldy, Primakoff, Payton, de Hoop, Knopoff, Gangi, Arntsen, Carcione, and Fokkema. There appears to be no universally applicable derivation, and no universally applicable reciprocity principle. The emergence of metamaterial systems poses new challenges and opportunities in regard to reciprocity. Given one such system, one naturally asks if there is some reciprocity theorem that applies. If one perceives that there is no such theorem, then one might naturally ask whether the perception is wrong, and if it is simply the case that no one has yet bothered to try to derive one or that one is not sufficiently clever to find one. It may also be that there is an analogous system for which there is a theorem, and that the theorem applies for the present system in some approximate sense. If so, how well? Present paper examines several metamaterial systems introduced in the recent literature from the viewpoint point of reciprocity theorems.

2:25

4pSA4. Nonlinear properties of an electrically coupled acoustic metamaterial. Jason J. Smoker and Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, jason.smoker@navy.mil)

Efforts in the field of acoustic metamaterials have often remained in the material design and property identification domain due to difficulty in realizing physical manifestations of necessary material properties. This is especially true in the domain of transformation acoustics. In a number of classical approaches, research has relied on physical dynamics in the linear region to design an effective density or bulk modulus. Such work has limited achievable designs. This work aims to demonstrate the use of electrically coupled non-linear dynamics to increase the design space of acoustic metamaterials. Numerical models are leveraged for analysis and utilized to demonstrate efficiencies in real world configurations.

2:40

4pSA5. Extreme anisotropy over an acoustic hyperbolic metasurface. Li Quan and Andrea Alu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu)

The emergence of optical hyperbolic metamaterials has attracted significant attention in the last decade due to its extreme anisotropy, linked to largely enhanced local density of states. This concept has been also extended to acoustics, and acoustic hyperbolic metamaterials have been realized. However, the fabrication complexity of these metamaterials, their inherent sensitivity to loss and disorder, and the fact that the hyperbolic modes are not easily accessible, as they are trapped in the bulk of the material prevent the wide use of hyperbolic metamaterials for sound. Our research group has recently proposed the idea of hyperbolic metasurfaces, for which extreme anisotropy over a surface enables plasmons that propagate with similar unbounded local density of states, but restricted to surface propagation. In this work, we extend these findings to acoustics, simplifying the realization of hyperbolic propagation and observation, and at the same time preserving the exciting properties of hyperbolic metamaterials. By carefully designing the acoustic microstructure decorating a metasurface, we successfully confine acoustic waves traveling along the interface with air and observe hyperbolic dispersion and in principle unbounded local density of states. These properties may open exciting opportunities for acoustic and surface acoustic wave devices.

2:55

4pSA6. Non-reciprocal bilinear structures. Andrew Norris, Zhaocheng Lu (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu), Samuel P. Wallen, Benjamin M. Goldsberry, Michael R. Haberman (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Tyler J. Wiest, and Carolyn C. Seepersad (Mech. Eng., Univ. of Texas, Austin, TX)

Structures that display non-reciprocal behavior resulting from nonlinear effects are presented. The nonlinear mechanism is bilinear stiffness, also known as bimodular elastic response. Specific realizations considered are in the form of bilinear springs in series making a finite degree of freedom unidimensional nonlinear structure. Recall that a system is reciprocal if $F_B u_{B,A}= F_A u_{A,B}$ where is $u_{B,A}$(respectively $u_{A,B}$) is the displacement at point *B* (*A*) resulting from a force F_A applied at *A* (F_B applied at *B*). We say the system is fully non-reciprocal if the reciprocity relationship is violated for any sign of the applied forcing, positive or negative, compressive or tensile. Perhaps the simplest system displaying full non-reciprocity is a two degree-of-freedom spring-mass-spring-mass-spring structure, fixed at both ends. We first describe the static and low frequency behavior, illustrating full non-reciprocity. Similar systems with many bilinear springs and masses display nonlinear traveling wave effects, including pulse spreading and shock formation depending on whether the leading edge of the incident pulse is compressive or tensile, respectively. The talk will also discuss fabrication and optimization of the bilinear springs using additive manufacturing. Work supported by NSF-EFRI.

3:10–3:25 Break

3:25

4pSA7. Linear nonreciprocal transmission of sound in a viscous fluid with asymmetric scattteres. Arkadii Krokhin (Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, arkady@unt.edu), Ezekiel Walker (Echonovus Inc, Denton, TX), Andrii Bozhko, and Arup Neogi (Phys., Univ. of North Texas, Denton, TX)

Reciprocity theorem is hold for wave equation due to time-reversal symmetry. Dissipative losses introduced by adding imaginary part to elastic modulus make sound propagation irreversible but the reciprocity theorem remains valid. In viscous fluid acoustic oscillations of local velocity follow the Navier-Stokes equation, which is not time-reversible. However, broken T-symmetry does not necessarily lead to nonreciprocity. Here we demonstrate that the necessary condition for nonreciprocal propagation in a viscous fluid is broken P-symmetry, which can be achieved by introducing asymmetric scatterers in a viscous environment. The experiment was done with a phononic crystal of Al rods in water. The rods have asymmetric cross-section in a form of a circular sector with an arc of 120 degrees. The measured transmission spectrum exhibits signatures of nonreciprocity within a range of frequencies from 300 to 450 kHz. The experimental results are in agreement with numerical simulations based on the linearized Navier-Stokes equation. The nonrecipocity is due to different viscous losses accumulated along sound propagation in two opposite directions with broken P-symmetry. Nonreciprocity increases for the rods with more rough surfaces since dissipation occurs mainly in a viscous boundary layer of thickness of few microns. Unlike previously proposed nonreciprocal devices based on nonlinearity or local spinning of fluid, our passive device is small, cheap, and does not require energy source. [Work supported by the NSF under Grant No. 1741677.]

3:40

4pSA8. Optimal absorption associated with wall impedance in acoustic waveguides. Matthew Kelsten and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu)

In a previous talk at ASA it was shown that the double root for modal frequencies in a 2D waveguide can exhibit almost perfect absorption over a relative broad frequency range. The key to the phenomenon is that the wall impedance is such that modes coalesce at a complex-valued frequency. In this talk we consider the more realistic case of circularly symmetric waveguides. Numerical simulations are shown for reflection and transmission from a waveguide with impedance boundary conditions associated with double roots. The results again show large and almost perfect absorption at frequencies near the real part of the complex root. The model is linked to realistic passive impedance values based on various models for boundary impedance, including perforated and porous panels. These comparisons are discussed to illustrate the feasibility of optimized wall impedances in absorbing sound passing through ducts. Preliminary experimental data will be presented. Work supported by NSF.

3:55

4pSA9. Tunable acoustic energy dissipation by fluid-structure interaction in elastomeric, resonant metamaterials. Shichao Cui and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave, Columbus, OH 43210, cui.408@osu.edu)

Helmholtz resonators are a classic means to absorb low frequency acoustic waves with great effectiveness and straightforward implementation. Yet, traditional Helmholtz resonators are tuned to absorb a specific frequency of wave energy and are unable to adaptively tailor damping capability. On the other hand, recent research on elastomeric metamaterials offers concepts for large and tunable damping properties using internal constituents that buckle to trap large elastic energy when subjected to geometric or load constraints. The research reported here draws from the principles of Helmholtz resonance and constrained metamaterials to device a resonant metamaterial with tunable acoustic energy dissipation. Using the fluid-structure interaction of an internal beam-like member with the resonator chamber, the metamaterial absorbs acoustic waves at a target frequency and tailors energy dissipation by changing external constraints that relatively magnify or suppress the interaction between acoustic pressure and damped beam member. An analytical model is developed to qualitatively characterize the behavior of the metamaterial observed in the laboratory. From the combined experimental and analytical studies, it is found that the metamaterial may significantly reduce the sound pressure level at the targeted frequency range while modulating the broadness of the absorption effect by way of external load control.

4:10

4pSA10. Acoustic gradient index lens design using gradient based optimization. Feruza Amirkulova, Samuel T. Caton, Morgan Schrader, Trenton J. Gobel (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, feruza.amirkulova@wne.edu), and Andrew Norris (Mech. engineeringand Aerosp., Rutgers Univ., Piscataway, NJ)

We demonstrate a gradient based optimization of the absolute acoustic pressure at a focal point by incrementally repositioning a set of cylindrical obstacles so that they eventually act as an effective gradient index (GRIN) lens. The gradient-based optimization algorithm maximizes the absolute pressure at the focal point by analytically evaluating its derivative with respect to the cylinder positions and then perturbatively optimizing the position of each cylinder in the GRIN lens device while taking into account acoustic multiple scattering between the cylinders. The method is presented by examples of sets of hard cylinders and sets of elastic cylindrical shells of uniform size. Designs using available cylindrical shells enable easy and realistic fabrication of acoustic GRIN lenses. Computations are performed on MATLAB using parallel optimization algorithms and a multistart optimization solver, and supplying the gradients of absolute pressure at the focal point with respect to position vectors. Providing the analytic form of the gradients enhances modeling by allowing the use of exact gradients with optimization algorithms and parallel computing. This results in reducing number of function calls and time needed to converge, and improving solution accuracy for large scale optimization problems especially at high frequencies and with a large number of scatterers.

