

Session 2aAAa

**Architectural Acoustics, Noise, Speech Communication, Psychological Physiological Acoustics, and Signal Processing in Acoustics: Sixty-Fifth Anniversary of Noise and Health:
Session in Honor of Karl Kryter I**

David M. Sykes, Cochair

Architectural Acoustics, Rensselaer Polytechnic Institute, 31 Baker Farm Rd., Lincoln, MA 01773

Michael J. Epstein, Cochair

Northeastern University, 360 Huntington Ave., 226FR, Boston, MA 02115

William J. Cavanaugh, Cochair

Cavanaugh Tocci Assoc. Inc., 3 Merifield Ln., Natick, MA 01760-5520

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. Auditory and non-auditory effects of noise on health: An ICBEN perspective. Mathias Basner (Psychiatry, Univ. of Pennsylvania Perelman School of Medicine, 1019 Blockley Hall, 423 Guardian Dr., Philadelphia, PA 19104-6021, basner@upenn.edu)

Noise is pervasive in everyday life and induces both auditory and non-auditory health effects. Noise-induced hearing loss remains the most common occupational disease in the United States, but is also increasingly caused by social noise exposure (e.g., through music players). Simultaneously, evidence on the public health impact of non-auditory effects of environmental noise exposure is growing. According to the World Health Organization (WHO), ca. 1.6 Million healthy life years (DALYs) are lost annually in the western member states of the European Union due to exposure to environmental noise. The majority (>90%) of these effects can be attributed to noise-induced sleep disturbance and community annoyance, but noise may also interfere with communication and lead to cognitive impairment of children. Epidemiological studies increasingly support an association of long-term noise exposure with the incidence of hypertension and cardiovascular disease. Up-to-date exposure-response relationships are needed for health impact assessments, to reassess the validity of current noise policy and to better mitigate the negative health consequences of noise. The International Commission on Biological Effects of Noise (ICBEN) is a non-profit organization constituted 1978 that promotes all aspects of noise effects research and its application through International Noise Teams and an International Congress every three years.

8:35

2aAAa2. Karl Kryter and psychoacoustics at Bolt Beranek and Newman. Leo Beranek (none, none, 10 Longwood Dr., Westwood, MA 02090, Beranekleo@ieee.org)

Karl Kryter joined Bolt Beranek and Newman from Harvard in 1957. He was named the head of the Department of Psychoacoustics where he worked with his close friend, J. C. R. Licklider. His first work was with speech intelligibility in noise. He was the psychoacoustician on the team of three with Laymon Miller and Leo Beranek in a multi-year project for the Port York Authority that led to the reduction of noise created by the first jet commercial aircraft, the Boeing 707, by 15 dB. He was responsible for developing a means for measuring the perception by people of the loudness of noise—particularly aircraft noise. Named the perceived noise decibel PNDB, it remains today the accepted method for measuring the effect of noise on people living around airports. After that he did sleep research until he departed in 1966 for the Stanford Research Institute.

8:55

2aAAa3. Effects of hospital noise on speech intelligibility. Frederick J. Gallun (VA Portland Health Care System, National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

The most straightforward way to consider the effects of noise on the ability to understand speech is in terms of overlap of energy at the level of the cochlea. This approach, sometimes called “energetic masking” has been well studied and is quantified in a number of international standards such as the Speech Intelligibility Index (ANSI, 2004). Recently, however, another aspect of noise has captured the interest of those studying speech intelligibility. Known as “informational masking,” this refers to the ability of noise to interfere with intelligibility even when energetic overlap is low. A study describing this phenomenon with respect to hospital noise will be described and it will be demonstrated that the impact of human voices on speech intelligibility provides a difficult problem for those trying to predict and quantify the effects of noise based only on the statistics of the acoustical signal.

9:15

2aAAa4. A tribute to Karl Kryter. Karl S. Pearsons (22689 Mulholland Dr., Woodland Hills, CA 91364-4941, karlkity@pacbell.net)

Karl Kryter came to Bolt Beranek and Newman in 1957 from Harvard to lead the Psycho-Acoustics Department. At that time, commercial aviation was transitioning from propeller driven aircraft to jet powered aircraft, like the Douglas DC8 and the Boeing 707. Public concern about the perceived increase in noise prompted the development of noise metrics. These new metrics provided better correlation with judged annoyance than overall sound levels; thus, perceived noise level (PNL) in PNdB units was introduced as a measure of airplane noise. He also contributed to establishing noise levels which may cause permanent hearing loss. Another interest was in creating a device that masked the noise of a dentist's drill. It was called the audio-analgesic. The last time I saw Karl was when we represented opposite sides in a court case. It was about the noise level of a medivac helicopter landing on a hospital roof. He was a wonderful boss and friend, a true role model. Fellow colleagues had only praise for him and the work environment he created. Born in 1914, he almost made it to 100 years, passing in 2013.

9:35

2aAAa5. Prescribing healthy hospital soundscapes. Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, Omaha, NE, eryherd@unl.edu) and Kerstin Persson Waye (Occupational & Environ. Medicine, The Sahlgrenska Acad., Gothenburg Univ., Gothenburg, Sweden)

In *The Effects of Noise on Man*, Karl Kryter emphasized that human reaction to sound is quantitatively related to the physical nature of sounds. He specified five unwanted characteristics of sound: "(a) the masking of unwanted sounds, particularly speech, (b) auditory fatigue and damage to hearing, (c) excessive loudness, (d) some general quality of bothersomeness or noisiness, and (e) startle." Hospital soundscapes have been shown to demonstrate all five characteristics in one fashion or another. Some unwanted effects of sound have also been shown, such as fragmented sleep and recuperation, cardiovascular response, pain and intensive care delirium—however, few studies have been able to causally link sounds to patient outcomes and few examine detailed characteristics other than loudness. This paper will summarize what we know about noise in hospitals and the health effects on occupants, including highlights from the Healthcare Acoustics Research Team (HART) body of research. Results will be used to identify important areas for future research.

9:55–10:05 Break

10:05

2aAAa6. Zeitgeist from Kryter's work at the PsychoAcoustics Laboratory at Harvard to the present. Mary Florentine (Dept. of Commun. Sci. and Disord. (226-FR), Dept. of Elec. and Comput. Eng., Northeastern Univ., Bldg. 226 FR, 360 Huntington Ave., Boston, MA 02115, m.florentine@neu.edu) and Michael Epstein (Auditory Modeling and Processing Lab., Dept. of Commun. Sci. and Disord., Dept. of Elec. and Comput. Eng., Dept. of BioEng., Northeastern Univ., Boston, MA)

Kryter's writings indicate that he was keenly aware that intense sounds damage auditory systems. He was also aware that there is no one-to-one correspondence between the physical world and our perception of it. In fact, much of Kryter's work sought to summarize that relationship in general terms as they applied to average observers. He consistently sought an inclusive theoretical framework. Kryter would have been fascinated by the recent discoveries regarding individual differences among listeners with different types of hearing losses. He would have delighted in the recent connections made between psychoacoustics and physiology, and the trend toward ecologically valid research. This presentation traces the *zeitgeist* from his work at the PsychoAcoustics Laboratory (PAL) at Harvard to the present, examining loudness across the entire dynamic range for listeners with normal hearing and different types of hearing losses. [This work was supported, in part, by NIH-NIDCD grants 1R03DC009071 and R01DC02241.]

10:25

2aAAa7. Neurophysiological effects of noise-induced hearing loss. Michael G. Heinz (Speech, Lang., and Hearing Sci. & Biomedical Eng., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, mheinz@purdue.edu)

Noise over-exposure is known to cause sensorineural hearing loss (SNHL) through damage to sensory hair cells and subsequent cochlear neuronal loss. These same histopathological changes can result from other etiological factors (e.g., aging, ototoxic drugs). Audiologically, each is currently classified as SNHL, despite differences in etiology. Functionally, noise-induced hearing loss can differ significantly from other SNHL etiologies; however, these differences remain hidden from current audiological diagnostic tests. For example, noise-induced mechanical damage to stereocilia on inner- and outer-hair cells results in neurophysiological responses that complicate the common view of loudness recruitment. Also, mechanical effects of noise overexposure appear to affect the tip-to-tail ratio of impaired auditory-nerve-fiber tuning curves in ways that create a much more significant degradation of the cochlear tonotopic representation than does a metabolic form of age-related hearing loss. Even temporary threshold shifts due to moderate noise exposure can cause permanent synaptic loss at inner-hair-cell/auditory-nerve-fiber synapses, which reduces neural-coding redundancy for complex sounds and likely degrades listening in background noise. These pathophysiological results suggest that novel diagnostic measures are now required to dissect the single audiological category of SNHL to allow for better individual fitting of hearing aids to restore speech intelligibility in real-world environments. [Work supported by NIH.]

10:45

2aAAa8. Noise-induced cochlear synaptopathy: Extending effects of noise on man? Sharon G. Kujawa (Massachusetts Eye and Ear Infirmary and Harvard Med. School, 243 Charles St., Boston, MA 02114, sharon_kujawa@meei.harvard.edu)

Kryter wrote that "...methods commonly used in medicine for the evaluation of impairment to hearing and the relation of this impairment to noise exposure may lead to significant underestimates of the severity of noise-induced hearing impairment and overestimations of tolerable limits for exposure to noise." (J. Acoust. Soc. Am. 53(5), 1211–1234). Although specifics differ, recent work in our laboratory provides strong evidence supporting this assertion. In a series of investigations conducted over the last 5 years, we have

shown that a primary consequence of noise exposure is acute loss of synapses between cochlear inner hair cells (IHC) and auditory nerve fibers followed by loss of the affected neurons themselves. Losses are robust, permanent, and progressive, even for temporary threshold shift-producing exposures, and aging of these ears is exaggerated, going forward. This noise-induced injury can hide behind normal thresholds, our traditional evidence of recovery; however, communication between IHCs and nerve fibers is nevertheless permanently interrupted. We hypothesize that these synaptopathic and neurodegenerative consequences of noise may contribute to speech discrimination difficulties in challenging listening environments. This talk will summarize what we know about structural and functional consequences and noise-risk implications of noise-induced cochlear synaptopathy. [Research supported by R01 DC 008577.]

11:05

2aAAa9. Age-related hearing loss, noise-induced hearing loss, and speech-understanding performance. Larry E. Humes (Dept. Speech & Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002, humes@indiana.edu)

Karl Kryter was keenly interested in the effects of noise on hearing, the interaction of age-related hearing loss (ARHL) with noise-induced hearing loss (NIHL), and, ultimately, the consequences of both on an individual's ability to understand speech in quiet and in noise. This review will touch on each of these topics and the interrelationships among them. The focus of this presentation will be on discussion of factors impacting the speech-understanding performance of older adults, with parallels drawn for NIHL where possible. Factors considered will include those represented by a variety of peripheral-auditory, central-auditory, and cognitive measures. Interactions among these factors will also be reviewed.

Contributed Paper

11:25

2aAAa10. The effects of noise on physician cognitive performance in a hospital emergency department. Peter Dodds, Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 161 Washington St. #4, Troy, NY 12180, phdodds@gmail.com), David Sykes (Acoust. Res. Council, Lincoln, MA), Wayne Triner, and Linda Sinclair (Albany Medical Ctr., Albany, NY)

The contemporary hospital emergency department presents a noisy and distracting sonic environment. While previous research has shown that the highly variable and overloud soundscape of hospitals leads to increased annoyance and physical stress for hospital employees, there is a need to objectively quantify the cognitive effects of noise on healthcare providers,

particularly physicians. An ongoing collaborative research effort by the authors seeks to better understand and measure the cognitive and psychoacoustic effects of noise in a busy urban emergency department. Using binaural room-acoustic measurements and psychoacoustic modeling, the soundscape of the emergency department at Albany Medical Center has been analyzed. Through the use of the n-Back cognitive test and calibrated binaural recordings and room-simulations, we are examining the effect of a variety of sonic phenomena on the working memory and cognitive control of emergency department physicians at the Albany Medical Center at various points throughout their shifts. This paper will discuss methods for in situ cognitive testing, preliminary results, and review possible interventions and their efficacy in reducing noise and distractions for physicians in the emergency department.

11:40–12:00

Panel Discussion and Open Microphone

TUESDAY MORNING, 19 MAY 2015

BALLROOM 2, 9:30 A.M. TO 12:30 P.M.

Session 2aAAb

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

David S. Woolworth, Cochair

Oxford Acoustics, 356 CR 102, Oxford, MS 38655

Andrew N. Miller, Cochair

Bai, LLC

Norman H. Philipp, Cochair

Geiler and Associates, LLC, 1840 E. 153rd. Cir., Olathe, KS 66062

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2015 Student Design Competition that will be professionally judged at this meeting. The competition involves the design of a performance venue in addition to a casino and hotel facility in downtown Pittsburgh, Pennsylvania. The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of US\$1250 will be made to the submitter(s) of the design judged "first honors." Four awards of US\$700 each will be made to the submitters of four entries judged "commendation."

Session 2aAB**Animal Bioacoustics and Psychological and Physiological Acoustics: Auditory Scene Analysis in Natural Environments**

Annemarie Surlykke, Cochair

Biology, University of Southern Denmark, Campusvej 55, Odense DK-5230, Denmark

Susan Denham, Cochair

*School of Psychology, University of Plymouth, Plymouth pl4 8aa, United Kingdom***Invited Papers****8:00**

2aAB1. Computational issues in natural auditory scene analysis. Michael S. Lewicki (Dept. of Elec. Eng. and Comput. Sci., Case Western Reserve Univ., CWRU/EECS Glennan Rm. 321, 10900 Euclid Ave., Cleveland, OH 44106, michael.lewicki@case.edu), Bruno A. Olshausen (Redwood Ctr. for Theor. Neurosci., UC Berkeley, Berkeley, CA), Annemarie Surlykke (Dept. of Biology, Univ. of Southern Denmark, Odense, Denmark), and Cynthia F. Moss (Dept. of Psychol. & Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Scene analysis is a complex process involving a hierarchy of computational problems ranging from sensory representation to feature extraction to active perception. It is studied in a wide range of fields using different approaches, but we still have only limited insight into the computations used by biological systems. Experimental approaches often implicitly assume a feature detection paradigm without addressing the true complexity of the problem. Computational approaches are now capable of solving complex scene analysis problems, but these often defined in a way that is of limited relevance to biology. The challenge is to develop approaches for deducing computational principles relevant to biological systems. I will present the view that scene analysis is a universal problem solved by all animals, and that we can gain new insights by studying the problems that animals face in complex natural environments. From this, I will present framework for studying scene analysis comprising four essential properties: (1) the ability to solve ill-posed problems, (2) the ability to integrate and store information across time and modality, (3) efficient recovery and representation of 3D scene structure, and (4) the use of optimal motor actions for acquiring information to progress toward behavioral goals.

8:20

2aAB2. The role of form in modeling auditory scene analysis. Susan Denham and Martin Coath (School of Psych., University of Plymouth, Plymouth pl4 8aa, United Kingdom, s.denham@plymouth.ac.uk)

Separating out and correctly grouping the sounds of a communicating animal from the natural acoustic environment poses significant challenges to models of auditory scene analysis, yet animals perform this task very effectively. To date, most models have focussed on simultaneous grouping cues and the segregation of discrete sound events, although some take the longer term context into account. Inspired by the important part that form plays in the segregation and recognition of visual objects, we consider the role of form in auditory scene analysis. By form in audition we mean the dynamic spectrotemporal patterns characterizing individual sound events as well as their timing with respect to each other. We present a model capable of segregating and recognizing natural communication calls within complex acoustic environments. Incoming sounds are processed using a model of the auditory periphery and fed into a recurrent neural network that rapidly tunes itself to respond preferentially to specific events. Representations of predictable patterns of events in the sequence are created on the fly and maintained on the basis of their predictive success and conflict with other representations. Activation levels of these representations are interpreted in terms of object recognition.

8:40

2aAB3. Audio-vocal feedback in bats and new roles for echolocation calls in social communication. Kirsten M. Bohn (Florida Int. Univ., 518 Giralda Ave., Coral Gables, FL 33134, bohnkirsten@gmail.com) and Michael Smotherman (Texas A&M Univ., College Station, TX)

An important aspect of auditory scene analysis is the specialized neurocircuitry required for vocal production in a dynamic acoustic environment. Although often taken for granted, enhanced audio-vocal feedback is relatively uncommon in animals and yet an important precursor to vocal learning. We argue that vocal complexity in bats is an exaptation of the highly specialized audio-vocal feedback system that has evolved for echolocation. First, we explore how audio-vocal feedback enhances echolocation. Second, we review how echolocation pulses serve social functions by providing information to receivers (like gender, identity, or food availability). Third, using our research on molossid bats (family Molossidae), we explore whether vocal plasticity in sonar has contributed to an expanded role for sonar pulses in social communication. In at least three molossids, roosting bats rapidly sing in response to echolocation pulses of flying conspecifics. However, more importantly, we show that in multiple species, echolocation is produced in purely social contexts. Roosting bats embed pulses and feeding buzzes into their courtship songs that are not acoustically distinct than when foraging. Finally, some

molossidids not only sing in roosts, but also in flight—that is, they echolocate and sing simultaneously. These findings indicate that echolocation plays even more of a role in social communication than commonly believed and that the production of echolocation and social communication is tightly coupled and coordinated at a high level.

9:00

2aAB4. Are the mechanisms for stream segregation shared among anurans and other tetrapods? Jakob Christensen-Dalsgaard (Biology, Univ. of Southern Denmark, Campusvej 55, Odense M DK-5230, Denmark, jcd@biology.sdu.dk)

Many male anurans (frogs and toads) call in large aggregations. Since the fundamental task of anuran auditory communication probably is to attract and localize potential mates, segregation of the callers is likely an important task for the auditory system. Behavioral experiments have shown that elements of stream segregation, based on frequency separation and spatial separation, can be demonstrated in anurans. The neural processing of these cues is interesting, because most auditory processing probably takes place in the midbrain torus semicircularis (TS, homolog to the inferior colliculus). It has been shown that spatial release from masking is sharpened by 6 dB in the TS, and that neurons in the TS are selective for call rates and number of call pulses. However, recently electrical stimulation of thalamic structures have demonstrated possible attentional modulation of TS responses that could also function in stream segregation. The modulation was call-specific and could also enhance binaural cues. In conclusion, many of the elements involved in auditory segregation in other animals are also found in anurans. However, it is uncertain whether the underlying mechanisms are similar. One crucial element in primate sound segregation—the representation of the competing streams in the neural responses—has so far not been demonstrated in anurans, and the most robust elements of stream segregation is really based on frequency processing, a relatively simple and ubiquitous property of most auditory systems.

9:20–9:35 Break

9:35

2aAB5. Measurement and control of vocal interactions in songbirds. Richard Hahnloser (Inst. of Neuroinformatics, ETH Zurich and Univ. of Zurich, Winterthurerstrasse 190, Zurich, Schweiz 8057, Switzerland, rich@ini.phys.ethz.ch)

One obstacle for investigating vocal interactions in vertebrate animals is difficulty of discriminating individual vocalizations of rapidly moving, sometimes simultaneously vocalizing individuals. To overcome this obstacle, we have developed an ultra-miniature back-attached sound/acceleration recording system that allows perfect separation of birdsong vocalizations irrespective of background noise and the number of vocalizing animals nearby. With this system, we have been able to identify hierarchies in the vocal interactions among adult male and female zebra finches. Furthermore, to causally interfere with vocal interactions in groups of up to four birds, we have developed an echo cancellation system that allows us to have full digital control of vocal interactions between any bird pair. With these tools, we hope to be able to further advance the dissection of vocal communication in songbirds.

9:55

2aAB6. Listening through the ears of echolocating *Myotis daubentonii* bats hunting in groups. Annemarie Surlykke, Mads N. Olsen (Biology, Univ. of Southern Denmark, Campusvej 55, Odense DK-5230, Denmark, ams@biology.sdu.dk), and Carien Mol (Biology, Univ. of Southern Denmark, Utrecht, Netherlands)

Echolocation allows bats to orient and catch insects in the dark. One intriguing question is how bats are able to recognize their own echoes when hunting in groups. Bats can adjust their call frequency, but this strategy may not be efficient for species like *Myotis daubentonii* emitting broadband frequency modulated signals. However, the actual masking may be reduced for bats like *M. daubentonii* emitting short directional signals with low duty cycle. We used a 12-microphone array and infrared camera to record flight and vocal behavior of groups of Daubenton's bat hunting over a river in the field. We used flight path reconstructions to analyze the acoustic world from the bat's perspective. For the focal bat, we reconstructed (1) its own emissions, (2) the pulses from conspecifics nearby, (3) its own insect echoes, and (4) insect echoes from pulses of the other bat. The data showed that when two bats fly together echoes were only rarely overlapped by the other sounds. We here provide a framework for obtaining detailed information of flight and echolocation behavior in complex natural surroundings, emphasizing the importance of adapting a "bat's viewpoint" when studying natural echolocation.

Contributed Papers

10:15

2aAB7. Nocturnal peace at a Conservation Center for Species Survival? Susan M. Wiseman (None, Waco, TX) and Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Austin, TX 78712, pswilson@mail.utexas.edu)

E.O. Wilson suggested that the natural world is the most information-rich acoustic environment, especially for animals and indigenous peoples, and that is directly and indirectly essential to their survival (Wilson, 1984, *Biophilia*, Harvard University Press). Agriculturalists still value and constantly monitor

their soundscape since it provides invaluable cues that may lead to success or failure. Krause reports that healthy natural soundscapes comprise a myriad of biophony, and indeed the ecological health of a region can be measured by its diverse voices (Krause, 1987, "Bio-acoustics: Habitat ambience & ecological balance," *Whole Earth Rev.*). But do such soundscapes fall silent for much of the night? And are there extensive periods of silence at a highly successful Conservation Center for Species Survival? This study analyzes the soundscape continuously recorded at Fossil Rim Wildlife Center in Texas for a week during Fall 2013, to determine the prevalence of quiet periods and the acoustic environment in which such periods appeared to occur.

10:30

2aAB8. Active enhancement found in responses of the cochlear nerve to detect weak echoes created by the auditory periphery in Japanese house bat, *Pipistrellus abramus*. Hiroshi Riquimaroux, Hiroshi Onodera (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp), and Ikuo Matsuo (Tohoku-gakuin Univ., Sendai, Japan)

Characteristics of frequency tunings related to echolocation were investigated in an FM bat, Japanese house bat (*Pipistrellus abramus*). A particular attention was paid on their terminal frequency of FM sweep. When search for preys in the field, they stretch duration of echolocation pulses and reduce FM sweeping range, resulting in pseudo constant frequency pulses. Distribution of best frequency in the neurons in the inferior colliculus indicated that about 50% of neurons appeared to be tuned to the pseudo constant frequency, around 40 kHz. The compound action potentials were recorded from medulla oblongata through a metal telectrode. We hypothesized that the bat has a peripheral system to enhance frequency component of 40 kHz to be detected easier than other frequency components. Active enhancement of 40 kHz frequency component by the outer hair cell and overwhelming number of cochlear nerves in charge of 40 kHz would create low threshold and high resolution sensitivity for frequency range around 40 kHz. Results of the present study show that compound action potential of the cochlear nerve (N1) generates nonlinearly enhanced amplification in frequency range of 40 kHz but linear amplification in lower frequency ranges.

10:45

2aAB9. Evaluation of a simple model for the acoustics of bat swarms. Mingyi Liu and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, lmingyi@vt.edu)

Bats using their biosonar while flying in dense swarms may face significant bioacoustic challenges, in particular mutual sonar jamming. While possible solutions to the jamming problem have been investigated several times, the severity of this problem has received far less attention. To characterize the acoustics of bat swarms, a simple model of the acoustically relevant properties of a bat swarm has been evaluated. The model contains only four parameters: bat density, biosonar beamwidth, duty cycle, and a scalar measure for the smoothness in the flight trajectories. In addition, a threshold to define substantial jamming was set relative to the emission level. The simulation results show that all four model parameters can have a major impact on jamming probability. Depending on the combination of parameter values, situations with or without substantial jamming probabilities could be produced within reasonable ranges of all model parameters. Hence, the model suggests that not every bat swarm does necessarily impose grave jamming problems. Since the model parameters should be comparatively easy to estimate for actual bat swarms, the simulation results could give researchers a way to assess the acoustic environment of actual bat swarms and determine cases where a study of biosonar jamming could be worthwhile.

11:00

2aAB10. Noise induced threshold shifts after noise exposure: Are bats special? Andrea Simmons (Brown Univ., Box 1821, Providence, RI 02912, Andrea_Simmons@brown.edu), Michaela Warnecke (Johns Hopkins Univ., Baltimore, MD), Kelsey Hom, and James Simmons (Brown Univ., Providence, RI)

We conducted psychophysical and neurophysiological experiments to test the hypothesis that big brown bats suffer less severe temporary threshold shifts after noise exposure than other small mammals that also hear high frequency sounds. Five big brown bats were trained in psychophysical detection experiments to obtain thresholds to FM sweeps spanning the frequency range of their echolocation calls. Bats were then exposed to 115 dB of broadband noise for one hour, and thresholds re-measured 20 min and 24 hour after exposure. For all bats, threshold elevations at 20 min post-exposure were 3 dB or less, while at 24 hour post-exposure, thresholds were similar to those obtained pre-exposure. Local field potentials were recorded in the cochlear nucleus of anesthetized big brown bats before and after noise exposure. Neural thresholds to FM sweeps and to single tone frequencies were unaffected by noise exposure. These data suggest that big brown bats are an excellent model system to study naturally occurring immunity to noise damage. [Work supported by ONR and the Capita Foundation.]

11:15

2aAB11. Underwater auditory scenes near coral reefs. Erica Staaterman and Claire Paris (Appl. Marine Phys. & Marine Biology and Fisheries, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149, e.staaterman@gmail.com)

Recent studies suggest that reef soundscapes may serve as long-distance navigational cues for settlement-stage larval fishes. To understand the role of acoustic signals during larval settlement, we investigated temporal and spatial patterns in coral reef soundscapes in the Florida Keys, USA. We used 14-month simultaneous acoustic recordings from two nearby reefs, coupled with environmental data, to describe temporal variability in the soundscape on scales of hours to months, and to understand abiotic and biological components. We also recorded acoustic pressure and particle acceleration (which fish larvae likely perceive) with increasing distance from the reef. High acoustic frequencies typically varied on daily cycles, while low frequencies were primarily driven by lunar cycles. Some of these patterns were explained by environmental conditions, while others were attributed to biological sounds. At both reefs, the highest sound levels (~130 dB re:1 μ Pa) occurred during new moons of the wet season, when many larval organisms settle on the reefs. Acoustic particle acceleration did not propagate as far as predicted from the plane-wave equation. The patterns uncovered here provide valuable insights into underwater acoustic scenes, and this study represents an example of novel acoustic instruments and analysis techniques that can be applied to any ecosystem.

2a TUE. AM

Session 2aBA

Biomedical Acoustics: Ultrasound Contrast Agents: Nonlinear Bubble Dynamics

Tyrone M. Porter, Cochair

Boston University, 110 Cummings Mall, Boston, MA 02215

Jonathan A. Kopechek, Cochair

University of Pittsburgh, 200 Lothrop Street, Pittsburgh, PA 15213

Invited Papers

8:00

2aBA1. Role of lipid coating properties on nonlinear microbubble dynamics. Klazina Kooiman (Dept. of Biomedical Eng., Thorax Ctr., Erasmus MC, P.O. Box 2040, Rm. Ee2302, Rotterdam 3000 CA, Netherlands, k.kooiman@erasmusmc.nl)

Although nonlinear microbubble dynamics are exploited in many contrast-enhanced ultrasound imaging modalities, the role of lipid coating properties on nonlinear dynamics is not fully understood. Our studies show that microbubble vibrations differ between bubbles with homogeneous lipid distribution (main component DPPC, C16) and bubbles with heterogeneous lipid distribution (main component DSPC, C18). Relative to DPPC microbubbles, DSPC microbubbles were shown to elicit more second harmonic emissions, but exhibit a lower propensity to initiate subharmonic emissions. This suggests that homogeneous microstructures favor subharmonic emissions, whereas heterogeneous microstructures produce higher second harmonic emissions. Additionally, the nonlinear "compression-only" oscillation behavior is hypothesized and theoretically shown to relate to buckling of the lipid coating. High-speed fluorescence recordings revealed for the first time formation of hot spots (i.e., high local concentrations of lipids) in the coating of oscillating microbubbles during the compression phase, suggesting evidence of buckling/folding of the coating. However, hot spots were not correlated to compression-only behavior, but were related to the level of compression of the microbubbles during insonification, with a relative compression threshold of 15–20%. This suggests that compression-only behavior is not directly related to buckling of the lipid coating. This work sheds insight into the role of lipid coating properties on nonlinear microbubble behavior.

8:30

2aBA2. Ultrafast frame rate microscopy of microbubble oscillations: Current studies employing the UPMC-Cam. Brandon Helfield, Xucai Chen, Bin Qin, and Flordeliza Villanueva (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Heart and Vascular Inst., Univ. of Pittsburgh Medical Ctr., 966 Scaife Hall Fl. 3550 Terrace St., Pittsburgh, PA 15213, helfieldb@upmc.edu)

Ultrasound-stimulated microbubbles are clinically employed for diagnostic imaging and have been shown to be a feasible therapeutic strategy for localized drug and gene delivery applications. In order to investigate microbubble oscillation dynamics, the University of Pittsburgh Medical Center has developed the UPMC-Cam—an ultrafast imaging system capable of recording 128 frames at up to 25 million frames per second, one of only two in the world currently in use. Current studies include elucidating the effect of fluid viscosity on microbubble behavior. At lower pressures (0.1–0.25 MPa), an increase in fluid viscosity from 1 to 4 cP alters the fundamental and second-harmonic oscillation amplitudes in a bubble size-dependent manner, related to resonance. At higher pressures (0.5–1.5 MPa), a significant decrease in the propensity for microbubble fragmentation was observed in the more viscous fluid environment. Additionally, studies aimed at gaining physical insights into sonoporation have been conducted in which individual bubbles lay adjacent to a cell monolayer. These studies have revealed a maximum microbubble expansion threshold above which sonoporation occurs, likely related to the associated shear stresses exerted by a microbubble on the adjacent cell membrane. The UPMC-Cam has shown to be an invaluable tool for investigating biophysical-related phenomena over a wide range of ultrasound-microbubble applications.

Contributed Papers

9:00

2aBA3. Radial excursions of bound and non-bound targeted lipid-coated single microbubbles. Tom van Rooij, Antonius F. van der Steen, Nico de Jong, and Klazina Kooiman (Dept. of Biomedical Eng., Thorax Ctr., Erasmus MC, Postbus 2040, Rotterdam 3000 CA, Netherlands, t.van-rooij@erasmusmc.nl)

One of the main challenges for ultrasound molecular imaging is acoustically distinguishing non-bound microbubbles from those that have bound to their molecular target. We previously showed that biotinylated DPPC-based

microbubbles (16 C-atoms) had a larger binding area and a more domed shape when bound to a streptavidin-coated surface than DSPC-based microbubbles (18 C-atoms) [1]. In the present *in vitro* study, we used the Brandaris 128 ultrahigh-speed camera (~15 Mfps) to compare the acoustical responses of biotinylated DPPC and DSPC-based microbubbles in a non-bound configuration and bound to a streptavidin-coated membrane, aiming to acoustically discriminate them from each other. The microbubbles were driven at a pressure of 50 kPa and at frequencies between 1 and 4 MHz. The main difference between bound and non-bound microbubbles was the lower radial excursion at the fundamental frequency for bound microbubbles.

Resonance frequencies and subharmonic responses were the same for bound and non-bound microbubbles. Finally, at the second harmonic frequency, we found higher relative radial excursions for bound DSPC-based microbubbles than for non-bound DSPC microbubbles, whilst there was no difference for DPPC-based microbubbles. This might provide opportunities to acoustically discriminate bound from non-bound DSPC microbubbles. [1] Kooiman *et al.*, *Eur. J. Lipid Sci. Technol.* (2014).

9:15

2aBA4. Heat and mass transfer effects on forced radial oscillations in soft tissue. Carlos Barajas and Eric Johnsen (Mech. Eng., Univ. of Michigan, 500 South State St., Ann Arbor, MI 48109, carlobar@umich.edu)

Cavitation has a vast array of biomedical purposes such as histotripsy, used to mechanically fractionate soft tissue. Alternatively, cavitation can cause unwanted biological damage in diagnostic ultrasound, e.g., when using ultrasound contrast agents. These benefits and harmful effects make it desirable to develop models to better understand cavitation in viscoelastic media. We numerically model cavitation in a viscoelastic (Kelvin-Voigt) medium using the Keller-Miksis equation to describe the radial bubble motion. We specifically focus on the effects of heat and mass transfer, through numerical solution of the full energy and mass equations, reduced models, and theoretical analysis. We will discuss how thresholds on the driving pressure and frequency where thermal and mass transfer effects become significant can be determined, as well as ranges of amplitudes for which reduced models can be used accurately. We will also present theoretical investigations on how oscillation properties depend on these effects. Our findings will provide guidelines to determine regimes when heat and mass transfer are dominant. [This work was supported in part by NSF grant number CBET 1253157 and NIH grant number 1R01HL110990-01A1.]

9:30

2aBA5. An *in vitro* study of subharmonic emissions from monodisperse lipid-coated microbubbles. Qian Li (Biomedical Eng., Boston Univ., 44 Cummington Mall, Rm. B01, Boston, MA 02215, qianli@bu.edu), Mark Burgess, and Tyrone Porter (Mech. Eng., Boston Univ., Boston, MA)

Subharmonic imaging is an ultrasound imaging method which utilizes the subharmonic response of a contrast agent to ultrasound excitation. It is possible to achieve high agent-to-tissue contrast because tissues do not emit subharmonic signals. In this project, we investigated the relationship between subharmonic emissions from monodisperse lipid-coated microbubbles and acoustic pressure. First, the resonance frequency for monodisperse microbubble suspension was determined from attenuation spectra measured using a through-transmission technique. Next, the microbubbles were excited at the resonance and at twice the resonance frequency. A transducer positioned orthogonal to the excitation transducer was used to detect acoustic emissions from the microbubbles at half the excitation frequency. It was found that the pressure required for subharmonic emissions was lower when monodisperse microbubbles were excited at twice the resonance frequency. For example, 6.5 μm diameter microbubbles with a resonance frequency of 1.4 MHz had a pressure threshold for subharmonic emissions of approximately 30 kPa at 2.8 MHz excitation while those excited at 1.4 MHz could not be forced into subharmonic oscillations for the pressure range we used in this study (i.e., below 150 kPa). Implications of these results on the use of monodisperse lipid-coated microbubbles for subharmonic imaging will be discussed.

9:45

2aBA6. Exploitation of nonlinear acoustical effects of air bubbles in water for a bubble/target discriminating sonar. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu) and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Target detection by traditional sonar can be undermined by air bubbles in various environments, including the surf zone, within gas-bearing sediments, or within seagrass beds. Bubbles introduce a variety of strong acoustical effects, including the onset of many nonlinear effects at sound pressure levels far below that required in bubble-free water. This paper describes laboratory tank experiments demonstrating a bubble/target discriminating sonar

that exploits some of these nonlinear effects. Bubble plumes were generated in the tank by releasing compressed air through a porous ceramic plate. Rigid targets were also positioned at various locations in the tank. A high amplitude driving pulse excited translation of bubbles in the water column yielding Doppler signatures from conventional imaging sonar, while Doppler signatures were absent from the heavy rigid targets ensounded by the same driving signal. For sufficient bubble density, subharmonic generation was also observed in regions of bubbly water, but not from the rigid targets. The goal of this work is to use one or both of these nonlinear effects to identify and reduce clutter caused by bubbles and improve target detection for sonar operation in areas where bubbles are present. [Work supported by ARL:UT and ONR.]

10:00–10:15 Break

10:15

2aBA7. Singular value decomposition: A better way to analyze cavitation-noise data. Parag V. Chitnis (Dept. of BioEng., George Mason Univ., 4400 University Dr., 1G5, Fairfax, VA 22032, pchitnis@gmu.edu), Caleb Farny (Dept. of Mech. Eng., Boston Univ., Boston, MA), and Ronald A. Roy (Dept. of Mech. Eng., Univ. of Oxford, Oxford, United Kingdom)

Detection of inertial and stable cavitation is important for guiding high-intensity focused ultrasound (HIFU). Acoustic transducers can passively detect broadband noise from inertial cavitation and the scattering of HIFU harmonics from stable cavitation bubbles. Conventional approaches to separating these signals typically involve a custom comb-filter applied in the frequency domain followed by an inverse-Fourier transform, which cannot be implemented in real-time. We present an alternative technique based on singular value decomposition (SVD) that efficiently separates the broadband emissions and HIFU harmonics in a single step. Spatio-temporally resolved cavitation detection was achieved using a 128-element, 5-MHz linear-array system operating at 15 frames/s. A 1.1-MHz transducer delivered HIFU to tissue-mimicking phantoms for a duration of 5 s. Beamformed radiofrequency signal corresponding to each scan line and frame were assembled into a matrix and SVD was performed. Eigen vectors that corresponded to HIFU harmonics were identified as ones whose spectrum contained a peak centered at one of the harmonic frequencies. The projection of data onto this eigen-base produced the stable-cavitation signal. The remaining eigenvectors were used to obtain the inertial-cavitation signal. The SVD-based method faithfully reproduced the structural details in the spatio-temporal cavitation maps produced using the comb-filter albeit with marginally lower SNR.