4:25

4pSA11. Distorting an impulse wave with phononic metamaterials– A numerical study. Jason R. Dorvee (US Army ERDC, US Army Corps of Engineers, ERDC, 72 Lyme Rd, Hanover, NH 03755, Jason.R.Dorvee@usace.army.mil), Michelle E. Swearingen (US Army ERDC, Champaign, IL), Donald G. Albert, Michael Muhlestein (US Army ERDC, Hanover, NH), Megan A. Krieger (US Army ERDC, Champaign, IL), and James L. O'Daniel (US Army ERDC, Vicksburg, MS)

This numerical study is focused on the effects of disordered arrays as phononic metamaterials on the distortion of low-frequency impulses. Through numerical simulations we have been able to determine the characteristic properties of such phononic metamaterials as they correlate to the disruption a propagating impulsive wave. The fast 2-D FDTD model used allows for rapid reconfiguration of theoretical designs of: arrangements, geometries, and materials for the meta-atom; the results are informing evolving theory of the disruption of impulsive waves using metamaterials. With the ability to share CAD designs with the computational model, the layouts of arrangements from the numerical simulations can be ported to a 3D printer for the fabrication of physical models, for validation of the numerical model. These simulations serve as a planning tool for the configuration of sensors for both bench-scale and field testing. The model used and results of the numerical study are presented. [The work described in this document was funded under the US Army Basic Research Program under PE 61102, Project T22, Task 01 "Military Engineering Basic Research"; Task 02 "Material Modeling for Force Protection;" Project T25, Task 01 "Environmental Science Basic Research" and was managed and executed at the US Army ERDC.]

4:40

4pSA12. Distorting an impulse wave with phononic metamaterials—An experimental study. Michelle E. Swearingen (US Army ERDC, Construction Eng. Res. Laborartory, P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Jason R. Dorvee, Donald G. Albert, Michael Muhlestein (US Army ERDC, Hanover, NH), Megan A. Krieger (US Army ERDC, Champaign, IL), and James L. O'Daniel (US Army ERDC, Vicksburg, MS)

Bench-scale experiments were performed to characterize impulsive signal distortion with metamaterial arrays. As physical models, the composition of these systems rely on acoustically rigid materials in order to be examined at these small scales. In the final design the propagation of the pulse was evaluated using an array of receivers. Unique elements of this experimental methodology include: the design and refinement of an anechoic chamber for high frequencies, an in-lab tunable impulse source for both high frequencies and high amplitude, bench scale experimental designs achieved through the 3-D printed configurations of computer aided designs (CAD) extracted directly from a numerical model for exacting representations of configurations, and a measurement scheme designed to map the 3-D distortion of the impulse wave. Data and a description of the experiment and results are presented. [The work described in this document was funded under the US Army Basic Research Program under PE 61102, Project T22, Task 01 "Military Engineering Basic Research"; Task 02 "Material Modeling for Force Protection;" Project T25, Task 01 "Environmental Science Basic Research" and was managed and executed at the US Army ERDC.]

Session 4pSC

Speech Communication: Non-Native Speech Perception and Production (Poster Session)

Jessamyn L. Schertz, Chair
Centre for French and Linguistics, Dept. of Lang. Studies, University of Toronto Scarborough, Suite 301, Erindale Hall, 3359 Mississauga Rd., Mississauga, ON L5L1C6, Canada

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

4pSC1. The influence of talker accent strength on accent categorization. Taylor Federspill and Tessa Bent (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, tfedersp@indiana.edu)

Listeners' identification accuracy for nonnative-accented words tends to decrease as the talker's accent strength increases. However, the influence of accent strength on accent categorization has not been investigated. We hypothesized that nonnative talkers with stronger accents would have more salient cues to their native languages and thus, listeners would show increased accent categorization accuracy. To test this hypothesis, monolingual American English listeners categorized sentences produced by 12 native and 24 nonnative talkers in a 12-alternative forced-choice task with 8 nonnative accent categories and 4 regional dialect categories. Three talkers from each language/dialect background with different accent strengths (i.e., mild, moderate, strong) were included. The talkers' accent/dialect strengths were determined by 20 listeners' ratings on a 9-point scale (nonnative talkers: 1 = no foreign accent and 9 = very strong foreign accent; native talkers: 1 = Standard American English and 9 = very strong regional dialect). For the nonnative talkers, preliminary results show a moderate correlation between accent categorization accuracy and accent strength. However, within specific language backgrounds, adherence to the expected accuracy pattern varied substantially. Future analyses will investigate how accent strength influences confusion patterns across accents and dialects. [Work supported by an Indiana University Hutton Honors College grant.]

4pSC2. Acoustic correlates of L2 Spanish judgments of accentedness and comprehensibility: A mixed-effects modeling approach. Ziwei Zhou and Charles Nagle (Iowa State Univ., 2603 Kent Ave. # 209, Ames, IA 50010, ziweizh@iastate.edu)

Two constructs—comprehensibility and accentedness—figure themselves prominently in listeners' judgments of L2 (second language) speech. Correlational analyses have shown that they make separate contributions to such judgments (Derwing and Munro, 1995, 1997, 2005). As it is important to set realistic goals for adult L2 learners by prioritizing understanding over nativelikeness (Levis, 2005), recent studies began to examine the relative contribution of linguistic aspects, especially acoustic features, to comprehensibility and accentedness (Munro & Derwing, 1999; Kang, Rubin, & Pickering, 2010; Trofimovich & Issac, 2012). Consistent with this agenda, this study first examines the reliability of scores (produced by the 9-point Lickert scale) under modern measurement framework (e.g. multi-facet Rasch measurement model). Then, the relationship between acoustic features, including the confidence values (for each utterance) obtained using Google Cloud Platform speech recognition engine as well as suprasegmental features extracted using Prosogram (Mertens, 2014), and ratings are established using mixed-effects modeling techniques. Results indicate that, even though it can be shown that comprehensibility and accentedness are statistically distinct constructs, both ratings share a set of common acoustic correlates—speech time, standard deviation of pitch values, and normalized pairwise variability index (nPVI).

4pSC3. Word recognition for regional dialects and nonnative accents in children and adults: Influence of psycholinguistic distance. Rachael F. Holt (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, holt.339@osu.edu), Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), Miller E. Katherine, Mone Skratt Henry, Melissa M. Martin, Izabela A. Jamsek, and Donna E. Green (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

Few studies have directly compared adults' or children's perception of nonnative accents and unfamiliar regional dialects. However, some evidence suggests that nonnative varieties cause greater decrements in intelligibility and processing than unfamiliar native dialects, while metalinguistic awareness for nonnative varieties develops earlier than awareness for regional variants. To directly examine regional and nonnative accent perception, we tested sentence recognition in American-English monolingual 5- to 7-year-old children and adults for three accents: Central Midland (familiar native), Scottish (unfamiliar native), and German (detectable but mild nonnative accent) in quiet and multitalker babble. In quiet, both children and adults showed highly accurate word recognition for all accents. Although children's performance was lower than adults' in noise, overall word recognition accuracy patterns across accents were similar: accuracy was highest for the Midland talker, followed by the German-accented talker, and poorest for the Scottish talker. These data suggest that the greater decrements for nonnative accents compared to unfamiliar regional dialects previously reported may have arisen from the specific varieties or talkers selected. Although both types of unfamiliar speech can cause listening difficulty in noisy environments, the acoustic-phonetic distance from the home dialect may predict both adults' and children's performance better than native vs. nonnative status.

4pSC4. The effect of clear speech on listener perception of nonnative speech. Sarah H. Ferguson (Dept. of Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Alison Behrman (Speech-Language-Hearing Sci., Lehman College/City Univ. of New York, New Rochelle, NY), Jessica J. Staples, Stephanie M. Williams, and Paige E. Ihrig (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Listener perception of nonnative speech may be measured as intelligibility, ease of understanding, or accentedness. The relationship among these variables is complex, as is the influence of clear speech on these measures. The purpose of this study was to explore the effect on listener perception of clear speech produced by Spanish-accented talkers of American English.

Specifically, four questions guided this study: Does clear speech enhance intelligibility and ease of understanding? What is the effect of clear speech on accentedness? How does speaking style (conversational versus clear) influence the relationship among these perceptual variables? Does the presence or absence of supporting contextual information modify the relationship among intelligibility, ease of understanding, and accentedness? Twenty-eight female talkers of American English (half native English-speaking and half Spanish-accented) read aloud 56 short sentences using conversational and clear speech. The 56 sentences were comprised of 28 pairs, each containing a high- or low-probability final word. Young, normal-hearing, monolingual English listeners transcribed and rated ease of understanding for sentences presented in 12-talker babble, and then rated accentedness for sentences presented in quiet. Listeners assigned ratings using visual analog scales. Results and their implications for models of listener perception and clinical management of accentedness will be discussed.

4pSC5. Sentence recognition in noise: The interplay between talker intelligibility, familiarity, and linguistic complexity. Dorina Strori (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, dorina.strori@northwestern.edu), Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL), and Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Foreign-accented speech recognition is typically tested with linguistically simple sentences, which offer a limited window into speech processing. In this study, participants transcribed simple (i.e. mono-clausal, canonical declarative syntax) and complex (i.e. multi-clausal, non-canonical syntax, and/or passive voice) sentences in noise at two signal-to-noise ratios. The sentences were spoken by three talkers: one native, one high and one low intelligibility L2 English speaker. Unsurprisingly, recognition accuracy dropped with decreasing intelligibility and signal-to-noise ratio. Interestingly, accuracy dropped with increasing linguistic complexity only for the native and high intelligibility talkers. For the low intelligibility talker, sentence complexity did not affect recognition accuracy. This pattern of findings was present in mixed and blocked experimental designs, indicating no effect of familiarity with the talker. We argue that listeners employ qualitatively different speech processing strategies for low versus high intelligibility talkers, and suggest that this difference may be driven by the talker's realization of prosodic structure. While native and high intelligibility talkers effectively encode the contrast between complex and simple sentences, low intelligibility talkers may not convey sufficient prosodic information, resulting in the recognition of both sentence types as non-hierarchically organized word sequences. [Work supported by Knowles Hearing Center, NIH grants R01-DC012289 and R01-DC005794.]

4pSC6. Interference of adaptation to foreign-accented speech from consecutive but not interleaved training of two accents. Kali Woodruff Carr and Beverly Wright (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60201, KALI@U.NORTHWESTERN.EDU)

For perceptual learning on fine-grained discrimination tasks, improvement can be enhanced or disrupted when two tasks are trained, depending on how the training on those tasks is distributed. To investigate if this phenomenon extends to speech learning, we trained native-English speakers to transcribe both Mandarin- and Turkish-accented English sentences using one of three different configurations of the same training stimuli. After training, all trained groups performed better than untrained controls, and similarly to each other, on a novel talker of a trained accent (Mandarin). However, for a novel accent (Slovakian), performance was better than untrained controls when training alternated between the two accents, but not when the two accents were trained consecutively. Performance for the novel accent decreased as the number of contiguous sentences per accent during training increased. One interpretation of these results is that accent information is integrated during a restricted time window. If two accents are encountered within this window, information from both accents is integrated, yielding accent-general learning. If two accents are encountered in separate consecutive windows, accent-specific learning occurs for each accent and accent-general learning is prevented. These results mirror patterns for fine-grained discrimination learning, and illuminate the processes underlying the success of high-variability training.

4pSC7. The influence of lexical characteristics and talker accent on the recognition of English words by speakers of Korean. Minkyoung Choi (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Jeffrey J. Holliday (Dept. of Korean Lang. and Lit., Korea Univ., Bloomington, IN), and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

A growing body of evidence that the influence of phonological neighborhood density (PND) on speech perception increases over the course of second language (L2) learning (e.g., Bradlow *et al.*, 1999; Imai *et al.*, 2005; Yoneyama & Munson 2017). This was first shown by Imai *et al.* (2005), who found that only high-proficiency Spanish-speaking L2 English-language learners (ELL) showed an influence of English PND on speech perception; low-proficiency ELLs did not. More recently, Yoneyama and Munson (2017) examined whether Imai *et al.*'s findings could be replicated with ELLs whose L1 was Japanese. In contrast to Imai *et al.*, Yoneyama and Munson found that there were strong effects of PND on word recognition for all three groups of subjects: low-proficiency ELLs (tested in Japan), high-proficiency ELLs (tested in the US), and native speakers of English. The contrast between Imai *et al.* and Yoneyama and Munson's findings suggests that a wider set of studies using many different L1-L2 mismatches are needed to understand the range of PND effects in L2 learning. The current study examines Korean-speaking L2 learners of English using the same methods as Imai *et al.* and Yoneyama and Munson. Testing is ongoing in the US and Korea.

4pSC8. Generalization in foreign accent adaptation is predicted by similarity of segmental errors across talkers. Jordan Hosier (Northwestern Univ., 2016 Sheridan Rd, Evanston, IL 60208, jhosier@u.northwestern.edu), Bozena Pajak (Duolingo, Inc., Pittsburgh, PA), Ann Bradlow, and Klinton Bicknell (Northwestern Univ., Evanston, IL)

Comprehension of foreign-accented speech improves with exposure. Previous work demonstrates that listeners who adapt to accented talkers generalize that adaptation to other accented talkers—exposure to multiple talkers of the same accent facilitates comprehension of a novel talker of that accent (e.g. Bradlow and Bent, 2008) and exposure to multiple novel accents facilitates comprehension of yet another novel accent (Baese-Berk *et al.*, 2013). To examine possible theories of accent adaptation and generalization, we created a new dataset of phonetically transcribed accented speech produced by the training and test talkers used as stimuli in studies of accented speech generalization (Bradlow and Bent, 2008; Baese-Berk *et al.*, 2013). Using this dataset, we computed the (cosine) similarities between accented talkers in a multidimensional accent space defined by the rates at which talkers made different segment-level phonetic errors. Results show that similarity in this space accounts for all significant differences between training conditions in Bradlow and Bent (2008) and Baese-Berk *et al.* (2013): training conditions including more of the same types of phonetic errors as those of the test talkers led to better test talker comprehension. These results suggest that prior accent generalization results are compatible with simple, segment-error driven theories of adaptation.

4pSC9. Perception of non-native clear speech: The gap between speakers' effort and actual increase in intelligibility. Misaki Kato and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 161 Straub Hall, 1290 University of Oregon, Eugene, OR 97403, misaki@uoregon.edu)

Non-native speakers are less effective at increasing intelligibility in clear speech than native speakers. However, examining only intelligibility may not fully address how non-native clear speech is perceived by listeners. We examined native English listeners' perception of non-native English speakers' clear speech in terms of intelligibility, comprehensibility, and speakers' effort. Non-native English speakers produced English sentences in conversational and clear speaking styles. These productions were evaluated by 117 native English listeners. In an intelligibility task, listeners typed what they heard. In two rating tasks, listeners heard the same sentence produced by the same speaker in both a conversational and a clear speaking style. One group of listeners were asked to choose the sentence that was easier to understand (comprehensibility). Another group of listeners were asked to choose the sentence that the speaker was trying to say more clearly (speakers' effort). The results showed that conversational-clear

modifications did not result in increased intelligibility or comprehensibility. However, the rating task showed that listeners were sensitive to the speakers' effort in the clear speaking style. These results suggest that non-native speakers' effort to speak clearly may not necessarily result in intelligibility gains, but listeners may still be sensitive to this effort.

4pSC10. Cognitive effort in comprehending non-native speech decreases with listening experience. Dave C. Ogden (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, ogdend@umich.edu)

Listeners adapt to non-native accents, as shown by improved accuracy transcribing a speaker's intended utterance during testing (Baese-Berk *et al.*, 2013; Bradlow & Bent, 2008; Sidaras *et al.*, 2009). However, this improvement does not address whether cognitive effort decreases with accent experience, or whether accented speech continues to require effort to compensate for atypical features. The present study addresses this question using pupil dilation, a physiological measure indexing cognitive effort (Sirois & Brisson, 2014). In a matched-guise experiment, 48 participants heard 90 sentences spoken in either a General American Accent (GAA) or a non-native accent (NNA) while an eye tracker measured pupil diameter. An equal number of sentences with words predictable from context (e.g., "Sugar tastes very sweet.") and without (e.g., "It can't be brown.") were included. Participants transcribed (typed) each sentence. Pupil dilation while participants listened to each sentence was measured as the difference between average diameter for 100 ms immediately before and after the sentence played. Dilation decreased significantly more in the NNA than in the GAA condition, indicating that perception of accented speech becomes less effortful with experience. No effects of semantic predictability were observed. The relationship between effort and transcription accuracy will be discussed.

4pSC11. Vowel perception in quiet and noise for native and non-native listeners: Analysis of vowel identifiability. Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu), Su-Hyun Jin (California State Univ. at Northridge, Northridge, CA), and Sha Tao (Beijing Normal Univ., Beijing, China)

For a number of years, research on phonetic perception has been primarily focused on the percentage correct of phonetic identification (e.g., hit rate) without considering listeners' response (e.g., false alarm). The purpose of this study was to compute d' of English vowel identification in quiet and noise including both hit rate and false alarm for native and non-native listeners. Both data analyses showed native listeners outperformed non-native listeners for vowel identification in quiet and noise. However, the d' data showed that the non-naïve disadvantage became larger in long-term speech shaped (LTSS) noise than in quiet, while the percentage correct data showed an oppositely finding that the non-native disadvantage was smaller in LTSS noise than in quiet. These preliminary results suggest that false alarms should be included in the data analysis of phonetic perception and the idenitificability (e.g., d') based on the signal detection theory may be more appropriate to evaluate listeners' capacity to identify phonemes. [Work supported by China Natural Science Foundation 31628009 and University of Texas at Austin Research Grant].

4pSC12. Speech perception for native and non-native English speakers: Effects of contextual cues. Ling Zhong, Chang Liu (Commun. Sci. and Disorder, Univ. of Texas at Austin, 2504A Whitis Ave. Stop A1100 Austin, TX 78712, Austin, TX 78712, ling.zhong@utexas.edu), Cuicui Wang, and Sha Tao (Beijing Normal Univ., Beijing, China)

Previous studies have shown that native English speakers outperformed non-native English speakers in perceiving English speech under quiet and noisy listening conditions. The purpose of this study is to investigate the difference between native English speakers and native Chinese speakers on using contextual cues to perceive speech in quiet and multi-talker babble. Three types of sentences served as the speech stimuli in this study: sentences with high predictability (including both semantic cues and syntactic cues), sentences with low predictability (including syntactic cues), and sentences with zero predictability (consisting random sequences of words). These sentences were presented to native-English and native-Chinese listeners in quiet and four-talker babble with the signal-to-noise ratio at 0 and -5 dB. Preliminary results suggested that native Chinese speakers primarily rely on semantic information when perceiving speech in quiet, whereas native English speakers showed greater reliance on syntactic cues when perceiving speech in noisy situations. The difference between native English speaker and native Chinese speakers on syntactic, and semantic information utilization in various listening conditions will be discussed. [Work supported by The University of Texas at Austin, Undergraduate Research Fellowship and China National Natural Science Foundation 31628009]

4pSC13. Identifying /l/ and /n/ in English by multi-dialectal speakers of Mandarin. Bin Li (Linguist and Translation, City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon Tong 000, Hong Kong, binli2@cityu.edu.hk)

Native speakers of Standard Mandarin are mostly proficient in their local dialects which were acquired at home before or parallel to the Standard Mandarin. The Mandarin dialects that share general grammar mainly differ in their phonological features. A well-known example is the contrast of /l/ and /n/, which is not fully maintained or completely absent in many southern Mandarin dialects. This study investigated how perception of the English consonantal contrast may be influenced by experience in Mandarin dialects with varying phonemic status and distribution of the sounds. We examined the identification of the two consonants in various phonetic environments by Chinese bi-dialectal speakers who were all advanced learners of English-as-a-foreign-language. Results confirmed predicted perceptual difficulties across dialect groups as well as revealed other phonetic factors affecting the perceptual accuracy.