10:30

2aBA8. Microbubble behavior during long tone-burst ultrasound excitation. Xucai Chen, Jianjun Wang, John Pacella, and Flordeliza S. Villanueva (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Univ. of Pittsburgh Medical Ctr., 3550 Terrace St., Scaife Hall, Rm. 969, Pittsburgh, PA 15261, chenxx2@upmc.edu)

Ultrasound-microbubble mediated therapies have been investigated to restore perfusion and enhance drug/gene delivery, using both short ($\sim\mu\text{s}$) and long ($\sim\text{ms}$) ultrasound tone-bursts. Microbubble oscillation behavior can be observed with ultra-high speed microscopy; however, due to the high frame rate required and limited number of frames available, microbubble dynamic behavior can only be observed for a few acoustic cycles ($\sim\mu\text{s}$). In this report, the fate of microbubbles throughout long tone-burst ultrasound exposures (5 ms) was explored via direct microscopic optical observation in conjunction with passive cavitation detection, over a range of acoustic pressures (0.25, 0.5, 1.0, and 1.5 MPa) and microbubble concentrations ($2e6$, $2e7$, and $2e8$ MB/ml). The ultrasound system was triggered from UPMC Cam, the ultra-high speed camera system at the University of Pittsburgh Medical Center, such that microbubble activity at various time points (0–5 ms) within the long tone-burst was captured. Microbubbles first underwent stable or inertial cavitation depending on the acoustic pressure used, and then formed gas-filled clusters that continued to oscillate, break up, and form new clusters. Cavitation detection confirmed continued, albeit diminishing acoustic activity throughout the 5 ms ultrasound excitation. This discovery suggests that persisting cavitation activity during long tone-bursts may confer additional therapeutic effect for ultrasound-microbubble mediated therapies.

10:45

2aBA9. Radiation damping of an arbitrarily shaped bubble. Kyle S. Spratt, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 4307 Caswell Ave. APT E, Austin, TX 78751, sprattkyle@gmail.com), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Mark S. Wochner (AdBm Technologies, Austin, TX), and Mark F. Hamilton (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, California)

Large encapsulated bubbles have recently been used for abating low-frequency anthropogenic underwater noise [J. Acoust. Soc. Am. **135**, 1700–1708 (2014)]. The use of encapsulation allows for the possibility of bubbles that are significantly nonspherical in their equilibrium state. Strasberg [J. Acoust. Soc. Am. **25**, 536–537 (1953)] investigated the resonance frequency of an ideal bubble with arbitrary shape and found that the dependence of resonance frequency on the shape of the bubble reduced to a well-known problem in electrostatics. The present work extends that analysis to include the effects of radiation damping on the oscillation of a bubble, and does so by including a loss term due to Ilinskii and Zabolotskaya [J. Acoust. Soc. Am. **92**, 2837–2841 (1992)] in the volume-frame dynamical equation for the bubble. An expression is given for the amplitude of the acoustic field scattered from the bubble, and it is shown that radiation damping scales as resonance frequency cubed for arbitrarily shaped bubbles having the same volume. Comparisons are made with previous work on scattering from spherical and prolate spheroidal bubbles, and various new bubble shapes are considered. [Work supported by AdBm Technologies, the ARL:UT McKinney Fellowship in Acoustics and ONR.]

11:00

2aBA10. Effect of blood viscosity and thrombus composition on sonoreperfusion efficacy. John Black, Frederick Schnatz, Francois T. Yu, Xucai Chen, Judith Brands, Flordeliza Villanueva, and John Pacella (Heart and Vascular Inst., Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall S960, Pittsburgh, PA 15261, yuft@upmc.edu)

Embolization during stenting for myocardial infarction (MI) causes microvascular obstruction (MVO). We demonstrated that ultrasound (US) and microbubble (MB) therapy [sonoreperfusion (SRP)] resolves MVO from venous microthrombi at plasma viscosity. However, blood is more viscous than plasma, and arterial microthrombi are mechanically distinct from venous clot. We tested whether blood viscosity and arterial microthrombi decrease SRP efficacy in our *in vitro* model of MVO. Lipid MB in plasma and blood viscosity PBS were passed through a 40 μm pore mesh. Arterial microthrombi were formed in a Chandler loop. Venous microthrombi were made statically. MVO was created by occluding the mesh with microthrombi until upstream pressure reached 40 mmHg. US (1 MHz; 5000 cycles; 0.33 Hz PRF; 0.6, 1.0, and 1.5 MPa peak negative pressure) was delivered with a single focused element transducer. MVO caused by arterial thrombi at increased viscosity resulted in less upstream pressure drop during therapy. PCD showed a decrease in inertial cavitation (IC) when viscosity

was increased. SRP efficacy is less with arterial thrombi compared to venous thrombi, and higher viscosity further reduces SRP efficacy by decreasing IC. These findings could help guide the selection of US parameters for *in vivo* optimization of SRP.

11:15

2aBA11. Subharmonic threshold quantification of ultrasound contrast agents. Rintaro Hayashi, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), Parag Chitnis (BioEng., George Mason Univ., Fairfax, VA), and Jeffrey A. Ketterling (Riverside Res., New York, NY)

The measurement and prediction of the subharmonic threshold of ultrasound contrast agents is of significant interest, particularly for high frequency applications. Theoretical analytical predictions typically follow from a weakly nonlinear analysis assuming continuous, single frequency forcing. Furthermore, numerical simulation and experimental definitions are often based on the quantification of spectral response of the nonlinear harmonics. Limitations of these approaches are investigated with respect to pulsed forcing and associated non-stationary (chirp) excitations. Novel quantification and definitions are proposed with respect to instantaneous frequency and relative energy content of an empirical mode decomposition of the scattered pressure. Teager-Kaiser energy operator allows for additional quantification. The methodology is examined with respect to experimental polymer contrast agent data.

11:30

2aBA12. Stress and strain fields produced by violent bubble collapse. Lauren Mancia (Mech. Eng., Univ. of Michigan, 2016 Walter E. Lay Automotive Lab, Ann Arbor, MI 48109-2133, lamancha@umich.edu), Eli Vlasisavljevich (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Matthew Warnez (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Cavitation finds a key application in therapeutic ultrasound. For example, histotripsy relies on the rapid expansion of cavitation bubbles to fractionate soft tissue. To fully understand the mechanisms responsible for tissue fractionation, we numerically model cavitation in a tissue-like medium, focusing on the effect of its viscoelastic properties (viscosity, elasticity, and relaxation). It is hypothesized that ablation is caused by high strain rates and stresses exerted on the surrounding tissue as bubbles rapidly expand from nanometer to micron scales. The present study uses robust numerical techniques to compute the stress fields in the surrounding medium produced by single-bubble expansion. Bubble expansion is driven by a waveform that approximates a histotripsy pulse with relevant parameters, and soft tissue surrounding the bubble is modeled as a viscoelastic medium with Neo-Hookean elasticity. We will examine the stress, strain, and temperature fields produced during this process to explain potential damage mechanisms.

Session 2aNSa**Noise, Psychological and Physiological Acoustics and ASA Committee on Standards: Noise, Soundscape, and Health**

Brigitte Schulte-Fortkamp, Cochair

Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Klaus Genuit, Cochair

*HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany***Chair's Introduction—8:30*****Invited Papers*****8:35****2aNSa1. Soundscape as a resource to balance the quality of an acoustic environment.** Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Impacts of noise on people's quality of life, on their health, and on the quality of the living environment, including economic costs will be addressed in this paper. It follows the understanding of potential health effects related to noise by the European Environment Agency. "Adverse effects of noise occur when intended activities of the individual are disturbed. The sound level of the acoustic stimulus, its psychoacoustical sound characteristics, the time of its occurrence, its time course, its frequency spectrum, and its informational content will modify the reaction." EEA 2010. To account for the diversity of living situations related research will focus on the cooperation of human/social sciences like psychology, sociology, architecture, anthropology, and medicine. Soundscape will provide the needed paradigm shift in research and evaluation. The International Standard ISO/DIS 12913-1 has as its purpose the enabling of a broad international consensus on the definition of "soundscape" and its respective evaluation. It is more than urgent to understand that there is the need to provide a solid foundation for communication across disciplines and professions with an interest in achievements on better solutions for the people concerned.

8:55**2aNSa2. The character of noise and its relation to noise effects.** André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

The existence of noise effects leading to adverse health effects is beyond controversy. However, the main factors responsible for specific noise effects are still not clear. The human hearing system does not only register the sound pressure level of noise. It shows a high performance in amplitude, frequency, and time resolution and recognizes patterns in noise. Interestingly, several physiological reactions are not only influenced by the sound pressure level of noise. Due to certain psychoacoustic properties of noise or the level of noise annoyance stronger physiological reactions are stimulated. In order to predict responses to noise more reliably, psychoacoustic parameters and further hearing related-parameters have to be applied. Moreover, it is decisive to distinguish between auditory sensation and sound perception. Sound perception refers to complex processes, where additional information is processed and "mixed" with basic auditory sensations. Sound perception implies the cognitive processing, interpretation, and assessment of sound. Some studies suggest that the sound perception dimension is highly relevant to evoked physiological reactions and noise effects. The paper illustrates the difference between auditory sensation and sound perception and highlights the need to analyze sound in terms of perception to understand noise effects more in detail.

9:15**2aNSa3. The operating theater—A paying challenge for soundscape design.** Holger Sauer (Inst. for Medical PsychoPhys., Klinikum Westfalen, Klinik am Park, Brechtener Straße 59, Lünen D-44536, Germany, holger.sauer@klinikum-westfalen.de)

The operating theater (OT) is virtually a parade example of a place with a high stress potential—not only for the working personnel but also for the patients. The reasons for this phenomenon are complex, and, partially inescapable. But since evidence reveals more and more the devastating consequences of this kind of stress for all persons concerned, there are good reasons to eliminate or at least reduce as many stress factors as possible. The soundscape is one of the major items which can work both stressing and de-stressing, however, quite diverging and dependant on the need and preferences of the several players. It is not sufficient simply to reduce the acoustic level since sound has different implications for the respective persons. Instead, the OT of the future requires a differentiated soundscape—implying hardware, software, and strategical patterns—which can be used in a more individualized way and, in addition, is adaptable to the respective circumstances. In order to achieve this, major efforts are required, among others due to complex preconditions; but it can be expected that these efforts will yield worthwhile results, which even will be transferable to other stress-sensitive domains.

2aNSa4. Soundscapes and human restoration in green urban areas. Irene van Kamp, Elise van Kempen, Wim Swart, and Hanneke Kruize (Sustainability, Environment and Health, National Inst. Public Health and the Environment, PO Box 1 Postbus 10, Bilthoven, Utrecht 3720 BA, Netherlands, irene.van.kamp@rivm.nl)

There is increasing interest in the association between landscapes, green and blue space, open countryside and human well-being, quality of life, and health. Most studies in this field do not account for the positive or negative moderating effects of the acoustic environment. This is partly due to the lack of relevant data, although basic models do refer to the role of traffic related noise and air-pollution. This paper reports on the results of a European study (Phenotype) into the health effect of access to and use of green area in four European cities. At the four study centers, people were selected from neighborhoods with varying levels of socioeconomic status and green space. By structured interview, information was gathered about availability, use, and importance of green space in the immediate environment, as well as the sound quality of favorite green areas used for physical activity, social encounters, and relaxation. Data are also available about perceived mental and physical health and medication use. This allows for analyzing the association between indicators of green and health, while accounting for perceived soundscapes. Audit data about the sound quality are also available at neighborhood level as a point of reference.

2aNSa5. Acoustic preservation: Preventing unseen loss in our historic sites. Pamela Jordan (Dept. of Eng. Acoust., Humboldt Fel-low, Tech. Univ. of Berlin, Krennener Strasse 15a, Berlin 10435, Germany, pam.f.jordan@gmail.com)

Soundscape research examines the vital role that sound plays within the integrated complexity of any specific environment, be it a rural landscape, cityscape, or indoor scenario. Historic preservation, on the other hand, generally seeks to safeguard shared built histories and frames the physical conservation of a site by prioritizing certain elements or activities for contemporary audiences. These designed interventions function similarly whether they are in “nature” or in the attic of a building. Yet the acoustic dimension of a site is seldom considered a vital part of preservation work. Even more rarely is the aural environment considered a contributing historic resource. Nevertheless, the soundscape of a place is an important cultural resource to learn from and protect—one that can potentially unlock a new dimension of understanding in a site’s past. This submission presents current work being carried out in Berlin that bridges the gap between preservation practice and soundscape research. Acoustic concerns specific to historic sites will be discussed through case study examples. A discussion of developing methodologies for mapping soundscapes of historic sites will follow.

10:15–10:30 Break

Contributed Papers

10:30

2aNSa6. Soundlapse: Audio strategies for documenting soundscapes with timelapse photography. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

Timelapse photography—the procedure of taking photographs at regular intervals and then sequencing them together at a rate typical of film/video (e.g., 24 frames per second) to produce a radically accelerated and compressed portrayal of time passing—is widely popular across such diverse areas as scientific research, weather tracking, filmmaking, and even reality TV shows, which commonly use timelapse footage to segue between scenes and reflect the passing of time. The compelling visual effect and broad legibility of timelapse photography makes it a powerful tool for revealing flows, shifts, and patterns in the environment, which would otherwise be too subtle to perceive in real time. One of the most important challenges in documenting the salient features of a soundscape is revealing similar shifts and patterns across time in the acoustic environment. However, temporal logic would suggest it is simply not possible to sequence such brief samples of the audio environment with any meaningful result. This project explores different audio capture and editing techniques to produce viable strategies for documenting the changes in a soundscape which accompany timelapse photography of the environment under study. Examples will be demonstrated, emphasizing the significant variables for adequately representing the elusive temporal characteristics of a soundscape.

10:45

2aNSa7. The effect of buffering noise in passenger car. Miguel Iglesias, Ismael Lengua, and Larisa Dunai (Dept. of Graphic Eng., Universitat Politècnica de València, St. Camino de Vera s/n 5L, Valencia 46022, Spain, inecav@inecav.com)

This paper presents the effect of buffering noise in passenger cars in real conditions at variable speed. At 25% of opened back window, the effect of buffering noise is not perceived. When increasing the aperture of the window to 50%, is perceived the creation of the buffering noise effect with resonance frequency = 15.6 Hz at speed = 80 km/h, but is not irritating because the pressure

level is low, pressure level = 110.2 dB. At 75% of the window aperture appears two air flows in the inside car: the one coming from the superior part of the superior zone of the inside and another one from the central zone of the inside. The effect of buffering noise is clearly perceived with resonance frequency = 16.4 kHz at speed = 80 km/h and pressure level = 119 dB and 19.5 Hz at speed = 110 km/h and pressure level at 122 dB. At speed higher than 103–140 km/h, the buffering noise disappears, depending on the car model. For Opel Corsa, the effect of buffering noises appears at the same speed range, whereas for Peugeot and BMW models, it appears three ranges, one from the buffering noise creation, persist in another range, and disappear in the third range.

11:00

2aNSa8. Innovative tools for urban soundscape quality: Real-time road traffic auralization and low height noise barriers. Alexandre Jolibois, Jérôme Defrance, Julien Maillard, Philippe Jean, and Jan Jagla (CSTB (Ctr. Scientifique et Technique du Bâtiment), CSTB, 24 rue Joseph Fourier, Saint-Martin-d’Hères 38400, France, alexandre.jolibois@cstb.fr)

Although noise exposure has been unanimously recognized for its impacts on people’s health, noise is still a current problem in many cities across the world. Besides, most noise analysis tools are based on long term average levels, which were initially meant to be used for inter-city infrastructures and are therefore insufficient to quantify the soundscape quality in urban environments. In this paper, we present some of the recent developments and results regarding two classes of innovative tools dedicated to the improvement of urban soundscape quality. First, recent advances in the field of real-time auralization of road traffic noise are presented and the benefit of realistic sound field restitution as a new analysis tool for the evaluation of urban planning projects is discussed and demonstrated. Second, some promising results regarding innovative urban noise reducing devices, taken among other works from the European project HOSANNA, are presented. In particular, the potential of vegetation and low height noise barriers are emphasized, from a numerical and experimental point of view. Results show that these devices can significantly enhance the soundscape quality in specific places, and therefore can help in the creation of so-called “quiet areas” within urban environments.

11:15–11:35 Panel Discussion

Session 2aNSb**Noise and Structural Acoustics and Vibration: Active Control of Sound and Vibration**

Scott D. Sommerfeldt, Cochair

Dept. of Physics, Brigham Young University, N181 ESC, Provo, UT 84602

Pegah Aslani, Cochair

*Physics and Astronomy, Brigham Young University, N203 ESC, Provo, UT 84602-4673***Chair's Introduction—9:00*****Invited Papers*****9:05**

2aNSb1. A feedback active noise control for in-ear headphone with robust on-line secondary path modeling. Young-cheol Park, Keun-sang Lee, and Youna Ji (Comput. & Telecomm Eng Div., Yonsei Univ., Changjo-hall 269, Yonseidae-ro 1, Wonju, Gangwon-do 220-710, South Korea, young00@yonsei.ac.kr)

The active noise control (ANC) technique can be used to improve the comfortness and quality of the music listening with an in-ear headphone. For a compact headphone design, the feedback ANC operated with the filtered-x LMS (FxLMS) algorithm is preferred. In practice, the error in the estimated secondary path can be problematic for the FxLMS algorithm, since it can lead the algorithm to an unstable state, and thus cause degradation of sound quality with the in-ear headphone. In this study, an adaptive residual music canceler (RMC) is proposed for enhancing the accuracy of the reference signal of the feedback ANC. Since RMC is designed to track the bias of the current secondary path estimate, the secondary path is continuously updated by combining the previous estimate of the secondary path with the current weight vector of RMC. In addition, variable step-size schemes are developed for both the control and secondary path estimation filters, which enable the ANC system to adapt quickly and robustly to the variation of the physical secondary path. Simulation results show that the secondary path can be accurately estimated and high quality of music sound can be consistently obtained in a time-varying secondary path situation.

9:25

2aNSb2. Real-time implementation of multi-channel active noise control systems for infant incubators. Lichuan Liu and Xianwen Wu (Electrical Eng., Northern Illinois Univ., 590 Garden Rd., Dekalb, IL 60510, liu@niu.edu)

High noise in infant incubators results in numerous adverse health effects, which have lifelong consequences for neonatal intensive care unit (NICU) graduates. This paper presents multi-channel feed forward active noise control (ANC) systems for infant incubators. Then, the proposed algorithm is implemented by using Texas Instrument TMS320 C6713. The real-time experiment results show that the proposed algorithm is effective in canceling the noises of infant incubators in real NICU environment

9:45

2aNSb3. Sweeping tunable vibration absorbers for structural vibration control. Paolo Gardonio and Michele Zilletti (DIEGM, Univ. of Udine, Via delle Scienze, 208, Udine 33100, Italy, paolo.gardonio@uniud.it)

Tunable vibration absorbers (TVAs) have been successfully implemented in distributed thin structures (e.g., panels shells) to control either tonal vibration, produced, for example, by unbalanced machinery, or low-frequencies broadband vibration, generated, for example, by stochastic pressure distributions due to diffuse sound fields or turbulent boundary layer fluid flows. This study is focused on the broadband control of structural vibration where often multiple TVAs are used to control the response of the natural modes that resonate in the controlled frequency band. Normally, each TVA is tuned to minimize the resonant response of a specific natural mode of the hosting structure by tuning the TVA natural frequency to the resonance frequency of the mode and setting the TVA damping ratio to properly dissipate energy. The proposed sweeping tuneable vibration absorbers (STVAs) have been conceived to operate on the multiple natural modes of the hosting structure where they are mounted by sweeping the TVA tuning frequency over the frequency band of control and simultaneously varying the TVA damping ratio to effectively dissipate energy over the desired frequency range. The operation principle of a single STVA is first discussed and demonstrated experimentally. The implementation of multiple STVAs is then examined.