4pSC14. Speaker variability in spoken word recognition: Evidence from non-native listeners. Yu Zhang (Commun. Sci. and Disord., Oklahoma State Univ., 48 E Stimson Ave Apt 2, Athens, OH 45701, yu.zhang10@okstate.edu)

Non-native listeners face many challenges during spoken word recognition. On the lexical level, the vocabulary size of non-native listeners is usually reduced relative to native listeners. Lexical competition can arise from the lexicon in both the native and the non-native language during speech perception. How do non-native listeners resolve speaker variability, a challenge that is presumably non-lexical in spoken word recognition? This study aims to explore non-native spoken word recognition in the face of speaker variability. Mandarin speakers using English as a second or foreign language are recruited to listen to pairs of English words (e.g., cat—dog). Lexical decision time and accuracy on the second item in a pair (e.g., dog) are measured. The pairs are either spoken by the same or different native speakers of American English to examine the effect of speaker change on reaction time and accuracy in a short-term priming paradigm. Non-native listeners' pattern of response is explored as a function of the amount of priming received against speaker variability.

4pSC15. Speech learning in noise or quiet. Lin Mi (State Key Lab. of Cognit. Neurosci. and Learning and IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Dept. of Psychiatry, The Affiliated Brain Hospital of Guangzhou Medical University (Guangzhou Huiai Hospital), 36 Mingxin Rd., Liwan District, Guangzhou The Affiliated Brain Hospital of Guangzhou Medical University (Guangzhou Huiai, China, linmi83@hotmail.com), Sha Tao, Wenjing Wang, Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning and IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China), and Chang Liu (Dept. of Commun. Sci. and Disord., the Univ. of Texas at Austin, Austin, TX)

Noise made speech perception much more challenging for non-native listeners than for native listeners. We proposed that speech training in noise may better improve non-native listeners'speech perception. In the present study, three groups of native Chinese listeners were trained on English vowel perception under three conditions including quiet (TIQ), noise (TIN), and watching videos in English as an active control (C). Vowel identification in quiet and noise was assessed before, right after training, and three months later after training. Results showed that all listeners significantly

improved their vowel identification after training, and the training effect was retained three months later. Compared with the control group, the TIQ group improved more for vowel identification in quiet condition, but not for that in noise conditions. The TIN group improved vowel identification in noise significantly more than the TIQ and control groups. Compared with the TIQ, less informational masking and energetic masking was also found in the post-training test for the TIN than that for the TIQ. These results suggested that vowel perceptual training in noise may better improve L2 vowel perception in adverse acoustic environments than training in quiet. Implications for second language speech perception training were discussed.

4pSC16. Voice onset time and onset f0 as correlates of voicing in American learners of French. Amy Hutchinson and Olga Dmitrieva (Purdue Univ., 100 N. University St., West Lafayette, IN, hutchi25@purdue.edu)

Voice Onset Time (VOT) and onset f0 are known correlates of voicing distinctions in stops and both contribute to the perception of voicing (House & Fairbanks, 1953; Abramson & Lisker, 1965). The values of VOT and onset f0 which correspond to voicing categories vary cross-linguistically. Second language (L2) learners often have to acquire a novel use of these acoustic cues to produce and perceive L2 voicing. The acquisition of primary voicing cue, VOT, has been studied extensively in L2 research (Flege & Eefting, 1988; Flege 1991; Birdsong *et al.* 2007) but less is known about the acquisition of secondary cues. The present study compares the use of VOT and onset f0 in French and English speech produced by American learners of French (22). The study also examines the role of back transfer in L2 learners by comparing their English productions to a monolingual control group (33). The results demonstrate that although learners' VOT values in French were heavily influenced by English, their onset f0 production in both English and French was on target. Little evidence of learners' second language affecting their first language was found. Individual trends, including the effect of L2 proficiency level will also be explored.

4pSC17. First language phonetic drift in second language instructional environment. Olga Dmitrieva and Alexis N. Tews (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu)

Recent research demonstrates that second language learning can affect first language speech production. This has been shown for proficient long-term second language learners in immigration settings (Flege, 1987) as well as novice learners in the study abroad setting (Chang, 2013). The question addressed in the present study is whether learners acquiring a second language in a classroom setting in the first language-dominant environment are also subject to L1 phonetic drift. A group of American students enrolled in intermediate-level Russian language courses at a major Mid-Western university were recorded reading a list of Russian and English words designed to investigate the acoustic realization if word-initial and word-final stop voicing. Several acoustic correlates of initial and final voicing were measured (VOT, onset f0, preceding vowel duration, glottal pulsing, etc.) and compared to data from monolingual speakers of American English from the same geographic region. Results demonstrate that learners' realization of both initial and final voicing in English are affected by exposure to Russian. In the majority of measures, the drift occurred in the direction of norms characteristic of Russian voicing. Divergence from Russian was detected in the use of negative VOT for initial voiced stops.

4pSC18. The effect of distributional training on the production of non-native vowel contrasts. Grechen Go, Heather Campbell, and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, gretchen.go@nyu.edu)

The current study tests production accuracy of a non-native vowel contrast following a modified perceptual learning paradigm. Previous research has found that adults can learn non-native sound categories with sensitivity to distributional properties of their input. In the current study, 34 native-English adults heard stimuli from /o/-/oe/ continuum. Half of the participants heard stimuli drawn from a bimodal distribution and half from a unimodal distribution. To support learning, we incorporated active learning with feedback, lexical support in the form of two images, and overnight consolidation. Production of this contrast was assessed through a repetition task at baseline and at the end of the experiment. Production accuracy is measured as the Euclidean distance between /o/ and /oe/ at baseline and post-training. Preliminary analyses suggest that the distance between these two vowels increased for both groups, indicating that listeners in both conditions were able to transfer perceptual learning to production. These results suggest a way to mitigate the disadvantages previously found for participants in the unimodal condition, by incorporating active engagement with the target stimuli using lexical support, accuracy feedback, and overnight consolidation.

4pSC19. Disruption of learning on a non-native phonetic contrast by a brief practice break. David F. Little (Elec. and Comput. Eng., Johns Hopkins Univ., 1316 Eutaw Pl., #31, Baltimore, MD 21217, david.frank.little@gmail.com) and Beverly Wright (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

What is required for long-lasting learning of a non-native phonetic contrast? Do principles from fine-grained discrimination learning apply? Discrimination learning often requires extensive practice within a day for performance to improve across days, and a 30-minute break midway through practice can disrupt that learning. Thus it appears that trials must integrate to a learning threshold in a transient memory store in order for durable discrimination learning to occur. We asked whether this same principle applies in phonetic-contrast learning. To do so, we trained monolingual speakers on a non-native phonetic contrast along a voice-onset-time (VOT) continuum. Those who engaged in a single continuous practice session on day 1 reached above chance accuracy on day 2 while classifying the non-native phoneme. In contrast, when day-1 practice included a 30-minute break halfway through training, day-2 performance was at chance. These results suggest that for phoneme learning, as for discrimination learning, long-lasting learning requires that practice trials be integrated up to a learning threshold within a transient memory store before they are sent en masse into a memory that lasts across days.

4pSC20. Effect of L1 phonation contrast on production of L2 English stops. Jessamyn L. Schertz (Dept. of Lang. Studies, Univ. of Toronto Mississauga, Ste. 301, Erindale Hall, 3359 Mississauga Rd., Mississauga, ON L5L1C6, Canada, jessamyn.schertz@utoronto.ca)

This work examines the influence of native phonation contrast type on the production of English stops in different phonetic contexts/reading styles. Proficient English speakers from four different L1 language backgrounds produced words in three different contexts: words in isolation, phrase-final in a carrier sentence, and in a reading passage. Language backgrounds were representative of three types of phonation contrasts: Mandarin (aspiration contrast: [$p^h \sim p$]), Tagalog (voicing contrast: [$p \sim b$]), and Urdu (4-way phonation contrast [$p^h \sim p \sim b^h \sim b$]). 11 participants from each group were compared with a control group of L1 English speakers. Aspiration and closure voicing were measured. All groups produced English voiceless stops as aspirated; however, there was considerable variation in the voiced stops, with Mandarin speakers producing less voicing, and Urdu and Tagalog speakers producing more voicing, than English speakers across speech styles, showing an asymmetrical influence of L1 phonation on production of the English contrast, in line with other recent work.

4pSC21. The production of American English vowels by native Japanese speakers in two different conditions. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp)

Native Japanese speakers produced American English vowels /i, ɪ, ɛ, æ, ɑ, ʌ/ in two different conditions. First they read aloud one syllable words and non-words that contain these vowels on the word list. The vowels produced in this condition are believed to be what Japanese speakers believe typical of each English vowel. The Japanese speakers also reproduced English vowels uttered by native speakers. They were not told what vowel they would hear in each trial. The vowels produced in this way should be close to how each vowel sounds to the Japanese speakers, regardless of whether they correctly identify each vowel. The results revealed that in the "read aloud" conditions, /

i/ and /ɪ/ are spectrally close, and so are /æ/ and /ʌ/ and in low central position. /ɑ/ is higher than /æ/ and /ʌ/. In "reproduction" condition, on the other hand, /i/ and /ɪ/ are spectrally more distant, and /æ/ is fronted. /ɑ/ and /ʌ/ spectrally closer than in "read aloud" condition. Taken together, it can be assumed that there is a discrepancy between how Japanese speakers believe English vowels sound and how English vowels actually sound to them.

4pSC22. Cross-modality interference effects of phonetic and prosodic information in second language learners. Chieh Kao and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, kaoxx096@umn.edu)

Successful communication relies on efficient use of information from multiple modalities and dimensions. Our previous research using a cross-modal priming paradigm showed that congruent visual primes facilitate faster recognition of phonetic and emotional prosodic information in speech. Event-related potential (ERP) data further revealed distinct brain mechanisms in the N400 and late positive response (LPR) components for processing linguistic and paralinguistic congruency. The current study extended the same paradigm to English-as-a-second-language (ESL) learners to examine possible interference between the two informational dimensions as a function of language experience. Participants were sixteen normal adult ESL learners. Monosyllables /bab/ and /bib/ in a happy or angry tone were used as the auditory stimuli, and pictures of the speaker articulating vowel /a/ and /i/ with a happy or angry facial expression were used as the visual primes. Compared to native English speakers, ESL learners showed significantly longer reaction time with inconsistent congruency effects in both conditions. But their behavioral accuracy data mirrored those of the native speakers, and native-like N400 and LPR components were also reliably elicited in the ESL group in both conditions. Together, these results indicate stronger cross-modal interference between phonetic and emotional prosodic information in speech for second language learners.

4pSC23. Second-language learning in adolescents with cochlear implants. Deniz Baskent, Dorit Enja Jung (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen, Univ. of Groningen, PO Box 30.001, Groningen 9700RB, Netherlands, d.baskent@umcg.nl), Wander Lowie (Appl. Linguist, Ctr. for Lang. and Cognition, Univ. of Groningen, Groningen, Netherlands), and Anastasios Sarampalis (Psych., Univ. of Groningen, Faculty of Behavioural and Social Sci., Groningen, Netherlands)

Speech signals delivered via cochlear implants (CIs) lack spectro-temporal details, yet, young-implanted children can develop good native language skills (L1). This study explores three research questions: 1. Can adolescents with CIs learn a second language (L2)? 2. Is there a difference in spoken (auditory-A) vs. written (visual-V) L2 skills? 3. Which perceptual and cognitive factors influence L2 learning? Two groups (L1 = Dutch, age 12—17 years), one with normal hearing (NH) and one with CIs, and both learning English (L2) at school, participated. L1 and L2 proficiency was measured in receptive vocabulary (A), comprehension (A, V), and general proficiency (V). Further, basic auditory functioning, in temporal (gap detection) and spectral (spectral ripple detection) resolution, and cognitive functioning, in IQ, working memory, and attention, were measured. Preliminary data ($n = 7$ per group) indicated comparable L1 proficiency between NH and CI groups. While some CI users showed L2 proficiency within the NH range, on average, L2 proficiency was lower for the CI group. This effect was more pronounced for auditory tests. Reduced temporal and spectral resolution, but no difference in cognitive tests, were observed in CI group compared to NH, emphasizing the importance of auditory factors in L2 learning.

4pSC24. Production and perception of phonetically "non-native" clusters by Georgian native speakers. Harim Kwon (George Mason Univ., 5 rue Thomas Mann, CEDEX 13, Paris 75205, France, harim.kwon@univ-paris-diderot.fr) and Ioana Chitoran (Université Paris Diderot, Paris, France)

This study examines the role of native inter-consonant timing patterns in perception and production of word-initial consonant clusters, asking how phonotactically native clusters with non-native timing patterns are perceived and produced by speakers of a cluster-heavy language. We tested Georgian speakers in two experiments using CCV/CVCV stimuli produced by a French talker. Georgian is a cluster-heavy language with a relatively long inter-consonant lag, often resulting in transitional vowels between two consonants within a cluster. French onset clusters have shorter inter-consonant lag than Georgian ones. In Experiment 1 (shadowing), Georgian participants ($n = 14$) were exposed to French CCV/CVCV stimuli and asked to produce what they heard. In Experiment 2 (discrimination), the same Georgian participants heard the same French stimuli in AX pairs and determined whether the two sequences (A and X) were the same or different. The results from the two experiments suggest that Georgian participants often confused French CCV (without any vocalic transition) with French CVCV. The confusion occurred almost exclusively when the first vowel of CVCV was /ø/, which is acoustically similar to the transitional vowel in Georgian, but did not seem to stem from Georgian speakers' incapability of producing French short-lag CCV. We claim that native inter-consonant timing patterns influence perception and production of consonant clusters even when the clusters are phonotactically licit in one's native language.

4pSC25. Does it have to be correct?: The effect of uninformative feedback on non-native phone discrimination. Annie J. Olmstead and Navin Viswanathan (Speech, Lang., Hearing, Univ. of Kansas, 1000 Sunnyside Ave, Lawrences, KS 66045, annie.j.olmstead@ku.edu)

Learning the phonetic inventory of a non-native language requires perceptual adjustment to non-native phones that sometimes belong to a single category in the learner's native language. For example, English native speakers often struggle to learn the distinction between the Hindi phonemes [ʈ] and [t̪] that are both categorized as [t] in American English (AE). Olmstead and Viswanathan (2017) showed that AE listener's discrimination of these non-native phones could be improved using short exposure to naturally produced Indian English (IE) words that contained the target contrast. In the current study, we examine how feedback affects this lexical retuning effect. Specifically, we set up feedback schedules that either reinforce the consistent mapping of these consonants onto the AE speaker's existing [t] and [θ], or that reinforce an inconsistent mapping. If the consistency of this mapping in IE is paramount to improving phonetic discrimination, then reinforcing it should strengthen the effect and providing a variable mapping should weaken it. Results suggest that the feedback given to participants did not change the effect on discrimination—discrimination improved whether or not the feedback was consistent. Implications of these findings are discussed.

4pSC26. Using high variability phonetic training to train non-tonal listeners with no musical background to perceive lexical tones. Alif Silpachai (Appl. Linguist and Technol., Iowa State Univ., 527 Farmhouse Ln., Ames, IA 50011, alif@iastate.edu)

Previous research has not extensively investigated whether High Variability Phonetic Training (HVPT) is effective in training listeners with no musical background and no prior experience with a tone language in their identification of non-native lexical tones. In this study, it was investigated whether HVPT is applicable to the acquisition of non-native tones by such listeners. Twenty-one speakers of American English were trained in eight sessions using the HVPT approach to identify Mandarin tones in monosyllabic words. Ten of the participants were exposed to words produced by multiple talkers (MT condition), and eleven participants were exposed to words produced by a single talker (ST condition). The listeners' identification accuracy revealed an average 44% increase from the pretest to the posttest for the MT condition and an average 30% increase for the ST condition. The improvement also generalized to new monosyllabic words produced by a familiar talker and those produced by two unfamiliar talkers. The learning however did not generalize to novel disyllabic words produced either by a familiar talker or an unfamiliar talker. Comparisons between two groups further revealed that the improvement of the listeners in the MT condition was significantly higher than that of the listeners in the ST condition.

4pSC27. Perceiving allophonic alternations in an immersion setting: The case of alveolars and rhotics in Spanish and English. Farrah A. Neumann (Linguist, Univ. of Pittsburgh, 4200 5th Ave, Pittsburgh, PA 15260, fan9@pitt.edu)

Recent research indicates that second language (L2) immersion influences the phonetic production not only of the L2 (Jacobs, Fricke, & Kroll, 2016), but also of the first language (L1) (Chang, 2012). However, less understood is the immediate impact of L2 immersion on cross-language influence in phonetic *perception*. This study tracked L2 Spanish study abroad learners' perceptual shifts due to immersion using auditory-visual cross-modal priming experiments in both Spanish and English. The target phones, [r, ɾ, t, d, ð, ɹ], were chosen for their differing phonemic and allophonic statuses in Spanish and English. For example, Spanish rhotics include the phonemic trilled /r/ and tapped /ɾ/, whereas the tapped /ɾ/ exists phonetically in English, but as an allophone of the alveolar stops /t d/. Because learners must adjust their phonological inventories to accommodate the L2 statuses of these phones, we expect their L1 inventories to also reflect these changes, as evidenced by slower RTs for the prime-target pairs that do not correspond to the L2. This study provides a fuller perspective on how immersion mediates L1-L2 interference, and offers insight as to whether perceptual changes in the two languages are necessarily related. Data analyses are currently underway.

4pSC28. Language experience outweighs perceptual enhancement in category learning. Meng Yang (Linguist, Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, mengyang@ucla.edu)

Some theorists claim that cues that contribute to the same acoustic percept are naturally coupled in category learning. In contrast, others claim that no pairs of cues are privileged in this way, and listeners simply learn cue correlations from language input. We compared the cue weighting and cue-shifting between pitch and breathiness, which are enhancing but do not signal a phonemic contrast in English (Experiment I), and pitch and closure duration, which are not enhancing but are correlated in signaling the English voicing contrast (Experiment II). When the cues were enhancing but not contrastive, English listeners successfully learned to weight the more informative cue higher. In contrast, preliminary results show that listeners have difficulty weighting the informative cue higher when the two cues are not enhancing but only contrastive. Results suggest that language experience outweighs auditory enhancement relationships in the perceptual coupling of cues.

THURSDAY AFTERNOON, 10 MAY 2018 GREENWAY H/I, 1:00 P.M. TO 4:30 P.M.