10:05–10:25 Break

10:25

2aNSb4. Active control of cylindrical shells using the weighted sum of spatial gradients control metric. Pegah Aslani, Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N203 ESC, Provo, UT 84602-4673, pegah.aslani@gmail.com), and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Often it is desired to reduce the sound radiated from vibrating structures, including cylindrical shells. Active structural acoustic control (ASAC) provides a means to control the structural radiation at low frequencies efficiently and effectively. The technique of using the weighted sum of spatial gradients (WSSG) as a control metric has been developed previously for flat structures. This paper will investigate control of WSSG for cylindrical shells. There are specific features associated with WSSG that tend to provide effective control of radiated sound power. The method has also been shown to be quite robust with respect to error sensor location. The effectiveness of WSSG control has been investigated through the use of radiation modes for cylindrical shells, which allows us to determine the radiated sound power both before and after control. Results using WSSG control will be reported, along with comparisons of the results obtained with some of the other possible control approaches reported previously.

10:45

2aNSb5. Adaptive vibration control of a mechanical system with nonlinear damping excited by a tonal disturbance. Michele Zilletti (Inst. of Sound and Vib. Res., Univ. of Southampton, Via delle Scienze, 206, Udine 33100, Italy, michele.zilletti@uniud.it), Stephen J. Elliott, and Maryam Ghandchi Tehrani (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom)

In this study, an adaptive control system to reduce the tonal vibration of a mechanical system characterized by nonlinear damping is considered. Since the response of the system to a sinusoidal excitation is however mainly sinusoidal, nonlinear controllers are not required for the control of a tone in such a mechanical system. The adaptation and stability of an adaptive control algorithm, however, depends on the accuracy of the plant model. Since the tonal response of the nonlinear system changes with excitation level, conventional adaptive algorithms, with a fixed linear model of the plant, can be slow to converge and may not achieve the desired performance. The use of an online observer is proposed, to estimate the describing function model of the plant, which will vary with excitation level. This allows the adaptive control algorithm to converge more quickly than using a fixed plant model, although care has to be taken to ensure that the dynamics of the observer do not interfere with the dynamics of the adaptive controller.

Contributed Papers

11:05

2aNSb6. A study on analysis of room equalizer using smart modules for analog high-quality audio. Bae Seonggeon (Soongsil Univ., 1118-408, Mokdong Apt. 11 Dangi, Yangcheon Gu, Seoul 148-771, South Korea, sgbae123@gmail.com), Kim Jaepyung (Daelim Univ., Seoul, South Korea), and Rhee Esther (Keimyung Univ., DaeGu, South Korea)

This study proposes room equalizer, which uses movable and variable modules and revises frequency response, and is also applicable to a variety of places. When equalizing frequency responses, commonly used electric graphic equalizers or parametric equalizers have the same effect as the sound signals such as audio passing through electrical elements like condenser or coil, and phase shift of audio signal occurs at this time. Accordingly, they secure frequency response by adjusting and equalizing the characteristics of sound-absorbing materials without any distortion of the original sound. Therefore, this study proposes a room equalizer that does not use these electrical characteristics, but uses non-electronic spatial matching with an excellent sound persistency rate to enable its application to various changes in space.

11:20

2aNSb7. Thermoacoustic approach for the cloaking of a vibrating surface. Avshalom Manela and Leonid Pogorelyuk (Aerosp. Eng., Technion, Technion City, Haifa 32000, Israel, amanela@technion.ac.il)

A vibrating surface in a quiescent fluid transmits pressure fluctuations, which propagate into far-field sound. The vibroacoustic mechanism involved, coupling vibration and sound, is common in a large number of applications. Yet, there is a growing interest in developing means for achieving “acoustic cloaking” of an animated boundary. We suggest the heating of a surface, generating thermoacoustic perturbations, as a mechanism for monitoring vibroacoustic sound. Considering a setup of an infinite planar wall interacting with a semi-infinite expanse of an ideal gas, we

investigate the system response to arbitrary (small-amplitude) vibro-thermal excitation of the confining wall. Analysis is based on continuum Navier-Stokes-Fourier and kinetic Boltzmann equations, and supported by stochastic direct simulation Monte Carlo calculations. Starting with a case of a sinusoidally excited boundary, a closed-form solution is derived in both continuum and collision-free limits. The results, found valid at a wide range of frequencies, indicate that effective cloaking of the boundary may be achieved through optimal choice of boundary-generated heat flux. The results are rationalized by considering the quasistatic limit of low-frequency excitation. The analysis is then extended to consider the system response to impulse (delta-function) actuation, where the independence of cloaking conditions on perturbation frequency is demonstrated.

11:35

2aNSb8. Monitoring floor impact sounds using detection algorithms in multi-story residential buildings. Jin Yong Jeon, Joo Young Hong, Hansol Lim, and Muhammad Imran (Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, jyjeon@hanyang.ac.kr)

A monitoring and alarming system was designed to detect household noise occurrence in the multi-story residential buildings. This system is composed of a microphone combined with a home smart pad installed on a wall and a main server that can store data to discriminate household noise occurrence. Sound levels such as L_{Aeq} and L_{Amax} were calculated in real time using developed digital signal processor (DSP) of the acoustical parameter analyzer in the home smart pad. A correction equation was applied based on the standard measurement method to provide accurate measurements. After measurement, the results were saved in the main server using Ethernet and inter-compared with the data of each household to discriminate the household noise occurrence. Finally, the alarm was raised depending on the decision criteria based on both the regulations of Korean government and subjective evaluation results from the previous study.

Session 2aPA**Physical Acoustics: Wind Noise Mitigation in Atmospheric Acoustics**

W. C. Kirkpatrick Alberts, Cochair

US Army Research Laboratory, 2800 Powder Mill, Adelphi, MD 20783

JohnPaul Abbott, Cochair

*NCPA and Dept. of Physics and Astronomy, University of Mississippi, 122 PR 3049, Oxford, MS 38655***Chair's Introduction—8:00*****Invited Papers*****8:05****2aPA1. The prediction of infrasonic wind noise in forests.** Richard Raspet and Jeremy Webster (NCPA, Univ. of MS, PO Box 1848, NCPA, University, MS 38677, raspet@olemiss.edu)

Recent research on the measurement and prediction of the infrasonic wind noise measured in pine and mixed deciduous forests with and without leaves has led to a good understanding of the turbulence mechanisms that generate the infrasonic wind noise. This talk will review the success of the predictions of the wind noise from the measured horizontal velocity spectrum above the canopy and in the canopy as well as the wind velocity profile above and in the canopy. The wind profiles measured in this study were not typical of those published in the meteorology literature due to their limited height and sparse regular planting. The height dependence of the 3-D turbulence spectra also differ from the model spectra. The modifications necessary to produce predictions for more typical forests from the meteorological literature will be described. [Work supported by the U. S. Army Research Office under grant W911NF-12-0547.]

8:25**2aPA2. Experimental quantification of microphone windscreens and noise cones in a low speed wind tunnel.** Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technol. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbar-nar@mtu.edu)

Wind noise is a common challenge in all types of atmospheric acoustics as well as in wind tunnel applications in the aerospace and automotive industry. The dynamic pressure field measured by the microphone includes both the acoustic pressure and the pressure induced by the turbulent air flow over the diaphragm. There are three types of microphone accessories that are commonly used to perform wind noise isolation: protective grid-caps (no isolation), windscreens, and nose-cones. In this study, each of these microphone accessories is tested in a wind tunnel using 1/4" and 1/2" microphones in turbulent and laminar flows at flow speeds up to 55 mph and in both the head-on and parallel diaphragm orientations. Two types of flush mount microphones, a surface mount and a side-vented pressure microphone, are also evaluated. The effects of background flow noise on the microphone measurement is shown for all conditions and a case study is presented with an arbitrary acoustic source. Finally, recommendations are presented for which accessories are best used in different measurement situations.

8:45**2aPA3. Wind noise suppression using compact arrays of infrasound sensors.** William G. Frazier (Ste. 142, 850 Insight Park Ave., University, MS 38677, gfrazier@hyperiontg.com)

This presentation describes how a compact array (aperture much less than the shortest wavelengths of interest) of infrasound sensors can be used to significantly enhance signal detection and waveform estimation in the presence of high levels of wind noise without the use of unwieldy mechanical screening. The methodology's effectiveness is founded on the fact that wind noise can be highly correlated on short spatiotemporal scales. This correlation structure is adaptively estimated from data and is used to formulate a generalized likelihood ratio detection problem and a minimum mean squared error waveform estimation problem. The infrasound waveform is explicitly represented by a user-definable, parametrically characterized, stochastic prior distribution. Choice of this prior can enhance detection of anticipated signals and suppress others and thus more than one prior can be utilized in order to perform some level of classification. The presentation provides typical performance results from a range of application scenarios. Application to more traditional infrasound arrays is also presented, illustrating that exploitation of only the temporal correlation properties of wind noise is also beneficial in the context of the stochastic models and estimation techniques utilized.

9:05

2aPA4. Pipe systems for infrasound wind-noise reduction: Issues and approaches. Thomas B. Gabrielson (Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Wind-noise-reduction for infrasound monitoring below 10 Hz often takes the form of multiple-inlet pipe networks that average the field over distances small with respect to the acoustic wavelength but large with respect to the turbulence scale. There is a long tradition of pipe systems; however, traditional approaches can result in mediocre performance. Three aspects of the traditional design—inlets arranged in a regular geometric pattern, resistive resonance suppression, and equal-length inlet-to-sensor paths—can all be questioned. This paper reviews pipe-system performance with respect to design, modeling, and data quality. Waveguide models that account for the acoustic adiabatic-to-isothermal transition inside the pipes and summing junctions have been validated for prediction of complex frequency response and horizontal and vertical directionality. Techniques have also been developed for *in-situ* measurement of system frequency response and wind-noise-reduction effectiveness. The importance of on-site measurement is underscored by the number of performance problems uncovered at operational infrasound monitoring sites by in-situ techniques. These issues will be illustrated with measurements at operational infrasound-monitoring stations and at test-bed systems.

9:25

2aPA5. Long duration wind noise abatement measurements at the University of Mississippi Field Station. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu)

A series of wind noise abatement tests were performed at the University of Mississippi Field Station from January through June of 2014. These tests compared bare sensors to sensors treated with porous hoses, and domes of different dimensions (40, 48, 60, and 96 in.ch) that were covered either with foam or with metal meshing (30% and 45% opening). Because these measurements were obtained continuously, measurements were obtained over a wide range of weather conditions. As expected, the frequency of maximum wind noise attenuation decreased with increasing diameter of the dome, and the maximum attenuation obtained increased commensurately with diameter. Tests were also performed with nested domes (96 and 40 in.ch), which provided greater attenuation than just the larger of the two domes. The 30% opening mesh proved superior to the 45% opening mesh. Generally, the porous hoses provided superior wind noise protection below 1-Hz, but did not provide as much wind noise attenuation as was provided by the 96 and 40 in.ch nested wind dome configuration. We found that a number of issues (e.g., less robust, more affected by wet conditions) make this a less ideal solution than the metal mesh covered nested domes, for permanent installations.

Contributed Papers

9:45

2aPA6. Wind-induced noise based upon stability dependent turbulent velocity spectrum models. Carl R. Hart and D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, carl.r.hart@usace.army.mil)

The ambient sound level is an important datum for noise control, quantifying soundscapes, and remote detection. For low acoustic frequencies and fast enough wind speeds, wind-induced noise contributes largely to the measured ambient sound spectrum. Predictive models for wind-induced noise are based upon an empirical relationship between fluctuating pressure, mean wind speed, and the fluctuating streamwise wind velocity. A wind noise model developed by Van den Berg [Van den Berg, G. P., *J. Acoust. Soc. Am.* **119**(2), 824–833, 2006] is derived from a turbulent velocity spectrum, which is applicable for neutral atmospheric stability and flat homogeneous terrain. In this work, several other wind noise models are formulated based upon different turbulent velocity spectrum models. The turbulent velocity spectrum models take into account atmospheric stability, with varying degrees of complexity. Differences in predicted wind noise levels among the models show nearly a 3 dB spread for some atmospheric conditions. A recommendation is given for a specific wind noise model based on intercomparisons.

10:00–10:15 Break

10:15

2aPA7. Statistical properties of the mirror flow model of boundary-layer turbulence. Gregory W. Lyons and Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, NCPA, P.O. Box 1848, University, MS 38677-1848, gwlyons@go.olemiss.edu)

The mirror flow model is a construction of inhomogeneous, anisotropic turbulence from the superposition of a homogeneous, isotropic turbulent field with itself in transformed coordinates. This model has been used for atmospheric turbulence to predict spectral contributions to low-frequency wind noise [Yu *et al.*, *J. Acoust. Soc. Am.* **129**(2), 622–632 (2011)]. Here, analytical solutions are reported for the longitudinal and vertical turbulence spectra as a function of elevation. The correlation functions, integral

turbulence scale, and profiles of mean-square velocity components are computed. The premultiplied spectra show remarkable agreement with observed scaling in the atmospheric boundary layer. Implications for computation of wind noise spectra are discussed.

10:30

2aPA8. Effects of ground characteristics on wind-seismic coupling. Vahid Naderyan, Craig Hickey, and Richard Raspet (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, NCPA, 1 Coliseum Dr., University, MS 38677, vnaderya@go.olemiss.edu)

In seismic surveys, wind noise at low frequency seriously degrades seismic data quality. In order to find a solution for this problem, the driving pressure perturbations on the ground surface associated with wind-induced ground motions were investigated in a previous work [Naderyan, Hickey, and Raspet, *J. Acoust. Soc. Am.* **136**, 2139 (2014)]. Multiple triaxial 10 Hz geophones were deployed at different depths to study the induced ground velocity as a function of depth. The displacement amplitudes of the ground motions as a function of depth and frequency were predicted, where the ground was modeled as a homogeneous half-space elastic media. In the current work, the predictions are extended for inhomogeneous ground. The measurements are conducted at a different site with different characteristics. Since wind noise is dominant at lower frequencies, the triaxial 4.5 Hz geophones were used to investigate a lower frequency range. [This work was supported by USDA under award 58-6408-1-608.]

10:45

2aPA9. Comparisons of semi-porous fabric domes and cylindrical wind fence enclosures for infrasonic wind noise reduction. JohnPaul Abbott and Richard Raspet (NCPA and Dept. of Phys. and Astronomy, Univ. of MS, 122 PR 3049, Oxford, MS 38655, jrabbott@go.olemiss.edu)

This investigation compares the measured wind noise of two types of infrasonic wind noise reduction barriers. The first type is a set of 2.0 m diameter semi-porous fabric domes constructed and tested by the Army Research Laboratory [J. Acoust. Soc. Am. **136**, 2139 (2014)]. The type second is a set of single and multi-layer cylindrically shaped variable-porous

wind fence enclosures measuring 5.0 m and 10.0 m, constructed and tested by the National Center for Physical Acoustics [J. Acoust. Soc. Am. **134**, 4161 (2013)]. Preliminary investigations showed that for wind speeds between 3–6 m/s the measured power spectra densities of the fabric domes were comparable to the measured power spectra densities of the single layer, 5.0 m diameter wind fence enclosures at mid-range porosities. The measured power spectra densities of the fabric domes were higher than the measured power spectra densities for the 10.0 m diameter and multi-layer wind fences enclosures. For wind speeds greater than 6 m/s, the measured wind noise for the fabric domes was comparable to or lower than the measured wind noise for the wind fence enclosures.

11:00

2aPA10. A comparison of sintered materials for the reduction of wind noise in acoustic sensors. Latasha Solomon, David Gonski, Stephen Tenney, and Leng Sim (US Army Res. Lab, 2800 Powder Mill RD, Adelphi, MD 20783, latasha.i.solomon.civ@mail.mil)

Wind noise reduction has been of particular interest in the acoustic sensor arena for many years. Ideally, a desirable noise-mitigating medium is resilient to extreme temperatures and dust, as well as insects/animals. Open cell polyurethane foam is the most widely chosen material for acoustic wind noise reduction. These windscreens tend to provide protection from humidity and moisture, as well as reduction of wind flow noise at the frequencies of interest (500 Hz and below). However, after prolonged use in the environment, degradation of the windscreen and its performance is inevitable. Exposure to high levels of UV rays tends to make the windscreens brittle, while sand and dirt may become lodged in the pores requiring frequent cleaning or replacement, and they can be vulnerable to wildlife. Ideally, one would like to replace the current foam windscreen with a more durable, smaller, lower-profile medium providing similar wind noise rejection to that of foam windscreen. This research will compare the effectiveness of porous metal and plastic windscreens fabricated from sintered material to that of the conventional open cell foam windscreen in reducing wind generated flow noise.

11:15

2aPA11. New calculation of the turbulence-turbulence contribution to wind noise pressure spectra by incorporating turbulence anisotropy. Jiao Yu, Yanying Zhu (Liaoning Shihua Univ., 1 Dandong Rd. West Section, Wanghua District, Fushun, Liaoning 113001, China, yujiaojoy@hotmail.com), and Richard Raspet (National Ctr. for Physical Acoust., University, MS)

Turbulence-turbulence interaction and turbulence-shear interaction are the sources of intrinsic pressure fluctuation for wind noise generated by atmospheric turbulence. In previous research [Yu *et al.*, J. Acoust. Soc. Am. **129**(2), 622–632 (2011)], it was shown that the measured turbulent fields outdoors can be realistically modeled with Kraichnan's mirror flow model and turbulence-shear interaction pressure spectra at the surface were predicted and compared to measurements. This paper continues to apply Kraichnan's model and calculates the turbulence-turbulence interaction wind noise pressure spectra under anisotropic turbulence conditions. Different from turbulence-shear interaction, the fourth-order moments of the turbulence are needed in the calculation. A new calculation of the turbulence-turbulence contribution to the wind noise pressure spectra is compared to that for isotropic turbulence and the results are analyzed. [Work supported by the U. S. Army Research Office (Grant No. W911NF-12-0547) and the National Natural Science Foundation of China (Grant No. 11304137).]

11:30

2aPA12. Reduced-size windscreens for infrasound. W. C. Kirkpatrick Alberts, William D. Ludwig (US Army Reserach Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net), and Carrick Talmadge (National Ctr. for Physical Acoust., University, MS)

The long wavelengths of infrasound necessitate large, often complex, wind noise reduction apparatus, e.g., porous hose, pipe arrays, or wind fences. For the infrasonic sources of military interest that fall between approximately 2 and 20 Hz, 6-m radius porous hose rosettes are often sufficient windscreens. However, the number of hoses required for a typical 4-element infrasound array (16–24) reduces its portability. Based upon recent research demonstrating that porous fabric domes could outperform porous hoses in the 2–20 Hz region and a need for more compact windscreens, 1.2-m diameter domes covered in three types of porous fabric and one folding perforated aluminum dome were investigated for their suitability as alternatives to porous hose. Evaluations of wind noise reduction, transfer functions, and environmental durability will be presented for these alternative windscreens.