Session 4pSP

Signal Processing in Acoustics: Topics in Signal Processing in Acoustics

Ryan L. Harne, Cochair
Mechanical and Aerospace Engineering, The Ohio State University, 201 W 19th Ave, E540 Scott Lab, Columbus, OH 43210

Buye Xue, Cochair
Starkey, 6600 Washington Avenue S., Eden Prairie, MN 55344

Contributed Papers

1:00

4pSP1. Analysis of foldable acoustic arrays from piecewise linear, conformal, and tessellated topologies. Chengzhe Zou and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave, Columbus, OH 43210, zou.258@osu.edu)

Arrays of acoustic sources distributed over the tessellated frameworks of origami-inspired folding patterns demonstrate exceptional and straightforward adaptivity of acoustic wave guiding architecture to contrast to the challenges manifest in digital wave guiding control methods. Yet, when the concept of folding arrays is applied to tessellations formed by single patterns, acoustic fields with distinct distributions of energy delivery cannot be realized. To overcome this limitation, here multiple folding patterns are assembled by piecewise geometries that permit the array to conform to unique topologies when reconfigured by physical folding actions. An analytical model is developed to predict the acoustic pressure radiated from the acoustic sources arranged on piecewise-continuous surfaces, while experiments are conducted to demonstrate the concept and validate the model. Parametric investigations are then undertaken to uncover the relations of modularity and folding of the acoustic array and the resulting acoustic energy delivery and distributions. The results find that the assembly of tessellated arrays from multiple distinct folding pattern sub-elements may replicate the acoustic wave guiding of complex acoustic sources that traditionally do not enable adaptive wave radiation properties.

1:15

4pSP2. Cross-correlation method for acoustic detection of small unmanned aerial vehicles. Alexander Sedunov, Hady Salloum, Alexander Sutin, and Nikolay Sedunov (Maritime Security Ctr., Stevens Inst. of Technol., One Castle Point on Hudson, Stevens Inst. of Technol., Hoboken, NJ 07030, asedunov@stevens.edu)

The availability of Unmanned Aerial System (UAS) to consumers has increased in the recent years, with it came the potential for negligent or nefarious misuse of them. Stevens Institute of Technology has built a passive acoustic system for low flying aircraft detection, the application of the developed principles and algorithms for UAS acoustic detection and tracking is presented in this paper. The application of the developed principles and algorithms for UAS acoustic detection and tracking is presented in this paper. Several experiments were conducted aiming to establish the characteristics of the emitted noise of UAVs of various sizes while airborne and demonstrate the processing required to detect and find the direction toward the source. The vehicles tested included popular quadrotors: DJI Phantom 2 Vision+, 3DR Solo, DJI Inspire 1 as well as larger semi-professional vehicles: Freefly Alta 6, DJI S1000. The small array of 16 microphones was used for data collection in the tests near local NJ airport. Acoustic signatures of the tested UAS were collected for stationary and flying UAS. We applied the algorithm for detection and direction finding based on fusing time difference of arrival (TDOA) estimates computed by finding peaks in the output

Generalized Cross-Correlation (GCC) function. The cross-correlation signal process provided UAS detection and bearing for distances up to 250m while the spectrograms did not reveal acoustic UAS signatures at that distance. This work is being supported by DHS's S&T Directorate.

1:30

4pSP3. Cross-term analysis in frequency-difference-based source localization methods. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave, 2010 Automotive Lab, Ann Arbor, MI 48104, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

In previous work, it has been shown that a quadratic product of frequency-domain acoustic fields at different frequencies but the same spatial location leads to an auxiliary field which may contain field information at frequencies below the original signal's bandwidth (Worthmann, Song and Dowling, 2015, JASA 138, 3549-3562). This quadratic product, termed the frequency-difference autoproduct, has been shown to be valuable for beamforming and source localization in the presence of environmental mismatch and/or array sparseness. However, in a multipath environment, this quadratic product leads to undesired cross-terms. Bandwidth averaging procedures have been found to suppress some of their detrimental influences in some cases, but not all. Additionally, the poor dynamic range observed in frequency-difference beamforming and frequency-difference matched field processing are associated with the imperfect mitigation of these cross-terms. In this presentation, the nature of these cross-terms is analyzed, and signal processing tools are developed which attempt to robustly mitigate the detrimental effects of these cross terms. These signal processing tools can be used to potentially improve localization performance when using frequency-difference autoproduct-based source localization schemes. [Sponsored by ONR and NSF]

1:45

4pSP4. Investigation of reconfigurable antennas by foldable, e-textile tessellations: Modeling and experimentation. Chengzhe Zou (Mech. and Aerosp. Eng., The Ohio State Univ., Columbus, OH), Shreyas Chaudhari, Saad Alharbi, Hamil Shah, Asimina Kiourti (Elec. and Comput. Eng., The Ohio State Univ., Columbus, OH), and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave, E540 Scott Lab, Columbus, OH 43210, harne.3@osu.edu)

Physical deformation mechanisms are emerging as compelling and simple ways to adapt wave propagation properties of antenna arrays in contrast to digital steering approaches acting on topologically fixed antennas. Concepts of physical reconfigurability also enable exceptional capabilities such as deployable and morphing antenna arrays that serve multiple functions and permit compact transport with ease. Yet, the emergent concepts lack broad understanding of effective approaches to integrate conformal, electrically conductive architectures with high-compliance foldable frameworks. To explore this essential interface where electrical demands and mechanical requirements may conflict, this research studies e-textile-based reconfigurable antenna arrays that conform to adaptable topologies by origami-inspired tessellations. The e-textiles leverage embroidered conductive threads along frameworks established on origami tessellations to permit large compliance at folding edges as needed while retaining the desired electromagnetic wave propagation characteristics. Computational modeling is used to guide experimental fabrications and validations. It is found that e-textile origami antenna arrays permit significant adaptation of wave radiation while maintaining the required mechanical robustness under folding sequences. These findings may motivate future concepts for reconfigurable antennas established upon physical deformation processes.

2:00

4pSP5. Mathematical properties of generalized Bessel functions and application to multi-tone sinusoidal frequency modulation. Parker Kuklinski (Naval Undersea Warfare Ctr., 1176 Howell St, Newport, RI 02841, parker.s.kuklinski1@navy.mil) and David A. Hague (Naval Undersea Warfare Ctr., North Dartmouth, Massachusetts)

The generalized Bessel function (GBF) is a multi-dimensional extension of the standard Bessel function. Two dimensional GBFs have been studied extensively in the literature and have found application in laser physics, crystallography, and electromagnetics. However, a more rigorous treatment of higher-dimensional GBFs is lacking. The GBF exhibits a rich array mathematical structure in regards to its partial differential equation representation, its asymptotic characterization, and its level sets. In this talk/paper, we explore these properties and connect spectral and ambiguity function optimization of a multi-tone SFM signal to finding the location of the roots of these generalized Bessel functions.

2:15

4pSP6. Consensus detection of a Gaussian underwater acoustic source by a distributed sensor network. Thomas J. Hayward and Steven I. Finette (Naval Res. Lab., 4555 Overlook Ave SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Probabilistic data fusion is a well-established algorithmic approach to the detection of underwater acoustic sources by a distributed sensor network. Direct communication of all of the network nodes with a fusion center would provide optimal joint detection via summation of log-likelihood ratios (LLR's) across the nodes. However, a network that relies on a single fusion center for data exfiltration is not robust to the loss of the fusion center. Distributed detection mediated by dynamic consensus algorithms, which have their roots in the broader system-theoretic discipline of dynamic state estimation, offers an alternative that is robust to node loss. The present work illustrates the application of dynamic consensus algorithms to the distributed detection of underwater acoustic sources, taking as an example the detection of a stationary, ergodic Gaussian source in a shallow-water waveguide by a network of vertical-array nodes. A general definition of consensus algorithms is followed by the construction of particular linear consensus dynamical systems that (1) provide asymptotic agreement of the time-averaged consensus LLR with the optimal central-fusion LLR; and (2) equalize the mean cross-node disagreements of the consensus LLR's. Performance of centralized and consensus detection is compared in simulations. [Work supported by ONR.]

2:30–2:45 Break

2:45

4pSP7. Integration of signal processing modules of hearing aids as real-time smartphone apps. Nasser Kehtarnavaz and Linda Thibodeau (Univ. of Texas at Dallas, 800 West Campbell Rd., Richardson, TX 75080, kehtar@utdallas.edu)

Our research group has been working on developing an open source research platform for hearing improvement studies based on smartphones noting that currently no open source, programmable, and mobile research platform exists in the public domain for carrying out hearing improvement studies. Such a platform allows smartphones, which are already widely possessed and carried by people, to be used for the development and field testing of existing or new hearing improvement signal processing algorithms. A number of smartphones apps have already been developed by our research group for specific or individual modules of the speech processing pipeline of a typical digital hearing aid. This work covers the integration of several individual signal processing modules into an integrated smartphone app. The steps taken in order to enable the individual modules to run together in real-time and with low audio latency will be presented and a demo of the integrated app running on both iOS and Android smartphones will be shown.

3:00

4pSP8. Stereo I/O framework for audio signal processing on android platforms. Abdullah Küçük, Yiya Hao (Elec. and Comput. Eng., The Univ. of Texas at Dallas, 17878 Preston Rd, APT 369, Dallas, TX 75252, axk166230@utdallas.edu), Anshuman Ganguly, and Issa M. Panahi (Elec. and Comput. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Smartphones and tablets are attractive cost-effective solutions for implementing multi-channel audio signal processing algorithms. Modern smartphones and tablets have at least two microphones and requisite processing capabilities to perform real-time audio signal processing. However, it is still challenging to integrate the dual-microphone audio capture and the audio

processing pipeline in real-time for efficient operation. In this paper, we propose an efficient stereo input/output framework for real-time audio signal processing on Android smartphones/tablets. This framework enables us to perform a large variety of real-time signal-processing applications on Android platform such as Direction of Arrival (DOA) estimation, Audio Compression, and Multi-Channel Speech Enhancement. The implementations and demos of Direction of Arrival (DOA), Audio Compression, and Multi-Channel Speech Enhancement algorithms are also presented to show the effectiveness of proposed framework.

3:15

4pSP9. Robust real-time implementation of adaptive feedback cancellation using noise injection algorithm on smartphone. Parth Mishra (Elec. and Comput. Eng., Univ. of Texas at Dallas, 7740 mccallum blvd, apt#315, DALLAS, TX 75252, pxm161830@utdallas.edu), Serkan Tokgoz, and Issa M. Panahi (Elec. and Comput. Eng., Univ. of Texas at Dallas, Richardson, TX)

We propose a novel low latency smartphone-based application that demonstrates the real-time operation to cancel the negative effects of acoustic feedback arising from the coupling between the speaker and the microphone of the smartphone or similar device utilizing the robust Noise Injection (NI) method. We make use of multiple noise injections of short durations to estimate the filter coefficients of an appropriate order between the speaker and the microphone, in order to perform the feedback cancellation effectively in real-time. Our motive behind the development of this application is to perform an effective acoustic feedback cancellation irrespective of the position of speaker and the microphone on the platform under consideration. With the proposed application, we can estimate the transfer function between speaker and microphone in the changing room acoustics making the feedback cancellation very effective. Objective and subjective tests were conducted and results of the proposed real-time application indicate significant acoustic feedback suppression in the presence of varying environmental conditions.

3:30

4pSP10. Comparison of processing speed between Arduino Due and Teensy 3.6 boards in a system identification application. Artur Zorzo (Acoust., UFSM, SQN 315 BLG APTO 205, ASA NORTE, BRASILIA 70774070, Brazil, arturzorzo@hotmail.com), William D. Fonseca, Paulo Mareze, and Eric Brandao (Acoust., UFSM, Santa Maria, RS, Brazil)

The system identification technique is a powerful real-time signal processing tool when dealing with the characterization of dynamic systems. This approach uses adaptive filtering techniques and the least mean square (LMS) algorithm to estimate the impulse response of a system in a finite impulse response (FIR) filter format. The main purpose of this paper is to compare the results and processing speed of two ARM-based microcontrollers as well as to give a theoretical background and an implementation procedure of the technique. ARM (Advanced RISC Machine) is an architecture of 32-bit processors usually applied in embedded systems due to its processing power, relatively low consumption, cost and small size. The impulse responses obtained by the method for several passive electronic filters were transformed to the frequency domain and compared with simulations of the circuits' frequency responses. This practice was repeated for both boards and for different parameters of the algorithm. The results have shown that the Arduino Due was unable to deal with higher sampling frequencies due to its processor limitations. On the other hand, the Teensy 3.6, a more powerful controller, yields great results even for higher frequencies.

3:45

4pSP11. Calibration of low-cost sound level meters using machine learning techniques. Jose Giraldo (AAC centro de acústica aplicada, circular 74b 39b-71, Medellin 050031, Colombia, jose091@gmail.com) and Alberto Bañuelos Irusta (AAC centro de acústica aplicada, Vitoria, Spain)

The assessment of environmental noise pollution from complex sources that are not represented in traditional noise maps requires a long term monitoring network to manage citizen complaints, a network of low-cost sound level meters is a practical option to analyze real-world cases, however the accuracy of the measurements is a concern when this information is used to evaluate regulation accomplishment. Previous work on low-cost monitoring devices has been carried out in microphone comparison, processing board selection and recently in MEMS microphone use. In the present work the use of reinforcement learning techniques is explored to calibrate the sound pressure levels generated by a low-cost monitoring device using a class one sound level meter as reference in a continuous measurement setup. Machine learning based models are known to take into account strong non-linearities which are effective to get an alternative calibration method for low-cost sensors.

4:00

4pSP12. Improved pre-filtering stages for GCC-based direction of arrival estimation using smartphone. Abdullah Küçük (Elec. and Comput. Eng., The Univ. of Texas at Dallas, 17878 Preston Rd, APT 369, Dallas, TX 75252, axk166230@utdallas.edu), Anshuman Ganguly, and Issa M. Panahi (Elec. and Comput. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Sound source localization (SSL) is one of the important areas in signal processing especially for hearing aid applications. Having at least two microphones and powerful processing capability makes smartphone as a good sound source locator for people who have hearing impairment. In our previous work, we showed that traditional Generalized Cross Correlation (GCC) method with spatial post-filtering stage would increase the performance of instantaneous estimation of direction of arrival (DOA). In this paper, we combine a spectral pre-filtering stage with our previous work to improve the noise robustness of the earlier method. By combining pre-filtering stage with GCC and post-filtering stage, we obtain robust DOA estimation under various realistic noise types. We experiment with several pre-filtering stages and study their impact on reducing root mean square error (RMSE) and mean absolute error (MAE) for DOA estimation. The real-time implementation of some of the proposed algorithms on smartphone is also presented.

4:15

4pSP13. Speech enhancement and speaker verification with convolutional neural networks. Peter Guzewich and Stephen Zahorian (Binghamton Univ., PO Box 6000, Binghamton, NY 13902, zahorian@binghamton.edu)

Convolutional neural networks have been used with great success for image processing tasks. The convolution filters can act as powerful feature detectors in images to find edges and shapes. Time domain speech signals are one dimensional, but frequency domain information can be viewed in the same way as an image, allowing one to use two-dimensional convolutional networks for tasks such as speech enhancement. In this paper, recently reported deep neural network preprocessing methods used for dereverberation are examined to explore their ability to remove noise and or reverberation. A convolutional network is then introduced and compared to the previous methods to explore the relative strengths and weaknesses of each approach. Comparisons are made through experimentation in the context of speech quality and speaker identification tasks.

Session 4pUW

Underwater Acoustics: Underwater Acoustic Communications, Positioning, and Signal Processing

Aijun Song, Chair
Electrical and Computer Engineering, University of Alabama, 401 7th Ave, Hardaway Rm284, Tuscaloosa, AL 35487

Contributed Papers

1:00

4pUW1. Unsupervised seabed characterization with Bayesian modeling of SAS imagery. T. S. Brandes and Brett Ballard (BAE Systems, Inc., 4721 Emperor Blvd., ste 330, Durham, NC 27703, tsbrandes@gmail.com)

Seabed characterization has utility for numerous applications that seek to explore and interact with the seafloor, ranging from costal habitat monitoring and sub-bottom profiling to man-made object detection. In the work presented here, we characterize the seabed based on the texture patterns within SAS images constructed from high-frequency side-scan sonar. Features are measured from the SAS images (e.g., lacunarity, an established texture feature coding method, and a rotationally invariant histogram of oriented gradients). Based on these SAS image features, we perform unsupervised clustering with a hierarchical Bayesian model, which creates categories of seabed textures. Our clustering model is a new variant of the hierarchical Dirichlet process that is both adaptive to changes in the seabed and processes batches of SAS imagery in an online fashion. This allows the model to learn new seabed types as they are encountered and provides usable clustering results after each batch is processed, rather than having to wait for all the data to be collected before being processed. The model's performance of seabed characterization by SAS image texture is demonstrated in the overall range and internal consistency of textures specific to each learned cluster. [Work supported by the Office of Naval Research.]

1:15

4pUW2. Use of adaptive thresholding to improve accuracy of small-aperture acoustic localization in shallow water. Paul Murphy (Univ. of Washington, Mech. Eng., UW Mailbox 352600, Seattle, WA 98195-2600, pgmurphy@uw.edu) and Brian Polagye (Univ. of Washington, Seattke, WA)

Passive acoustic localizing arrays are utilized in shallow water for applications ranging from environmental monitoring to marine security. Although large-aperture hydrophone arrays can provide accurate estimates of source position, they can be costly to deploy and maintain. Small-aperture arrays can be used to estimate the incident azimuth and elevation angles of acoustic sources, but their effectiveness may be hindered by multipath interference when deployed in reverberant environments. Here, a small-aperture hydrophone array is used to estimate the direction of acoustic sources with center frequencies of 69 and 120 kHz and durations less than 5 ms which were deployed at ranges of up to 75 m in a bay with a water depth on the order of 10 m. Time-differences of arrival are determined using an adaptive thresholding algorithm, which is shown to have superior performance to a cross-correlation method due to the prevalence of multipath interference in the signals. Errors in source bearing estimates and in the underlying time-differences of arrival calculations are evaluated by comparing to ground-truth values generated from the GPS position of the source.

1:30

4pUW3. Time delay estimation via zero-crossing of envelope. Donghwan Jung and Jea Soo kim (Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, Busan 49112, South Korea, wjdehdghkss@gmail.com)

Time-delay estimation of an acoustic signal in an ocean waveguide is a challenging due to the received signal corrupted by noise and phase distortion caused commonly by reflection of the transmitted signal traveling to a remote receiver via more than one path. In this study, envelope-based Hilbert transform method is applied to enhance the accuracy of time-delay estimation. Through the mathematical interpretation and numerical simulations, the proposed method is compared to the conventional time-delay estimation methods (e.g., cross correlation and zero-crossing of Hilbert transform). It is found that the envelope-based Hilbert transform method more efficiently estimates the time-delay than the conventional methods and resolve ambiguity of time delay estimate.

1:45

4pUW4. Tracking and classification of targets detected by a moving multibeam sonar. Emma D. Cotter (Mech. Eng., Univ. of Washington, 3900 E Stevens Way NE, Seattle, WA 98195, ecotter@uw.edu), James Joslin (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Brian Polagye (Mech. Eng., Univ. of Washington, Seattke, WA)

There are multiple applications where tracking and classification of targets detected in multibeam sonar data is required, ranging from environmental studies to marine security. Target tracking from a fixed frame of reference is well-established, but if the sonar is mounted to a surface-following platform or vessel, the tracking task is complicated by the moving frame of reference. Here, accelerometer data is integrated into a Kalman filter-based tracking scheme to correct for the motion of a non-stationary sonar. This technique is first verified using a vessel-mounted sonar and target with known position. Tracking of marine animals from a moving platform is then demonstrated in real-time for the case of a sonar mounted on the hull of a wave energy converter. These target tracks are classified in real time (e.g., diving bird, seal) using a Random Forest algorithm. The effectiveness and limitations of real-time target tracking and classification from a moving multibeam sonar are discussed.