Session 2aPP

Psychological and Physiological Acoustics, Biomedical Acoustics, Speech Communication, and Signal Processing in Acoustics: Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention I

Michael Dorman, Cochair

Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85258

Blake S. Wilson, Cochair

Duke University, 2410 Wrightwood Ave., Durham, NC 27705

Chair's Introduction—8:30

Invited Papers

8:35

2aPP1. From W. House to the present: A brief history of cochlear implants. Michael Dorman (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85258, mdorman@asu.edu)

The history of cochlear implants can be divided into four stages. The first, in the early 1960s, was the period of the pioneers (e.g., House, Doyle, and Simmons) who found minimal interest, even disdain, from others for their work. The second stage, in the 1970s, saw a worldwide proliferation of groups who designed devices with multiple sites of cochlear stimulation and minimally effective, but very useful, speech coding strategies. The third period (1983–2003) saw the development of more effective processing strategies that utilized multiple sites of stimulation. These strategies, or their derivatives, are the ones in use today. Unfortunately, there have been no gains in group mean word understanding scores for patients fit with a single cochlear implant over a period of about 20 years. For that reason, in the current, fourth stage, researchers have improved performance by adding information (i) from residual low-frequency hearing (under 500 Hz) either in the implanted ear or in the ear contralateral to the implant or (ii) from a second implant. In this talk, I will view these stages from the standpoint of speech acoustics and the seemingly minimal amount of information that is sufficient to support high levels of speech understanding.

8:55

2aPP2. The punctuation mark in an equilibrium state: Modern signal processing. Blake S. Wilson (Duke Univ., 2410 Wrightwood Ave., Durham, NC 27705, blake.wilson@duke.edu)

As recently as the late 1980s, the speech reception performance of cochlear implants (CIs) was relatively poor compared to today's performance. As noted in the 1988 NIH Consensus Statement on CIs, only about 1 in 20 users of the best CIs at the time could carry out a normal conversation without the aid of lipreading. In contrast, the great majority of today's users can understand speech in relatively quiet conditions with their CIs and restored hearing alone. Indeed, most users communicate routinely via telephone conversations. As noted in the 1995 NIH Consensus Statement on CIs in Adults and Children, "A majority of those individuals with the latest speech processors for their implants will score above 80 percent correct on high context sentences, even without visual cues." Such abilities are a long trip from total or nearly total deafness. In this talk, I will (1) present some historical aspects; (2) describe the jump up in performance that was achieved in 1989 and provided the basis for many subsequent developments; and (3) mention some possibilities for further improvements in these already marvelous devices.

9:15

2aPP3. Additions to a single CI to improve speech understanding. René H. Gifford (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, 9302 MCE South, Nashville, TN 37232, rene.gifford@Vanderbilt.edu), Louise Loiselle (MED-EL, Durham, NC), Sarah Cook (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ), Timothy J. Davis, Sterling W. Sheffield (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN), and Michael F. Dorman (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Speech understanding was assessed in a simulated restaurant and cocktail party environment for 39 adult cochlear implant recipients (18 bilateral, 21 hearing preservation). Speech was presented as either collocated (S_0N_0) or spatially separated from the noise (S_0N_{0-360} or $S_0N_{90\&270}$). We calculated the benefit of adding a second CI, acoustic hearing from the non-implanted ear, and the implanted ear as well as spatial release from masking. The addition of either a second CI or binaural acoustic hearing yielded mean benefit of 13-percentage points. The additional benefit afforded via acoustic hearing in the implanted ear in addition to the bimodal configuration was an additional 10-percentage points. Little-to-no spatial release from masking was noted for either the simulated restaurant or the cocktail party environments. Spatial release from masking was not found to be significant at the group level nor was it different across the hearing configurations. We draw two primary conclusions: (1) adding either a second CI or binaural acoustic hearing to a single CI yields similar levels of benefit, but with different underlying mechanisms, and (2) the lack of spatial release is likely related to the presence of noise at both ears directed toward the ports of the HA and CI microphones.

9:35

2aPP4. Inserting the speech signal above the cochlea: How can this work? Colette M. McKay (Bionics Inst., 384-388 Albert St., East Melbourne, Victoria 3002, Australia, cmckay@bionicsinstitute.org)

Although cochlear implants (CIs) have successfully restoring hearing to many deaf people, there remains a subgroup who cannot benefit from them due to malformation of the cochlea or damage to the cochlear nerve. Auditory brainstem implants (ABIs) and auditory midbrain implants (AMIs) aim to restore hearing to this group. ABIs stimulate the surface of the cochlear nucleus (CN). Despite some ABI users gaining benefit similar to a CI, the majority achieve poor speech understanding with the device alone. The reasons for poor outcomes are not fully understood but include tumor damage to particular cells in the cochlear nucleus, surgical approach, and placement of the electrodes. One factor in poor outcome is the difficulty of stimulating specific tonotopic planes within the CN. This factor and tumor damage to the CN were reasons for the development of the AMI, which directly stimulates the tonotopic layers of the inferior colliculus (IC) using a shank electrode. Psychophysics and physiological studies have thrown light on the way that new electrode designs and processing strategies may provide patients with AMIs and potentially ABIs with hearing commensurate with CIs. [The Bionics Institute acknowledges the support it receives from the Victorian Government through its Operational Infrastructure Support Program.]

9:55

2aPP5. Neuroplasticity in deafness: Evidence from studies of patients with cochlear implants. Anu Sharma (Speech Lang. and Hearing Sci., Univ. of Colorado at Boulder, 2501 Kirtredge Loop Rd., Boulder, CO 80309, anu.sharma@colorado.edu)

Deafness alters the normal connectivity needed for an optimally functioning sensory system—resulting in deficits in speech perception and cognitive functioning. Cochlear implants have been a highly successful intervention because they bypass cochlear damage and directly stimulate the auditory nerve and brain, taking advantage of high degree of neuroplasticity of the auditory cortex. Deaf adults and children who receive intervention with cochlear implants have provided a unique platform to examine the trajectories and characteristics of deprivation-induced and experience-dependent neuroplasticity in the central auditory system. I will describe changes in neural resource allocation secondary to hearing impairment, cross-modal cortical re-organization from the visual and somatosensory modalities, and the multimodal and cognitive re-organization that results from auditory deprivation. Overall, it appears that the functional activation of cognitive circuitry resulting from cortical reorganization in deafness is predictive of outcomes after intervention with electrical stimulation. A better understanding of cortical functioning and reorganization in auditory deprivation has important clinical implications for optimal intervention and re-habilitation of cochlear implanted patients. [Work supported by NIH.]

10:15

2aPP6. Language emergence in early-implanted children. Ann E. Geers (Behavioral and Brain Sci., Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, ageers@utdallas.edu), Johanna G. Nichols (Otolaryngol., Washington Univ., St. Louis, MO), Emily A. Tobey (Behavioral and Brain Sci., Univ. of Texas at Dallas, Dallas, TX), and Lisa Davidson (Otolaryngol., Washington Univ., St. Louis, MO)

This study develops a model to differentiate early-implanted children with preschool language delays that persist into the elementary grades from children who achieve normal language as they gain auditory, linguistic, and academic experience. A nationwide sample of 60 children with congenital profound deafness, auditory-oral education, and early cochlear implantation (12–36 months) was assessed at average ages of 3.5, 4.5, and 10.5 years. Children were classified with: Normal Language Emergence (NLE: N = 19), Late Language Emergence (LLE: N = 22), or Persistent Language Delay (PLD: N = 19) based on standardized language test scores. Children with PLD did not differ from children with LLE on a number of variables that have previously been associated with post-implant outcome, including pre-implant hearing, age at first CI, gender, maternal education and nonverbal intelligence. Logistic regression analysis identified variables that predicted whether children with early language delay would catch up with hearing age-mates: (1) preschool speech and language level, (2) ear placement of first CI, (3) upgrades in speech processor technology, and (4) CI aided thresholds. A majority of the 60 children in this study reached age-appropriate levels of language, reading, and verbal reasoning and were placed in primary grades with hearing age-mates.

10:35–10:50 Break

10:50

2aPP7. Pitch perception and representations of frequency in the peripheral auditory system: What's missing in cochlear implants? Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Pitch is a crucial feature of auditory perception. It conveys information about melody and harmony in music, and conveys prosodic and (in tone languages) lexical information in speech. Although pitch has been studied formally for many decades, there still remains considerable uncertainty regarding how frequency and pitch information is coded and represented in the auditory periphery. This uncertainty means it is unclear whether and how pitch information can best be conveyed via cochlear implants. Cochlear-implant users have been shown to be sensitive to changes in the place of stimulation within the cochlea (“place” pitch) and to changes in the pulse rate presented to the electrode (“rate” pitch), at least up to rates of around 300 Hz. Place pitch in cochlear implants is probably best compared to “brightness” and rate pitch has most in common with the relatively weak pitch produced by temporal-envelope fluctuations in normal-hearing listeners. Neither type of pitch seems to provide the salient and accurate pitch sensations associated with normal hearing. Studies in both normal-hearing listeners and cochlear-implant users will be reviewed to shed light on the mechanisms of pitch perception and on the challenges involved in restoring pitch to cochlear-implant users. [Work supported by NIH grant R01DC005216.]

2a TUE. AM

11:10

2aPP8. The sound of a cochlear implant: Lessons learned from unilaterally deaf patients fit with a cochlear implant. Michael Dorman, Sarah Cook (Dept. of Speech and Hearing Sci., Arizona State University, Tempe, AZ 85258, mdorman@asu.edu), and Daniel Zeitler (Denver Ear Assoc., Englewood, CO)

Starting about 10 years ago in Europe, patients who had one normally hearing ear and one deafened ear were fitted with cochlear implants in an effort to suppress tinnitus in the deafened ear. That effort was very successful. More recently, single sided deaf patients without tinnitus have been implanted in an effort to improve health-related quality of life. This effort, too, has been successful. This patient population, with one normal hearing ear and one implanted ear, offers an unprecedented window into the sound quality of a cochlear implant. That is, these patients can tell us whether a candidate stimulus presented to the normally hearing ear sounds like the stimulus presented to the cochlear implant. In this talk, we describe our efforts to reproduce the sound of a cochlear implant.

TUESDAY MORNING, 19 MAY 2015

KINGS 3, 9:00 A.M. TO 11:30 A.M.

Session 2aSA

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

Christina J. Naify, Cochair

Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Michael R. Haberman, Cochair

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

Invited Papers

9:00

2aSA1. Transforming acoustic sensing with high refractive index metamaterials. Miao Yu and Yiongyao Chen (Mech. Eng., Univ. of Maryland, 2181 Glenn Martin Hall, College Park, MD 20742, mmyu@umd.edu)

Acoustic sensors play an important role in many areas, such as safety (e.g., sonar arrays), public health (e.g., ultrasonic imaging), surveillance (e.g., underwater communication and navigation), and industry (e.g., non-destructive damage detection). However, conventional acoustic sensors inevitably suffer from the fundamental pressure detection limit, which hinders the performance of current acoustic sensing technologies. Here, we design high refractive index acoustic metamaterials that have strong wave compression effect to obtain direct amplification of pressure fields in metamaterials. This enables a novel metamaterial enhanced acoustic sensing system that can help overcome the detection limit of conventional systems. Through analytical, numerical, and experimental studies, we demonstrate that this novel acoustic sensing system can help achieve over an order of magnitude enhancement in acoustic pressure detection limit. This will allow the detection of weak acoustic signals below the noise floor of the conventional acoustic sensing systems. This work is expected to impact many fronts that require high performance acoustic sensing.

9:20

2aSA2. Imaging with reconfigurable active acoustic metamaterials. Bogdan Popa and Steven Cummer (Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, bogdan.popa@duke.edu)

We present a design procedure to obtain active metamaterial structures that cover a large range of linear and non-linear acoustic responses, and whose flexibility makes it suitable to numerous applications. The presentation will briefly outline the types of material properties enabled, and then focus on the potential of this approach to various imaging applications. The design method is illustrated experimentally using a very thin metamaterial slab whose functionality is set electronically. This ensures that multiple imaging techniques can be demonstrated using the same physical slab by simply swapping the electronic modules that determine the metamaterial behavior. Using this method, we implement a thin acoustic lens whose focal length can be controlled at will. In addition, we show how time reversal imaging techniques can be implemented in real-time and in a continuous manner. Finally, the slab is configured to act as an acoustic multiplexer that takes sound incident from multiple directions and funnels it in exactly one direction.

2aSA3. Acoustic meta-structures based on periodic acoustic black holes. Fabio Semperlotti and Hongfei Zhu (Aerosp. and Mech. Eng., Univ. of Notre Dame, 374 Fitzpatrick Hall, Notre Dame, IN 46556, fsemperl@nd.edu)

This paper presents a class of three-dimensional fully isotropic acoustic metamaterials that support the development of load-bearing thin-walled structural elements with engineered wave propagation characteristics, i.e. the meta-structures. This class of metamaterials is based on the concept of Acoustic Black Hole (ABH) that is an element able to bend and, eventually, trap acoustic waves. A periodic lattice of ABHs enables wave propagation characteristics comparable with resonant metamaterials but without the need and the fabrication complexity associated with the use of either multi-material or locally resonant inclusions. The dispersion characteristics show the existence of several interesting phenomena such as strong mode coupling, zero group velocity points on the fundamental modes, and Dirac points at the center of the Brillouin zone. Numerical simulations, conducted on a meta-structure made of a thin metal plate with an embedded slab of ABH material, show the existence of a variety of wave propagation effects (e.g., collimation and bi-refraction) in an extended operating range. Such an engineered material can pave the way to the design of thin-walled adaptive structures with fully passive wave management capabilities.

10:00–10:30 Break

Contributed Papers

10:30

2aSA4. Ultrasonic subwavelength focusing above two dimensional membrane metasurface using time reversal. Shane W. Lani, Karim Sabra, and F. Levent Degertekin (Georgia Tech, 801 Ferst Dr., Atlanta, GA 30332, shane.w.lani@gmail.com)

Surface acoustic waves can propagate above immersed membrane arrays, such as of capacitive micromachined ultrasonic transducers (CMUTs). Similar waves on metamaterials and metasurfaces with rigid structures (typically in the kHz range) have been studied and used for tunable band gaps, negative refraction, and subwavelength focusing and imaging. This work demonstrates through simulation and experiments that a 2D membrane array can be used for subwavelength focusing utilizing a time reversal method. The studied structure consisted of the focusing region, which is a dense grid of 7x7 membranes (6.6 MHz resonance) that support the slow surface acoustic waves. Eight additional membranes are located on the same surface outside the focusing region. Subwavelength focusing was performed by using a time reversal method in which the external eight membranes were used as excitation transducers. Modeling results were verified with experiments that were performed with the membranes being actuated electrostatically and the membrane displacements were measured with a laser Doppler vibrometer. Subwavelength focusing ($\lambda/5$) was achieved on the metasurface while a modal decomposition of the spatial focus from an iterative time reversal method was done to illustrate that optimal focusing resolution requires efficient excitation of the mode shapes containing sub-wavelength features.

10:45

2aSA5. Enhanced acoustic transmission through a slanted grating. Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

It is known that an acoustic wave incident on an infinite array of aligned rectangular blocks of a different acoustic material exhibits total transmission if certain conditions are met which relate the unique "intromission" angle of incidence with the geometric and material properties of the slab. This extraordinary acoustic transmission phenomenon holds for any thickness of the slab, making it analogous to a Brewster effect in optics, and is independent of frequency as long as the slab microstructure is sub-wavelength. Here, we show that the enhanced transmission effect can be obtained in a slab with parallel grating elements oriented obliquely to the slab normal. The dependence of the intromission angle θ_i is given explicitly in terms of the orientation angle. Total transmission is also achieved at incidence angle $-\theta_i$ although there is a relative phase shift between the transmitted amplitudes of the $+\theta_i$ and $-\theta_i$ cases. These effects are easily understood when the grating elements are rigid. In particular, any angle of intromission can be obtained by with thin rigid elements by orienting them to the desired value of θ_i .

11:00

2aSA6. Effect of waveguide impedance on radiation in underwater acoustic leaky wave antennas. Christina J. Naify (Acoust., Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify@nrl.navy.mil), Matthew Guild (National Res. Council, Washington, DC), David C. Calvo, and Gregory J. Orris (Acoust., Naval Res. Lab, Washington, DC)

Leaky wave antennas (LWAs) have been examined for decades as a way to steer electromagnetic waves as indexed by input frequency, enabling rapid video scanning of an environment with a simple compact device. Recently, an LWA device constructed of a rigid waveguide with open shunts was shown to produce a similar steering effect for acoustic waves in air [Naify *et al.*, Appl. Phys Lett. **102**, 203508 (2013)]. The shunts serve two purposes, to couple acoustic energy from the waveguide to the surrounding area, as well as control directionality of the radiated wave by changing the impedance, and thus modulating the phase speed of the waves in the waveguide. While this impedance contrast of the waveguide wall relative to that of the surrounding fluid is very large for an air background, when designing a structure for use in water the material of the waveguide can drastically change the directionality of the radiation profile. This study examines the effect of waveguide wall impedance on directionality of an acoustic LWA using both numerical and finite element methods. Three material impedance conditions are examined, including acoustically rigid, and two materials with finite impedance. [Work sponsored by the Office of Naval Research.]

11:15

2aSA7. Effect of air volume fraction and void size on the performance of acoustically thin underwater noise isolation panels. Ashley J. Hicks, Michael R. Haberman, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Office # N642, Austin, TX 78758, pswilson@arlab.utexas.edu)

We present experimental measurements of the acoustic behavior of underwater sound absorbing and isolating panels composed of three deeply sub-wavelength layers. The panels were constructed with an inner layer of Delrin plastic containing through-thickness circular air-filled holes sandwiched between two rubber layers. Panel isolation efficacy was quantified by measuring insertion loss for non-dimensional panel thicknesses, kd , ranging from 7.2×10^{-3} to 8.5×10^{-2} . Finite element numerical simulations will be discussed which indicate that the panel structure mimics a planar encapsulated bubble screen exactly one bubble thick but displays measured performance that is significantly more broadband. Experimental and numerical results indicate that panels with kd ranging from 2.8×10^{-2} to 7.2×10^{-2} can generate a frequency-averaged insertion loss ranging from 5 to 12 dB, dependent on air volume fraction in the panel. The effect of individual air pocket resonance frequency on insertion loss will also be discussed. [Work supported by ONR.]

Session 2aSC

Speech Communication: General Topics in Speech Production (Poster Session)

Brad H. Story, Chair

Speech, Language, and Hearing Sciences, University of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 8:15 a.m. to 9:45 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m. There will be a 15-minute break from 9:45 a.m. to 10:00 a.m.

Contributed Papers

2aSC1. Experimental analysis of intraglottal flow and geometry. Liran Oren, Sid Khosla, and Ephraim Gutmark (Otolaryngol., Univ. of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

During phonation, skewing of the glottal area and glottal flow refers to phenomenon that occurs when the area or the flow decelerates more rapidly than it accelerates. This skewing is clinically important because it increases the glottal efficiency (defined by the acoustic intensity divided by the subglottal pressure). Current theoretical models predict that skewing of the area is not directly proportional to skewing of the flow due to the affects of vocal tract inertance. Recently, we have developed a method that can measure air-flow velocity, the distance between the folds, and the glottal flow rate in excised canine larynx. In the current study, PIV measurements are taken in five excised canine larynges having all the structures above the vocal folds removed. The results show strong correlation between the maximum closing rate at the superior aspect of the folds and the lowest negative pressure that is computed inside the glottis during the closing phase. The results also show that the magnitude of the divergence angle that develops in the glottis during closing is directly proportional to the glottal efficiency. These are important observations because they hold new mechanisms that can increase glottal efficiency without the affects of vocal tract inertance.