2:00

4pUW5. Time reversal based frequency-domain equalization for underwater acoustic single-carrier transmissions. Xingbin Tu, Aijun Song (Elec. and Comput. Eng., Univ. of Alabama, 401 7th Ave, Hardaway Rm-284, Tuscaloosa, AL 35487, song@eng.ua.edu), and Xiaomei Xu (Xiamen Univ., Xiamen, Fujian, China)

Time reversal processing generates a time-focused signal, after combining dispersive reception from a hydrophone array in the underwater acoustic

channel. Because of this property, the time reversal method has been studied as a preprocessing scheme to support decision feedback equalization for underwater acoustic communications. Orthogonal frequency division multiplexing has also been combined with time reversal processing to improve spectral efficiency, as the equivalent compact channel impulse response allows short prefixes in the acoustic channel. Here, we show that time reversal processing can be used to facilitate prefix-free frequency domain equalization. Traditional frequency domain equalization schemes rely on the use of periodic prefixes, which bring a loss of spectral efficiency in the already-bandlimited underwater channel. The core idea is to use time reversal preprocessing to generate a delta-like equivalent impulse response to facilitate robust reconstruction of prefixes at the receiver for prefix-free single-carrier transmissions. The resultant receiver not only improves the spectral efficiency, but also supports communications over channels of variable fluctuation rates. The effectiveness of the receiver has been tested using the field measurements from the Gulf of Mexico in August 2016.

2:15

4pUW6. M-ary cyclic shift keying spread spectrum underwater acoustic communication using frequency domain energy detector. zhuoqun wei, Xiao Han, and Jingwei Yin (College of Underwater Acoust. Eng., Harbin Eegineering Univ., 145 Nantong St., Harbin, Heilongjiang Province, Harbin, Heilongjiang 150001, China, weizhuoqun@hrbeu.edu.cn)

M-ary cyclic shift keying (M-CSK) method can greatly improve the communication rate of conventional direct sequence spread spectrum, but the robustness is seriously influenced by the fluctuation of sea surface and Doppler effect. As to this problem, M-ary cyclic shift keying spread spectrum (M-CSKSS) underwater acoustic communication (UAC) based on frequency domain energy detector (FDED) algorithm is proposed in this paper. Time reversal (TR) is firstly adopted to suppress multipath spread using channel impulse response (CIR) estimated from the former symbol. And then the baseband signal after TR processing goes through a FDED, decisions are made according to the output energy later. Simulation results show the robustness of this algorithm under low signal-to-noise ration and multipath environment. In a field experiment, data transmission with very low bit error rate (BER) was achieved and the communication rate was 375bits/s.

2:30–2:45 Break

2:45

4pUW7. A method of distinguishing between surface and underwater targets based on time delay in deep water. Chunpeng Zhao and Guolong Liang (Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St.,Nangang District,Harbin,Heilongjiang Province,China, Harbin, Heilongjiang 150001, China, 1067283316@qq.com)

It is difficult to distinguish between surface and underwater targets in passive detecting with a single sensor or a horizontal linear array in deep water. It is noticed that targets in different depths have different characteristics in time delay difference of rays. This paper takes advantage of this feature to discriminate targets' depths in direct-arrival zone and shadow zone in deep water. And the relation between judgment boundary and depth of sensor is analyzed. The simulation result shows that when the depth of sensor is 50m, the boundary which can distinguish between surface and underwater targets can be set to 10m. And the depth of the boundary decreases with the increase of the depth of sensor.

3:00

4pUW8. Research on integrated positioning approach based on long/ultra-short baseline. Jingfei Wang, Nan Zou, and Fu Jin (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, 527656530@qq.com)

With the development of underwater acoustic technology, the combination of multiple positioning means becomes the trend of underwater acoustic positioning technology. Based on an integrated long/ultra-short baseline location equipment, this paper discusses a multi-solution fusion method based on the minimum mean square error criterion. This method fuses long baseline and ultra-short baseline information on output, and it is relatively simple. The results of integrated long/ultra-short baseline location are compared with the results of long baseline location and ultra-short baseline location by simulation and experience. It shows that the integrated long/ultra-short baseline location system has the higher positioning accuracy than long baseline location and ultra-short baseline location. It realizes the high precision positioning of underwater target.

3:15

4pUW9. Study of long baseline optimization positioning algorithms Bbsed on redundancy measurement. Ying Guo, Yan Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 15000, China, guoyinggy@hrbeu.edu.cn), and Nan Zou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Generally, underwater vehicles real-time navigation system works through the principle of spherical intersection. The number of array elements required for this method is fixed, but in the case of redundant array elements, they can't be fully utilized. This paper presents two optimization methods based on redundancy measurement. One approach is the steepest descent method and the other is Gauss-Newton method. The former searches for the minimum value of the measurement's sum of squared residuals along the negative gradient direction, and the later, a non-linear least-squares optimization method that constantly revise regression coefficients through multiple iterations to minimize the sum of squared residuals. Simulation show that both methods are greatly affected by the initial position, the closer the distance between initial position to the target is, the smaller positioning error is. On the same initial position, the Gauss-Newton method has less positioning error than the steepest descent method. The conclusions have been proved to be valid by sea trial.

3:30

4pUW10. Direction estimation of coherent signals based on the symmetry of uniform linear array. Yunqi Tian, Guolong Liang, and Fu Jin (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St.,Harbin, Heilongjiang Province, Harbin, Heilongjiang 150001, China, tianyunqi@yahoo.com)

A large number of coherent signals are generated due to interface reflection in the ocean. A novel algorithm based on the symmetry of uniform linear array (ULA) is proposed to solve the problem of direction of arrival (DOA) estimation in coherent environment. Considering the center sensor as the reference, ULA can be divided into two symmetric parts. Firstly, the conjugate data of one part is rearranged, meanwhile, the other part data is maintained. Secondly, the correlation function calculated by processed data is averaged to construct a full rank Hermitian Toeplitz matrix. Finally, we can estimate the directions of coherent sources by eigen-decomposition. The results of simulation show that the proposed algorithm not only can effectively distinguish coherent signals, but also can improve the resolution, the accuracy and the probability of success in DOA estimation. Furthermore, the proposed algorithm is robust in the case of lower SNR or fewer snapshots.

3:45

4pUW11. An effective method of choosing vertical beam pitch angle of sonar array. Mengnan He, Fan Zhan, Guolong Liang, and Ao Shen (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St.,Nangang District, Harbin, Heilongjiang 150001, China, hmn111@126.com)

In order to obtain a better array gain, the sonar main-beam should point to the incident direction of acoustic waves. For stereoscopic sonars, such as cylindrical array sonar, the incident direction of acoustic wave contains both horizontal angle and vertical pitch angle (VPA). Due to the variation of acoustic VPA caused by hydrological condition, the array gain would decrease seriously when the VPA of sonar beam doesn't match the VPA of acoustic waves. To solve the above problem, we proposed a VPA optimization method for stereoscopic sonars. First,we analyzed the variation rule of both sonar array gain and the acoustic VPA. Then,search for the VPA of the sonar beam in the vertical directions by beam scanning, so as to maximize the array gain. Simulations results are provided to validate the performance of the proposed method.

4:00

4pUW12. Methods of suppressing tow ship noise with a horizontal linear array. Jia Feng, Nan Zou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China), Yan Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 903, NO. 145, Nantong St., Harbin, China, wangyan@hrbeu.edu.cn), and Yu Hao (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Tow ship noise seriously influence the detection and direction of arrival (DOA) estimation performance of towed array. Null steering beamforming(NSBF), inverse beamforming(IBF) and non-data based beam-space transformation(NDBBT) are applied to suppress interference of tow ship in this paper. These methods are evaluated according to the improvement of signal interference noise ratio and DOA estimation precision. Simulation and experimental results show that all three methods can suppress interference of tow ship effectively so as to improve DOA estimation precision. In addition, IBF and NDBBT do better in suppressing interference than NSBF.

4:15

4pUW13. Generalized likelihood ratio test based on mode-space beamfoming with a horizontal line array. Yifeng Zhang (Underwater Acoust. College, Harbin Eng. Univ., Harbin NanGang District Nantong St. 145, Harbin, Hei Longjiang 150001, China, zhangyifeng2587@163.com), Nan Zou, Jinjin Wang, and Jia Feng (Underwater Acoust. College, Harbin Eng. Univ., Harbin, Heilongjiang, China)

The conventional detection method has difficulty in detecting the narrowband signal in the noisy shallow water waveguide, because detection under the plane-wave model mismatches the shallow water waveguide. In this paper, a generalized likelihood ratio test based on mode-space beamfoming is proposed, considering the acoustic field in a shallow water waveguide can be characterized by normal mode model. Due to the otherness of the mode wavenumbers, the generalized likelihood ratio test model is constructed by mode-space beamforming. As the mode-space beamfoming has an ill-posed problem, we ignore the small eigenvalues of the coefficient matrix, and then has a benefit on the performance of detector. Theoretical analysis and simulation result validate the effectiveness of the proposed algorithm.

THURSDAY EVENING, 10 MAY 2018 7:30 P.M. TO 9:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

Noise	Nicollet D3
Speech Communication	Nicollet D2
Underwater Acoustics	Greenway B