2aSC2. Sufficiency of a four-parameter spectral model of the voice source. Jody Kreiman, Bruce R. Gerratt, Rosario Signorello, and Shaghayegh Rastifar (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, jkreiman@ucla.edu)

Our proposed perceptually motivated spectral-domain model of the voice source comprises spectral slopes in four ranges (H1-H2, H2-H4, H4-the harmonic nearest 2 kHz, and that harmonic to the harmonic nearest 5 kHz). Previous studies established the necessity of these parameters by demonstrating that listeners are sensitive to all of these parameters, and that all are needed to adequately model voice quality. The present study examines the sufficiency of the model to quantify voice quality by applying it to a very large set of voices. To determine the range of phenomena for which the source model is valid, 200 natural voices (half male; 50 normal/mildly pathologic, 30 moderately pathologic, 20 severely pathologic) were copy-synthesized using the model, and adequacy of synthesis was assessed in a listening test by comparing the synthetic tokens to the natural targets. Validity of the model as a function of severity of pathology and the particular manner in which the stimuli deviate from normal will be discussed. Sources of mismatches between synthetic and natural stimuli will be identified, and their status with respect to definitions of voice quality will be addressed. [Work supported by NIH.]

2aSC3. Hysteresis in an extension of Titze's surface wave model and contact with phonation onset/offset experiments. Lewis Fulcher (Phys. & Astronomy, Bowling Green State Univ., Ridge Ave., Bowling Green, OH 43403, fulcher@bgsu.edu) and Ronald Scherer (Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

Three modifications of Titze's surface wave model are required to incorporate the property of hysteresis. (1) Adding a nonlinearity to the damping term of this model allows it to undergo sustained oscillations above the onset threshold, and thus the model becomes more robust. (2) Adding a nonlinearity to the stiffness term gives the model a frequency dependence that allows contact with the dynamic viscosity measurements carried out by Chan and Rodriguez [J. Acoust. Soc. Am. **124**, 1207–1219 (2008)]. These measurements observed a reduction in the dynamic viscosity as the frequency of oscillation increased. In order to include this property in the surface wave model, (3) the size of the damping constant is reduced whenever the energy of the oscillating vocal fold exceeds a certain value. Thus, the nonlinear properties introduce a dependence of the damping term upon the oscillation history of the vocal fold and set the stage for hysteresis effects. Calculations with the extended surface wave model are compared with the observed onset/offset pressures measured by Titze, Schmidt, and Titze [J. Acoust. Soc. Am. **97**, 3080–3084 (1995)] and by Chan and Titze [J. Acoust. Soc. Am. **119**, 2351–2362 (2006)]. The extended model readily provides an explanation for why the observed phonation offset pressures are always lower than the onset pressures as well as several other trends in the data.

2aSC4. Respiratory-laryngeal coordination in airflow conservation and reduction of respiratory efforts of phonation. Zhaoyan Zhang (Head and Neck Surgery, UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zy Zhang@ucla.edu)

Although many previous studies investigated the effects of respiratory-laryngeal coordination during voice production, the importance of respiratory-laryngeal coordination to airflow conservation and reducing respiratory efforts of phonation has not been systematically investigated. In this study, a computational model of the pressure-volume-flow relationship in the respiratory system is developed and coupled to a three-dimensional continuum model of phonation. Simulations at different conditions of the lung initiation volume and laryngeal resistance show that small laryngeal resistance, as observed in vocal fold paralysis or soft voice conditions, leads to large airflow rate out of the lungs and thus below-normal duration of speech between breaths. Although the speech duration can be increased by an increase in either the lung initiation volume or laryngeal resistance, increased laryngeal resistance also reduces the respiratory effort required to maintain a target subglottal pressure. The demand for laryngeal resistance increase in order to maintain a normal breath group duration increases when phonating at a lower lung initiation volume or when a higher subglottal pressure is desired. Finally, the simulations show that tightening vocal fold approximation is more effective than vocal fold stiffening in increasing speech duration between breaths and reducing respiratory effort of phonation. [Work supported by NIH.]

2aSC5. Fundamental frequency movements in one-word imperatives. Sergio Robles-Puente (World Lang., Literatures and Linguist, West Virginia Univ., Chitwood Hall, P.O. Box 6298, Morgantown, WV, Morgantown, WV 26506-6298, seroblespuente@mail.wvu.edu)

F0 contours in Spanish declarative and imperative sentences have traditionally been described as identical and it has not been until recently that several phonetic variables have been identified as markers of imperativity. These include higher overall F0s, early peak alignments in pre-nuclear pitch-accents, upstepped nuclear pitch-accents and marginal use of high boundary tones. Since previous analyses have concentrated on utterances with multiple pitch-accents, not much is known about productions with just one word. This study focuses on one-word utterances given the tendency of imperatives to be short and taking into account that some of the aforementioned phonetic markers cannot be used with a sole pitch-accent. The analysis of 117 declarative sentences and 256 imperatives produced by eight Spanish speakers demonstrated that in cases where declaratives and imperatives share the same contour (L+>H*L%), the latter tend to have higher F0 peaks (Wilcoxon signed-rank test; $p < 0.000$). Besides, 32.4% of the imperatives showed F0 contours ending in high boundary tones, a marker not used in declaratives and only marginally found in imperatives with multiple pitch-accents. Current findings suggest that when segmental material limits F0 movements, speakers look for alternative phonetic strategies to distinguish between declaratives and imperatives.

2aSC6. Indirect evidence of perturbation leads to changes in production of voice amplitude and fundamental frequency. Elizabeth D. Casserly (Psych., Trinity College, 300 Summit St., Life Sci. Ctr., Hartford, CT 06106, elizabeth.casserly@trincoll.edu), Lily Talesnick (Neurosci., Trinity College, Hartford, CT), and Nicholas Celestin (Psych., Trinity College, Hartford, CT)

In this study, we explore the juncture of real-time feedback-based changes in speech production and those initiated by top-down knowledge of external factors such as audience characteristics or audience listening conditions. Twenty-four speakers were asked to solve interactive puzzles via remote auditory connection with an experimenter under five signal transmission conditions: normal/baseline, increased signal amplitude, decreased amplitude, +100 cents shift in pitch, and -100 cents shift in pitch. Evidence of these changes was available to talkers only through the speech of the experimenter; no sidetone was introduced in the connection and listening conditions for talkers were otherwise normal. In response to hearing amplitude shifts in their partner's speech, 19/24 talkers significantly altered their mean voice amplitude, while 6/24 altered f0 in response to shifted experimenter vocal pitch (RM-ANOVA, all corrected p 's < 0.05). Approximately 30% of responses countered the signal manipulation (e.g., f0 increase in response to low experimenter f0), while the remainder imitated the observed change. This combination of perturbation-like compensation and imitation/accommodation suggests that speakers can interpret transmission circumstances differently, or pursue different speech-related goals, during even very simple, constrained tasks.

2aSC7. Effect of "only" on prosodic focus marking. Elizabeth Chalmers and Jonathan Howell (Linguist, Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, chalmers1@mail.montclair.edu)

This study seeks to determine how the inclusion of the focus-sensitive word "only" affects the focus-marking prosody of a sentence. Twenty-one native English speaking adults read 8 question-answer pairs. The questions were written to elicit focus on an object NP (e.g., Who did you see?). In one condition, the answer contained only (e.g., I only saw Mary); in the other condition, "only" was omitted (e.g., I saw Mary.). The recordings were annotated in Praat using forced alignment. We performed linear residualization of F0, amplitude and duration (cf. Breen *et al.* 2009) to remove effects of item and participant. Statistical models of residual pitch and duration on object-NP and verb failed to show any significant differences between the sentences that contain "only" and those not containing "only". These results fail to support theories of utterance-level prominence, which posit a categorical distinction between presentational and contrastive focus (e.g., Katz and Selkirk 2011).

2aSC8. Fathers' use of fundamental frequency in motherese. Mark Vandam, Paul De Palma, and William E. Strong (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu)

Studies of motherese or child-directed speech (CDS) have paid scant attention to fathers' speech when talking to children. This study compares mothers' and fathers' use of CDS in terms of fundamental frequency (F_0) production, examining natural speech from a very large database of hundreds of hours of family speech including mothers, fathers, and preschool children. The day-long recordings are collected with specialized recording software and body-worn hardware, then analyzed with automatic speech recognition (ASR) technology (LENA Research Foundation, Boulder, CO). CDS is defined here as speech in a conversational exchange between the parent and child, and adult-directed speech (ADS) is speech between adults or speech in which the child is not a (vocal) participant. Results confirm many reports in the literature of mothers' increased F_0 during CDS. Results fail to show a difference in the F_0 characteristics between fathers' CDS and ADS speech. This shows that children's linguistic experience with fathers is different than with mothers. This result could be useful to improve ASR techniques and better understand the role of usage in natural language acquisition and the role fathers play in the language acquisition process.

2aSC9. Acoustic correlates of creaky voice in English. Sameer ud Dowla Khan, Kara Becker (Linguist, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, skhan@reed.edu), and Lal Zimman (Linguist, Stanford Univ., Stanford, CA)

We compared auditory impressions of creaky voice in English to acoustic measures identified as correlates of contrastive voice qualities in other languages (e.g., Khmer, Chong, Zapotec, Gujarati, Hmong, Trique, and Yi). Sixteen trained linguistics undergraduates listened to the IP-final word "bows" produced five times each by five American English speakers reading the Rainbow Passage, and gave a rating from 0 (no creak) to 5 (very creaky). Results show that stronger auditory impressions of creak are significantly correlated with lower f0, lower cepstral peak prominence (CPP), lower harmonics-to-noise ratios (HNR), and higher subharmonics-to-harmonics ratio (SHR). This suggests that listeners perceive greater creakiness as the voice becomes lower pitched, less periodic, and more audibly interspersed with subharmonic frequencies (i.e., diplophonia). Notably, none of the spectral amplitude measures proposed as acoustic correlates of glottal configurations for creaky voice in other languages (e.g., lower H1-H2 for smaller open quotient, lower H1-A1 for smaller posterior aperture, lower H1-A3 for more abrupt closure, etc.) was significantly correlated with these judgments in any expected direction. Taken together, these results suggest that while listeners consistently use pitch and periodicity as cues to creak, speakers might be varying in their articulatory strategies to achieve those acoustic effects.

2aSC10. Detecting palatalization in spontaneous spoken English. Margaret E. Renwick (Linguist Program, Univ. of Georgia, University of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu) and Caitlin N. Cassidy (Inst. for Artificial Intelligence, Univ. of Georgia, Athens, GA)

We present an analysis of palatalization from /s/ to [ʃ] at word boundaries in UK English. Previous work has considered the effect of lexical frequency (LF) on this phenomenon, but without combining acoustics and spontaneous speech in one study, which we undertake using data gathered from the Audio BNC (<http://www.phon.ox.ac.uk/AudioBNC>). We analyze 5,259 word pairs in five phonological contexts, comparing the acoustics of test tokens subject to palatalization (e.g., /s/ in *miss you*), to control tokens containing non-alternating [s] (*miss it*) or [ʃ] (*mission, wish it, wish you*). Word and segment boundaries were obtained via forced alignment, but hand-checked. We measured the spectral moments and duration of each fricative; following vowel duration and formant values were also extracted. LF was calculated using the Audio BNC. We find that the spectral center of gravity (CoG) of [s] before [j] (*miss you*) is intermediate ($p < 0.01$) between those of [s] and [ʃ]. Furthermore, CoG of test tokens correlates negatively with LF, indicating increased palatalization in high-frequency contexts; no control pair exhibits such correlations. LF predicts CoG even considering speaker gender and other phonetic co-predictors. This supports the view of palatalization as gestural overlap, which increases with LF in casual or fast speech.

2aSC11. Coarticulation and speech stability for three lifespan groups.

Alissa Belmont, Stefan A. Frisch, and Nathan Maxfield (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD1017, Tampa, FL 33620, abelmont@mail.usf.edu)

Using ultrasound recording of tongue movement during speech production, repetitions of kV initial monosyllabic words were measured for nine vowel contexts in three different age groups (children 8–12, young adults 18–35, and older adults 50–65). Coarticulation was measured by average minimum point-to-point distance between tongue traces (Zharkova & Hewlett 2009). Speech stability was quantified by average distance between productions with the same vowel. Coarticulation was quantified by average distance between productions with different vowels. For all three talker groups ($n = 11$ or greater per group), the measure of coarticulation was greater than the measure of stability, demonstrating coarticulatory differences. Across the lifespan, measures of coarticulation and stability both decreased from group to group. This suggests that speech production becomes more segmental (less coarticulated) and more stable (less variable) continuously throughout the lifespan of speaking experience.

2aSC12. Influences of word frequency and response delay on imitation of visual speech.

James W. Dias and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, 900 University Ave., Riverside, CA 92521, jdias001@ucr.edu)

Human perceivers unconsciously imitate the subtle articulatory characteristics of perceived speech (*phonetic convergence*) during live conversation [e.g., Pardo, 2006] and when shadowing (saying aloud) pre-recorded speech [e.g., Goldinger, 1998]. Perceivers converge along acoustical speech dimensions when shadowing auditory (heard) and visual (lipread) speech [Miller, Sanchez, & Rosenblum, 2010], suggesting the information to which perceivers converge may be similar across sensory modalities. It is known that phonetic convergence to auditory speech is reduced when shadowing high-frequency words and when shadowing responses are delayed. These findings suggest that stored lexical representations and working memory processing time may modulate speech imitation [Goldinger, 1998]. The question arises of whether phonetic convergence to shadowed *visual* speech would demonstrate the same effects of word frequency and shadowing response delay. Phonetic convergence was reduced when shadowing lipread high-frequency words, suggesting lexical information may be similarly accessed by auditory and visual information, consistent with the effects of word frequency on auditory and visual speech comprehension [e.g., Auer, 2002, 2009]. However, where shadowing delay reduced phonetic convergence to auditory speech, it *increased* phonetic convergence to visual speech. The results are discussed with respect to possible differences in processing of auditory and visual speech information across response delays.

2aSC13. Individual differences in phonetic drift by English-speaking learners of Spanish.

Marie K. Huffman (Linguist, Stony Brook Univ., SBS S 201, SBS S 201, Stony Brook, NY 11794-4376, marie.huffman@stonybrook.edu) and Katharina Schuhmann (Freie Universität Bozen - Libera Università di Bolzano, Bolzano, Italy)

Second language learning affects L1 pronunciation even in early stages of adult L2 acquisition (e.g., Guion 2003, Chang 2012), an effect sometimes called “phonetic drift.” English voiceless stops assimilate to Korean stops in novice L1 English-L2 Korean learners in Korea, arguably due to English voiceless stops and Korean aspirated stops being linked to the same phonological category (Chang 2012), and the “novelty” of the Korean language input with its longer VOTs (Chang 2014). We tested whether novice L1 English-L2 Spanish learners in the United States also show assimilatory L2-to-L1 effects. Stronger typological and orthographic similarities might lead speakers to equate English and Spanish stops, but phonetic differences—short-lag VOT on voiceless stops and pre-voicing of voiced stops in Spanish—might impede L2-to-L1 assimilation. Monolingual English college-students were recorded every other week for the first 12 weeks of an introductory Spanish course. VOT data show considerable individual differences—some subjects show assimilation, some dissimilation, between English and Spanish stops. This suggests that while orthography and typological similarity may play a role, how a learner’s first and second language interact in novice adult L2

acquisition is also affected by individual factors such as perceptual sensitivity to phonetic differences, sociolinguistics considerations, and possibly speaker strategies for maintaining contrast (e.g., Nielsen 2011).

2aSC14. Auditory feedback perturbation of vowel production: A comparative study of congenitally blind speakers and sighted speakers.

Pamela Trudeau-Fisette, Marie Bellavance-Courtemanche, Thomas Granger, Lucile Rapin, Christine Turgeon, and Lucie Ménard (Département de linguistique, Université du Québec à Montréal, C.P. 8888, Succursale Centre-Ville, Montréal, Quebec H3C 3P8, Canada, trudeau-fisette.pamela@courrier.uqam.ca)

Studies with congenitally blind speakers show that visual deprivation yields increased auditory discrimination abilities as well as reduced amplitude of labial movements involved in vowel production, compared with sighted speakers. To further investigate the importance of auditory and visual feedback in speech, a study of auditory perturbation of rounded vowels was conducted in congenitally blind and sighted French speakers. Acoustic and articulatory (electromagnetic articulography) recordings from ten congenitally blind speakers and ten sighted speakers were obtained during the production of the French rounded vowel /ø/. All participants were first asked to produce the vowel repeatedly in a normal condition, i.e., with regular auditory feedback. In the perturbed condition, participants received, in real-time through headsets, an altered version of their speech, in which F2 was gradually increased up to 500 Hz. To compensate for this perturbation, speakers had to enhance lip protrusion and/or tongue retraction. These adaptive maneuvers should have been concurrent with auditory perception abilities. Preliminary results show that congenitally blind speakers gave greater weight to auditory perception than their sighted peers, while compensating differently for the perturbations. These findings support the hypothesis that vision plays a significant role in the implementation of phonological targets.

2aSC15. Vocal imitations of basic auditory features.

Guillaume Lemaitre, Ali Jabbari, Olivier Houix, Nicolas Misdariis, and Patrick Susini (IRCAM, IRCAM, 1 Pl. Stravinsky, Paris 75004, France, GuillaumeJLemaitre@gmail.com)

We recently showed that vocal imitations are effective descriptions of a variety of sounds (Lemaitre and Rocchesso, 2014). The current study investigated the mechanisms of effective vocal imitations by studying if speakers could accurately reproduce basic auditory features. It focused on four features: pitch, tempo (basic musical features), sharpness, and onset (basic dimensions of timbre). It used two sets of 16 referent sounds (modulated narrow-band noises and pure tones), each crossing two of the four features. Dissimilarity rating experiments and multidimensional scaling analyses confirmed that listeners could easily discriminate the 16 sounds based on the four features. Two expert and two lay participants recorded vocal imitations of the 32 sounds. Individual analyses highlighted that participants could reproduce accurately pitch and tempo of the referent sounds (experts being more accurate). There were larger differences of strategy for sharpness and onset. Participants matched the sharpness of the referent sounds either to the frequency of one particular formant or to the overall spectral balance of their voice. Onsets were ignored or imitated with crescendos. Overall, these results show that speakers may not imitate accurately absolute dimensions of timbre, hence suggesting that other features (such as dynamic patterns) may be more effective for sound recognition.

2aSC16. Real-time three-dimensional ultrasound imaging of pre- and post-vocalic liquid consonants in American English.

Brandon Rhodes, Kelly Berkson, Kenneth de Jong, and Steven Lulich (Dept. of Linguist., Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Speech sound articulation is typically characterized in the mid-sagittal plane. However, lateral segments, such as the /l/ category in most varieties of English, are usually identified in introductory phonetics courses as exceptions to this rule. On the other hand, many productions of post-vocalic /l/ in varieties spoken in the lower Mid-west U.S. are noted as being non-lateral, involving only a dorsal articulation very different from the canonical

coronal occlusion. Furthermore, a large body of literature indicates multi-constriction articulations, which vary by syllable position, for liquids like American English /l/ and /r/. This research presents results from a study of constriction location and laterality in pre- and post-vocalic /l/ and /r/ using real-time 3D ultrasound images of tongue motion, digitized impressions of the palate, and time-aligned acoustic signals.

2aSC17. Coupling relations underlying complex coordinative patterns in speech production. Leonardo Lancia, Sven Grawunder, and Benjamin Rosenbaum (Linguist, Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, Leipzig 04103, Germany, leonardo_lancia@eva.mpg.de)

In studying linguistic behavior, we are often faced to complex dynamical patterns arising from the highly coordinated activity of many partially autonomous processes. In this work, we apply a new approach aimed at studying abstract coupling relations coordinating the behavior of dynamical systems governed by goal oriented behavior. The approach, based on an original version of recurrence analysis, allows to deal with the principal difficulties of this task, which are mainly related to the heterogeneity, the lack of separability and the lack of stationarity of the processes under study. The method is validated through simulations of theoretical systems and it is adopted to capture (1) invariant abstract coupling structure underlying systematically varying trajectories of the speech articulators involved in the production of labial and coronal plosive and fricative consonants (produced at slow and fast speech rate by five German speakers and recorded via electromagnetic articulography); (2) systematic differences in the coordination between energy modulations observed at different frequency bands in the acoustic signal and showing the interplay between syllable and stress related processes and its relation to rhythmic differences between typologically different languages (pear story narrations were recorded from five speakers per language in German, Italian, French, English, and Polish).

2aSC18. Kinematic properties of concurrently recorded speech and body gestures and their relationship to prosodic structure. Jelena Krivokapic (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220, jelenak@umich.edu), Mark Tiede (Haskins Labs., New Haven, CT), and Martha E. Tyrone (Dept. of Commun. Sci. and Disord., Long Island Univ. Brooklyn, Brooklyn, NY)

Previous research has shown that body gesturing, such as hand and torso movements, is related to prosodic structure (e.g., Esteve-Gibert & Prieto 2013). However, the details of the relationship are not well understood and specifically the effects of prosodic structure on gesturing duration have received little attention. We present an experiment investigating the effects of stress and prosodic boundaries on the temporal properties of body gesturing and gestures of the vocal tract. Two speakers read 6 repetitions of 12 sentences examining the effect of boundary (no boundary, ip boundary, IP boundary), stress (no stress, stress) and their interaction, phrase finally and phrase initially (144 sentences total). They were observed through concurrent recordings of speech audio, vocal tract gestures using electromagnetic articulometry, and body gesturing using Vicon motion capture. We test the hypothesis that manual movements lengthen under prominence and at boundaries and that the lengthening at boundaries is cumulative, parallel to the behavior of gestures of the vocal tract. Analyzing speech prosody and body gesturing in this manner will allow us to investigate whether prosodic control extends to body gesturing and to identify coordinative patterns that are relevant to fields such as language pedagogy and speech pathology. [Work supported by NIH.]

2aSC19. Articulation of vowel height in Taiwanese Vowels: An EMA study. Yuehchin Chang, Yi-cheng Chen, and Feng-fan Hsieh (Inst. of Linguist, National Tsing Hua Univ., 101, sec. II, Kuang Fu Rd., Hsinchu, Taiwan 300, Taiwan, ycchang@mx.nthu.edu.tw)

Whalen *et al.*'s (2010) ultrasound study has suggested that constriction degree "may be the best descriptor for height" in American English front vowels {i, e, ε}. The present study took a further step, investigating the case of height and backness distinction in Taiwanese front and back vowels. Stimuli consisted in /CV/ sequences with V corresponding to one of the six

vowels {i, e, a, ə, ɤ, u} (Southern variety). Five adult speakers were asked to repeat each sequence six times in random order. The articulatory data were collected using the Carstens AG500 EMA system. Our results show that vowel height and backness in Taiwanese vowels can be reliably differentiated as follows: (1) for front and central vowels, height is primarily distinguished by Tongue Body Constriction Degree (TBCD) (i>e>a) (less robustly by Tongue Tip Constriction Degree), whereas height distinction of back vowels is made in Tongue Dorsum Constriction Degree (TDCD) (u>ɤ>ə), (2) backness can be distinguished by Tongue Tip and Tongue Body Constriction Location. Therefore, Whalen *et al.*'s (2010) claim can be extended to back vowels as well (at least in Taiwanese). Furthermore, our results are compatible with Ladefoged's (2001) "holistic" view of vowel height as long as his "highest point of the tongue" can be "locally" interpreted: TBCD for height of front and central vowels and TDCD for back vowels. In sum, the present study suggests that distinction in vowel height be relativized to different gestural constriction degree.

2aSC20. The effect of palate morphology on lingual shape during vowel production. Jennell Vick, Michelle L. Foye (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Nolan Schreiber, and Gregory S. Lee (School of Eng., Case Western Reserve Univ., Cleveland, OH)

A previous report described significant differences between males and females in the shape of the tongue during the production of vowels (Vick *et al.*, 2014). The purpose of the present study was to understand the influence of palate morphology on these measured differences in tongue shape. This study examined tongue movements in consonant-vowel-consonant sequences drawn from words in phrases as produced by 32 participants, including 12 adults (6m, 6f) and 20 children (2m and 2f at each age of 11, 12, 13, 14, and 15 years). Movements of four points on the tongue were tracked at 100 Hz using the Wave Electromagnetic Speech Research System (NDI, Waterloo, ON, CA). Palate curvature measures were derived from a hard palate trace made with a 6dof probe sensor. The phrases produced included the vowels /i/, /I/, /æ/, and /u/ in words (i.e., "see," sit," cat," and "zoo"). The horizontal curvature of the tongue was calculated at the trajectory speed minimum associated with the vowel production using a least-squares quadratic fit of the positional coordinates of the four points on the tongue. A regression model was used to determine the variance in tongue shape explained by palate morphology in the 32 talkers.

2aSC21. An articulatory model of the velum developed from cineradiographic data. Yves Laprie (LORIA/CNRS, 615 rue du jardin botanique, Villers-lès-Nancy 54600, France, yves.laprie@loria.fr)

The objective of this work is to develop an articulatory model which comprises the velum so as to be able to synthesize nasal vowels and consonants in French. Unlike one exception of a velum model developed from static 3D MRI images there is no model derived from articulatory data that can be used within a vocal tract articulatory model. Our model is derived from X-ray films of the DOCVACIM database. The velum contour has been tracked semi-automatically and then carefully checked and corrected from 1050 mid-sagittal images of the vocal tract. Principal component analysis has been applied onto these contours. The first component represents more than 50% of the variance and controls the velum opening. The next two components capture 20 and 12 % of the variance, respectively, and explain the deformation of the velum between open and close configurations. In parallel a series of MRI images of nasal vowels and consonant of French was analyzed. The velum opening roughly has a shape of a bean, whose length is almost constant for one speaker and width is given by the 2D articulatory model. Together with an evaluation of the nasal cavities this model can be used for articulatory synthesis.

2aSC22. A task dynamic approach to the coda-voicing effect on vowel duration. Robert Hagiwara (Dept. of Linguist., Univ. of MB, Winnipeg, MB R3T 5V5, Canada, robh@umanitoba.ca)

The coda-voicing effect on vowel duration (CVE), in which vowels are longer with voiced codas and shorter with voiceless, is well attested. However, the precise mechanism of this relationship is not well studied.

Task-dynamic models such as articulatory phonology (e.g., Browman & Goldstein, 1986) offer two ways to affect the duration of a vowel in a CVC syllable: the relative *phasing* of the onset, vowel, and coda gestures (when they start relative to one another), and the relative *stiffness* of an individual gesture (roughly how quickly it is executed). Onosson (2010) argued that for /ai/, the principal mechanism of Canadian Raising (CR) was vowel shortening via increased overlap of the onset gesture and the vowel, rather than vowel-

coda phasing or stiffening the vowel gesture. This study investigates whether this explanation holds generally for 15 vowels in Canadian English, using controlled wordlist-style data (originally from Hagiwara, 2006). Preliminary investigation suggests that onset-phasing may hold for the diphthongs, but does not adequately characterize CVE across the vowels. The observable mechanism(s) of CVE will be discussed, as well as implications for CVE and related effects (e.g., CR) across dialects/languages generally.

TUESDAY MORNING, 19 MAY 2015

BALLROOM 1, 8:00 A.M. TO 11:55 A.M.

Session 2aSP

Signal Processing in Acoustics, Speech Communication and Psychological and Physiological Acoustics: Characteristic Spectral and Temporal Patterns in Speech Signals

Xiao Perdereau, Cochair

Burgundy University, 9, Av. A. Savary, BP 47870, Dijon 21078, France

Daniel Fogerty, Cochair

Communication Sciences and Disorders, University of South Carolina, 1621 Greene St., Columbia, SC 29208

Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. The contribution of temporal modulations to speech intelligibility: Some thoughts about the speech envelope. Ken W. Grant (Audiol. and Speech Ctr., Walter Reed National Military Med. Ctr., 301 Hamilton Ave., Silver Spring, MD 20901, ken.w.grant@gmail.com)

Rosen [Phil. Trans. R. Soc. Lond. B **336**, 367–373 (1992)] laid out an extremely useful framework for discussing temporal information in speech. Three broad categories were defined: temporal envelope (2–50 Hz), periodicity (50–500 Hz), and temporal fine-structure (600–10,000 Hz). The slower modulations associated with the speech envelope convey segmental cues for voicing and manner-of-articulation, and prosodic cues for syllabification and stress. These cues are manifest in the attack and fall times as well as the overall duration of the envelope segments. The mid-rate fluctuations associated with periodicity convey manner cues (periodic vs aperiodic) as well as voice pitch (an essential cue for intonation and stress). The faster rate modulations associated with temporal fine-structure (modulations within each pitch period) convey place-of-articulation cues and spectral-shape cues for manner-of-articulation and voicing. Studies that try to separate the effects of each of these temporal categories on speech perception often use signal processing strategies that filter out the temporal modulations of interest and present this information without periodicity or temporal fine-structure to listeners. The processing details of how the temporal envelope is extracted are critical to whether cues more readily associated with periodicity or fine-structure are inadvertently presented to listeners. Examples will be provided.

8:25

2aSP2. Developmental time course of auditory perception of modulation speech cues. Christian Lorenzi (Institut d'Etude de la Cognition, Ecole normale supérieure, 29 rue d'Ulm, Paris, Ile de France 75005, France, christian.lorenzi@ens.fr)

Speech contains strong amplitude modulation (AM) and frequency modulation (FM) cues, which are commonly assumed to play an important role in speech identification for adults. We will review recent studies aiming to characterize the development of auditory perception of AM and FM speech cues for 6 and/or 10-month-old infants learning French or Mandarin [e.g., Cabrera, L., Tsao, F.-M., Gnanasia, D., Bertoncini, J., & Lorenzi C. (2014), *J. Acoust. Soc. Am.* **136**, 877–882; Cabrera, L., Bertoncini, J., & Lorenzi, C. (2013), *J. Speech, Lang., Hearing Res.* **56**, 1733–1744]. These studies were based on vocoders, which are analysis and synthesis systems designed to manipulate the modulation components of speech sounds in a given number of frequency bands. Overall, the results suggest that: (i) the auditory processing of AM and FM speech cues is “functional” by 6 months, (ii) the auditory processing of the AM and FM cues is fine-tuned by language exposure between 6 and 10 months. These findings may help improving current models of modulation processing that do not take into account the plasticity of the auditory and speech-processing system.

8:45

2aSP3. Contributions of the temporal envelope in adverse listening conditions. Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, COMD, Keenan Bldg., Ste. 300, Columbia, SC 29208, fogerty@sc.edu)

Amplitude modulations of the temporal envelope provide important cues for intelligibility, particularly in quiet listening environments. However, the speech understanding of listeners based on this temporal speech signal is often impaired when temporal interruptions or maskers interfere with the transmission of this important information. Performance in these adverse listening conditions is variable, based, in part, upon the properties of the interfering masker and the degree to which the modulation properties of the target speech signal are preserved. In some cases, listeners may be required to shift perceptual weighting to alternate, preserved modulation cues in order to maximize intelligibility of the speech signal. However, modulation properties of the speech signal are related to specific speech events. Therefore, different modulation properties are likely to contribute diverse information for speech understanding. The ability to access and process temporal modulation properties of the speech signal that unfold over different timescales is essential for maximizing speech intelligibility in everyday listening conditions. [Work supported by NIH/NIDCD and ASHA.]

9:05

2aSP4. Segmental contribution to the intelligibility of noise-suppressed speech. Fei Chen (Dept. of Elec. and Electron. Eng., South Univ. of Sci. and Technol. of China, Shenzhen 518055, China, fchen@sustc.edu.cn)

Many studies suggested that most speech enhancement algorithms do not improve the subjective intelligibility scores of noise-suppressed speech when presented to normal-hearing listeners. Nevertheless, the reason for lacking intelligibility improvement is still unclear. This study assessed the segmental contribution to the intelligibility of noise-suppressed speech. Mandarin sentences were corrupted by steady-state speech-spectrum shaped noise and multi-talker babble. Stimuli were synthesized by using noise-suppressed speech processed by three types of speech enhancement algorithms and a noise-replacement paradigm. The noise-replacement paradigm preserved selected speech segments (i.e., vowel-only and vowel-plus-consonant-onset), and replaced the rest with noise. Listening experiments were performed to collect the intelligibility scores from normal-hearing listeners. Experimental results showed that the selectively synthesized stimuli (i.e., vowel-only and vowel-plus-consonant-onset) from noise-suppressed speech may be more intelligible than those from noise-corrupted speech. However, this benefit was only observed when the speech signal was corrupted by speech-spectrum shaped noise but not by babble masker. This work suggested that the segmental distortion caused by speech enhancement algorithms may affect the information integration from noise-suppressed speech, and high-level top-down processing may account for the deficiency of the present speech enhancement algorithms for improving subjective intelligibility performance.

9:25

2aSP5. Using speech intelligibility metrics to predict the performance of hearing-impaired listeners. Karen Payton (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, kpayton@umassd.edu)

When fitting a hearing aid to a hearing-impaired listener, it would be very beneficial for an audiologist to be able to use an objective metric to determine the best signal processing algorithm to compensate for the user's loss and improve the user's understanding of speech in degraded acoustic environments. Several metrics have been considered to predict impaired listeners' speech recognition, including modified versions of the Speech Intelligibility Index (SII) [e.g., Ricketts, *Ear Hear.* **17**, 124–132 (1996)] and the Speech Transmission Index (STI) [e.g., Humes *et al.*, *J. Speech Hear. Res.* **29**, 447–462 (1986)]. For impaired listeners using linear amplification hearing aids, there are metrics that work fairly well. For listeners using hearing aids that include nonlinear processing, such as amplitude compression, results have not been as successful. This talk will review the current status of metric capabilities, approaches that have been proposed to mitigate metric limitations and which metric features appear most promising for future research.

9:45

2aSP6. A novel method to measure the resonances of the vocal tract. Bertrand Delvaux and David Howard (Dept. of Electronics, Univ. of York, Heslington, York, York YO10 5DD, United Kingdom, bertrand.delvaux@gmail.com)

In speech/singing, knowledge of the frequencies of the resonances of the vocal tract gives access to the vowel type (lower resonances) and the voice timbre (higher resonances). It is therefore crucial to be able to measure accurately these resonant frequencies directly. Several approaches have been developed such as glottal excitation, excitation at the lips, or medical imaging. More accurate measurements of the vocal tract resonances have emerged from physical modeling in recent years but a simple-to-use *in vivo* measurement tool would be much more valuable to give feedback to the singer/actor and their teacher(s), as well as voice therapists, language teachers, speech scientists, phoneticians, and linguists. In this article, we suggest a novel method using a sinesweep to measure the resonant frequencies of the vocal tract simultaneously with the voice spectrum.

10:05–10:20 Break

10:20

2aSP7. An acoustic phonetic study on the production accuracy of lexical stress contrasts in Arabic and Mandarin-accented English. Paul R. Keyworth (Appl. Linguist, Saint Cloud State Univ., 3710 W. Saint Germain St., Apt. #234, Saint Cloud, MN 56301-7319, kepa1104@stcloudstate.edu)

The researcher explored the socio-morphophonetic characteristics of English as a second language lexical prosodic competence. One hundred participants from Midwest USA (29), Saudi Arabia (38), and China (33) were recorded producing English lexical stress (ELS) in tokens containing seven different stress-moving suffixes—[-ic], [-ical], [-ity], [-ian], [-ify], [-ial], and [-ious]. Fundamental frequency, duration, and intensity productions were analyzed using *Praat*. In total, 2125 vowels in 800 spectrograms were analyzed (excluding stress placement and pronunciation errors). Statistical sampling techniques were used to evaluate acquisition of accurate ELS production versus native language and language proficiency. Speech samples of native-speakers were analyzed to provide norm values for cross-

reference and to provide insights into the proposed *Salience Hierarchy of the Acoustic Correlates of Stress* (SHACS). The results support the notion that a SHACS does exist in the L1 sound system and that native-like command is attainable for English language learners through increased study/L2 input. Saudi English speakers, who often do not fully reduce vowels, produced durational contrasts more accurately when they had studied English for longer. Similarly, proficient Chinese learners of English seemed to be able to overcome negative transfer from their tonal system as they produced pitch in a more native-like manner.

10:40

2aSP8. Investigating the effectiveness of Globalvoice CALL software in native Japanese English speech. Hiroyuki Obari (Economics, Aoyama Gakuin Univ., 2-12-21, Nakatakatsu, Tsuchiura, Ibaraki 300-0815, Japan, obari119@gmail.com), Hiroaki Kojima (National Inst. of Adv. Industrial Sci. and Technol. (AIST), Tsukuba, Ibaraki, Japan), and Shuichi Itahashi (Univ. of Tsukuba, Aobaku, Sendai, Japan)

Many variables exist in the evaluation of the speech of non-native speakers of English. In previous research, rhythmic accent and pauses are more salient than segmental features in English utterances to make one's speech more intelligible. In this regard, prosodic features such as intonation and rhythm are crucial in comprehensible speech. One of the main goals of English education in Japan is to help Japanese students speak English more intelligibly in order to be more clearly understood while taking part in international communication. In this talk, several key parameters such as speech duration, speech power, and F0 (pitch), are all introduced to help determine to what extent Japanese students are able to improve their English pronunciation through the use of Globalvoice CALL software. Globalvoice CALL enables students to input English words or sentences for the purpose of practicing their pronunciation and prosody in order to reduce their Japanese-accented speech. The purpose of this paper is to investigate the effectiveness of Globalvoice CALL software to improve the English pronunciation of Japanese EFL learners. Student productions of English sentences were analyzed with regard to speech duration, speech power, and F0 (pitch) to determine how much progress was attained in their English pronunciation and overall proficiency. The results showed improvement in acoustic properties between the pre- and post-recorded readings, indicating the software helped the students to improve their English pronunciation.

11:00

2aSP9. Contribution of the acoustic cues to the non-native accent. Yves Laprie (LORIA/CNRS, 615 rue du jardin botanique, Villers-lès-Nancy 54600, France, yves.laprie@loria.fr)

This communication concerns the contribution of acoustic cues to the perception of non-native accents and presents some solutions to help learners to master the acoustic cues of the foreign language. Elementary acoustic cues (F0 frequency, formants...) will be presented by using copy synthesis used to pilot the Klatt formant synthesizer. This enables their perceptual contribution and that of more complex acoustic cues, which play a major role in the identification of speech sounds, to be evaluated easily. The non-native accent is mainly due to the:—realization of incorrect tones, stress patterns, and more generally prosody,—use of the mother tongue phonemes instead of those of the second language,—use of inappropriate acoustic cues to realize a phonetic feature. This is often the case with the voiced/unvoiced feature which covers a range of acoustic cues. These deviations depend on the phonetic specificities of the mother tongue with respect to those of the second language. The first aspect of corrections is the automatic diagnosis of non-native deviations. Learners can be made aware of them, trained to learn new phonetic contrasts, or guided by providing them with appropriate articulatory information, or acoustic feedback which is particularly efficient for prosody.

11:20

2aSP10. Speech rhythm processing. Xiao Perdereau (Laboratoire Interdisciplinaire Carnot de Bourgogne UMR 6303 CNRS, 9, Av. A. Savary, BP 47870, Dijon 21078, France, amiscec@free.fr)

Speech rhythm is one of the main prosodic features of spoken languages. The quantitative research data on this topic in the literature are not numerous may be due to the confusing divergence in this concept definition. In order to render clear this concept, we investigated a speech segment in Mandarin. Due to its delicate temporal structure, it leads easily to the semantic ambiguities, well known in speech recognition. In order to understand how human manage the speech meaning in a natural language processing, we designed a series of lexical duration and pause interval modulations based on the speech segment for production. By analyzing the resultant acoustic patterns, we observed two types of temporal grouping. The type 1 is context dependent, the time range has no regularity, the modulations suffered from semantic ambiguities. The type 2 includes the stable patterns presenting temporal regularities. We consider the temporal structures in this type as speech rhythm. This definition corresponds to the articulatory periodicity that produces the consonant and vowel alternations. They are context independent and reproducible.

Contributed Paper

11:40

2aSP11. Using automatic speech recognition to identify pediatric speech errors. Roozbeh Sadeghian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, 4400 Vestal Parkway East, Binghamton, NY 13902, rsadegh1@binghamton.edu), Madhavi Ratnagiri (Speech Res. Lab., Nemours Biomedical Res., Wilmington, DE), Stephen A. Zahorian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, Binghamton, NY), and H. Timothy Bunnell (Speech Res. Lab., Nemours Biomedical Res., Wilmington, DE)

Speech delay is a childhood language problem that might resolve without intervention, but might alternatively presage continued speech and language deficits. Thus, early detection through screening might help to identify children for whom intervention is warranted. The goal of this work is to develop Automatic

Speech Recognition (ASR) methods to partially automate screening for speech delay in young children. Speech data were recorded from typically developing and speech delayed children (N=63) aged 6 to 9 years old during administration of the Goldman Fristoe Test of Articulation (GFTA). Monophone Hidden Markov Model (HMM) acoustic models were trained on speech data obtained from 207 typically developing children in the same age range. These training data consisted of a total of about 18,000 single-word utterances. The HMMs were then used to develop an utterance verification system to distinguish correct versus error productions. Several variations of the recognition strategy, feature extraction, and scoring methods were investigated. The best overall ASR result for distinguishing normal versus abnormal speech is approximately 86%. It is hypothesized that the ASR methods could approach the level of accuracy of speech therapists for this task (agreement among multiple therapists is over 95%), but a much larger database may be needed.

Session 2aUW**Underwater Acoustics: Historical Perspectives on the Origins of Underwater Acoustics I**

David Bradley, Cochair

Penn State University, PO Box 30, State College, PA 16870

Thomas G. Muir, Cochair

*Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713***Chair's Introduction—8:00*****Invited Papers*****8:10****2aUW1. Paul Langevin's contributions to the development of underwater acoustics.** Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Paul Langevin made significant contributions to the understanding of piezoelectricity and the development of piezoceramics materials. For instance, Professor Langevin's invented the quartz sandwich transducer in 1917 for underwater sound transmission. He subsequently used this to develop the first underwater Sonar for submarine detection during World War I, and his work was extensively used the French and American Navy. After world war I, SONAR devices he helped developed were used in several French ocean-liners. We will review Paul Langevin's most noteworthy contributions to the development of underwater acoustics.

8:30**2aUW2. Harvey C. Hayes: First superintendent of the Sound Division at the Naval Research Laboratory, Washington, D.C.** Fred T. Erskine (none, 135 Kensington Dr., Littlestown, PA 17340-9767, kx3z@aol.com)

Harvey Cornelius Hayes, physicist, had a long tenure (1923–1947) as the first Superintendent of the Sound Division (later renamed Acoustics Division) at the Naval Research Laboratory (NRL), Washington, D.C. In the period prior to World War II, Hayes led a small group of only five to eight researchers that was devoted to developing active (echo-ranging) sonar and improved passive (listening) sonar for the navy's surface ships and submarines. They developed a tunable type of sonar that found widespread use in World War II. They conducted field experiments to take detailed measurements on the propagation of sound in oceanic acoustic ducts. They developed techniques for silencing "singing" propellers in navy ships and aircraft carriers by sharpening propeller edges. During World War II the Sound Division expanded in size about twentyfold and NRL researchers conducted numerous experiments to address navy concerns regarding sonar performance. Sound Division researchers made significant advances including the development of communications equipment for divers, torpedo countermeasures, the development of streamlined sonar domes for ships, control and stabilization of sonar transducers, methods for localization of a submerged target beyond the resolution of the sonar beam, and coordination of sonar systems with fire-control systems. In the period following World War II, Hayes initiated new efforts on shock and vibration, crystal development, and physiological acoustics.

8:50**2aUW3. The Submarine Signal Company.** Thomas R. Howarth (NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

The Submarine Signal Company (SSC) was established in 1901 in Boston, MA, and was the first commercial enterprise organized to conduct underwater sound research and to develop equipment to be used for increasing safety of navigation. The initial product line (prior to 1914) included underwater bells for shore based stations, buoys, and lightships, as well as encased microphones for sound detection on the ships. The bells were used for navigational and hazard warning purposes. In April 1912, future SSC President H. J. W. Fay had a chance meeting with Professor Reginald Fessenden at a Boston railroad station. This meeting led to an invitation for a Fessenden visit to SSC where upon Reginald so impressed company managers and engineers that SSC hired Fessenden consulting services to redesign their hydrophones. Based on this and later contributions to underwater sound products, H. J. W. Fay and R. A. Fessenden are two of the five named pioneers of the ASA award known as "The Pioneers of Underwater Acoustics Medal". This presentation will discuss the formation and product lines of SSC through their formation till 1946 when they were merged with Raytheon Company.

9:10

2aUW4. The naval science of Albert Beaumont Wood, O.B.E., D.Sc. Michael J. Buckingham (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

A. B. Wood's career in Naval scientific research spanned a period of nearly 50 years, from the early days of the first World War to the time of his death in 1964. After graduating from Manchester University with a first class honors degree in Physics he seemed destined for a career of distinction in academia. But, like many young men at the time, when the U.K. was becoming deeply embroiled in war, he felt dissatisfied with the cloistered academic life and instead became one of the first two physicists to receive an official appointment with the Admiralty under the newly formed "Board of Invention and Research". Thus, the Royal Navy Scientific Service was born and Dr. Wood began his celebrated research into the propagation of sound underwater, about which little was known at the time. Many of his technical achievements were made at the Admiralty Research Laboratory, Teddington and, shortly before his death, he spent a year at the U.S. Naval Electronics Laboratory, as it was then called, in San Diego. He was awarded the Pioneer of Underwater Acoustics medal by the Acoustical Society of America and his Text Book of Sound is still a standard work on the subject.

9:30

2aUW5. Leonid Brekhovskikh and his lasting impact on underwater acoustics. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

This paper will discuss Brekhovskikh's groundbreaking contributions into physical oceanography and the theory of wave propagation, scattering, and diffraction from the perspective offered by current developments in underwater acoustics. In hindsight, three major contributions stand out in Brekhovskikh's legacy, which highlight different facets of his personality and his talents. For many decades, ocean acoustics was Leonid's passion. He developed and promoted an approach to underwater acoustics, which embraced the ocean's complexity and centered on devising techniques to reliably characterize the uncertain environment by acoustic means. The approach led to creation of an informal but rather productive and enduring "Brekhovskikh's school." Leonid worked only briefly on wave scattering by rough surfaces. However, his findings, which became known as the tangent plane approximation, revolutionized the field. Universally recognized are Brekhovskikh's systematic theoretical studies of underwater sound propagation, which are summarized in his celebrated book *Waves in Layered Media*. The theory includes spectral representations of wave fields, normal mode theory for open waveguides, and a clear treatment of diffraction phenomena attendant to caustics, lateral waves, and reflection of wave beams and pulses. The book charted the ways forward which have been and are followed by numerous researchers around the globe.

9:50

2aUW6. R. J. Urick—Introduction of the sonar equations. David Bradley (Penn State Univ., PO Box 30, State College, PA 16870, dlb25@psu.edu)

The Sonar Equations were developed during the Second World War to calculate the maximum ranges of sonar equipment. They were formally discussed in a series of reports published by the Office of Naval Research in the mid 1950's entitled "A Summary of Acoustic Data" authored by R. J. Urick and A. W. Pryce. Later codified in Urick's Book, *Principles of Underwater Sound*, they are the primary tool for design, development, and testing of undersea acoustic hardware. They also provide a useful means for estimating expected values of marine physical properties (e.g., ambient noise, transmission loss) or, conversely, a sanity check on data collected at sea. The presentation is focused on the history of the sonar equations, their early use and transition to modern form and employment.

10:10–10:30 Break

10:30

2aUW7. The University of California Division of War Research and the Marine Physical Laboratory. W A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

In 1941, motivated by the large-scale destruction of ships by submarines in World War II, three university-operated laboratories were established: Columbia University Division of War Research, Harvard Underwater Sound Laboratory, and The University of California Division of War Research (UCDWR). UCDWR was led initially by Vern Knudsen of the University of California, Los Angeles, with senior staff recruited from academic institutions across the country. After the war, UCDWR was dissolved with the The Navy Electronics Laboratory absorbing most of the UCDWR employees. Further, by agreement between the Navy and the Regents of the University of California, the Marine Physical Laboratory (MPL) was formed with an initial scientific staff of five people from UCDWR with Carl Eckart as its Director. We review this transition period and MPL's subsequent trajectory within the University of California system to its ultimate venue in the Scripps Institution of Oceanography.

10:50

2aUW8. Contributions to the development of underwater acoustics at the Harvard Underwater Sound Laboratory. Frederick M. Pestorius (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78759, pestoriu@arlut.utexas.edu) and David T. Blackstock (Appl. Res. Labs. and Mech. Eng. Dept., Univ. of Texas at Austin, Austin, TX)

The Harvard Underwater Sound Laboratory (HUSL) began in June 1941, six months before Pearl Harbor, and closed on January 31, 1946. HUSL was directed throughout its existence by (then) Assoc. Prof. Frederick V. Hunt, who incidentally coined the acronym sonar. This paper traces several contributions HUSL made to undersea warfare through exploitation of underwater acoustics. Especially significant were developments by HUSL in sonar and underwater ordnance (torpedoes). Both developments were supported by numerous sub-projects on a wide range of applications such as reverberation suppression, Doppler enhancement, and torpedo transducer design. Particular attention is paid to work on air-dropped and submarine-launched acoustic homing torpedoes. HUSL's far-reaching influence continues to be evident in today's sonar and torpedo systems. Closure of HUSL spawned Penn State's Ordnance Research Laboratory (now Applied Research Laboratory) and enriched sonar development at the US Navy Underwater Sound Laboratory, New London.

Moreover, HUSL personnel returning to the University of Texas soon founded the Defense Research Laboratory (now Applied Research Laboratories) at Texas. We pay particular homage to living HUSL alumnus A. Wilson Nolle, who, along with our recently departed colleague, Reuben Wallace, and 12 other deceased alumni, had come to HUSL from the University of Texas.

11:10

2aUW9. Underwater acoustics research at the Woods Hole Oceanographic Institution, 1930–1960. James Lynch, Arthur Newhall, and Robert Frosch (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu)

The Woods Hole Oceanographic Institution (WHOI) was founded in 1930, and throughout its history has had a strong involvement in research into the science and applications of sound in the ocean. In terms of a brief history, three eras stand out: (1) pre-WWII, (2) WWII, and (3) the postwar years. Though the most colorful pictures and stories come from the war years, the other two eras also hold much interest. Many of the names associated with research at WHOI also commonly appear associated with other laboratories, and the interplay between various institutions is a fascinating part of the history. Personal reminiscences, technical report details, photos, audio, and even some video records will be presented, courtesy of the WHOI Archives.

11:30

2aUW10. Columbia Division of War Research and the work of Ewing, Worzel, and Pekeris. D. K. Knobles, Evan K. Westwood, and Thomas G. Muir (Appl. Res. Labs., Univ. of Texas at Austin, P.O Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu)

Three University laboratories were established in WWII to conduct underwater acoustics research, in response to widespread attacks on allied shipping by German submarines; the Columbia University Division of War Research, Harvard Underwater Sound Laboratory, and the University of California Division of War Research, and the last two of these are being discussed by other speakers in this session. During the war, outstanding work was done at Columbia by Maurice Ewing, J. Lamar Worzel, and C. L. Pekeris, which has been summarized in the book, *Propagation of Sound in the Ocean*. Experiments utilizing explosive sources were done at sea in 1943 and 1944, in both shallow and deep water, off the east coast of the United States, with the USS Saluda (IX87) and other vessels, in cooperation with the Naval Ordnance Laboratory, Woods Hole Oceanographic Institution, and the U.S. Navy Underwater Sound Laboratory. Pekeris is noted for his wave theoretic analysis of the shallow water propagation results and is credited with being the first to apply normal mode theory to shallow water waveguides. Ewing and Worzel are noted for their development of some of the first long range ocean acoustics experiments, which continue to this day in similar context and importance. Some highlights of this work are presented to illustrate the beginnings of long range underwater acoustic propagation research in the United States.

2a TUE. AM

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

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

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

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

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

M. L'vov, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 2 Measurement and
evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures
Siemens Energy, Inc., 5101 Westinghouse Blvd., Charlotte, NC 28273

D.J. Evans, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 3 Use and calibration of
vibration and shock measuring devices
13 Watch Hill Place, Gaithersburg, MD 20878

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

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

D. A. Preves and C. Walber, U.S. Technical Co-advisors for IEC/TC 29, Electroacoustics
(D. Preves) Starkey Hearing Technologies, 6600 Washington Ave., S., Eden Prairie, MN 55344
(C. Walber) PCB Piezotronics, Inc., 3425 Walden Ave., Depew, NY 14043 2495

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

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S2, which will be held on Monday, 18 May 2015 from 5:00 p.m. - 6:00 p.m.

The Standards Committee Plenary Group meeting will precede the meetings of the Accredited Standards Committees S1, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 19 May 2015	11:00 a.m. - 12:00 p.m.	S1, Acoustics
Tuesday, 19 May 2015	1:45 p.m. - 2:45 p.m.	ASC S3/SC 1, Animal Bioacoustics
Tuesday, 19 May 2015	3:00 p.m. - 4:15 p.m.	ASC S3, Bioacoustics
Tuesday, 19 May 2015	4:30 p.m. - 5:45 p.m.	ASC S12, Noise

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

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

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

2a TUE. AM

TUESDAY MORNING, 19 MAY 2015

DUQUESNE, 11:00 A.M. TO 12:00 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S1
5012 Macon Road, Rockville, MD 20852

A. A. Scharine, Vice Chair ASC S1
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459 Mulberry Point Road Aberdeen Proving Ground, MD 21005 5425

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note-those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance, and comfort.

Session 2pAA**Architectural Acoustics, Noise, Speech Communication, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Sixty-Fifth Anniversary of Noise and Health: Session in Honor of Karl Kryter II**

Larry E. Humes, Cochair

Dept. of Speech & Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002

Kenric D. Van Wyk, Cochair

*Acoustics By Design, Inc., 124 Fulton Street East, Second Floor, Grand Rapids, MI 49503***Chair's Introduction—1:00*****Invited Papers*****1:05****2pAA1. The cardiovascular effects of noise on man.** Wolfgang Babisch (Himbeersteig 37, Berlin 14129, Germany, wolfgang.babisch@t-online.de)

Noise is pervasive in everyday life and induces both auditory and non-auditory health effects. Systematic research of the effects of noise on the cardiovascular system has been carried out for more than 50 decades. Noise is a stressor that affects the autonomic nervous system and the endocrine system. Animal experiments, laboratory and field studies carried out on humans provide evidence that persistent exposure to environmental noise affects physiological endpoints, which in turn are adversely associated with cardiovascular diseases. These include hypertension, coronary heart disease, and stroke. New endpoints have been studied, including clinical states of metabolic syndrome such as diabetes mellitus. Chronic sleep disturbance is considered as an important mediator of the effects. Public health policies rely on quantitative risk assessment to set environmental quality standards and to regulate the noise exposure that is generated by environmental noise sources in the communities. Meta-analyses were carried out to derive exposure-response relationships between noise and cardiovascular health. Most of the epidemiological studies refer to road traffic and aircraft noise. No biologically determined threshold values can be determined. Cardiovascular effects due to noise and traffic-related air pollutants are largely independent of one another due to different biological mechanisms.

1:35**2pAA2. The behavioral impacts and spatial dynamics of alarm fatigue in reverberant healthcare treatment spaces.** Paul Barach (Univ. of Oslo, 31 Baker Farm Rd., Lincoln, Massachusetts 01773, pbarach@gmail.com)

Medical device alarms are deliberately designed to alert attention. Conditions found in hospitals produce the unintended consequences triggered by alarms called "alarm fatigue" that is not attributable to individually alarmed devices but rather to the aggregate conditions in which the alarms occur. "Alarm fatigue" is a condition of recurrent noises from myriad uncorrelated medical devices, set at maximum loudness, occurring in hard-walled, reverberant spaces and produces elevated stress, sleep impairment, disorientation and dangerously irrational behavior. Four conditions cause "alarm fatigue": (1) burgeoning use of alarmed devices none of which are prioritized or correlated with others and all of which are set to maximum sound pressure levels; (2) advanced technologies that have caused a steady increase in hospital noise levels; (3) small enclosed spaces that surround dense clusters of healthcare workers and patients with hard, sound-reflecting surfaces and high levels of noise; and, (4) codes and standards that require alarms to be significantly louder than background, ambient sound. These conditions produce an "acoustic feedback loop" in which noise inevitably and rapidly escalates to intolerable levels and interferes with behavior. Understanding these conditions and developing practical and effective ways to interrupt "acoustic feedback loops" requires trans-disciplinary approaches and mixed implementation methods that include health service researchers, signal-processing, bioacoustics, and room acoustics.

1:55**2pAA3. Transdisciplinary, clinical analysis of the effects of noisy environment on the cognitive performance of healthcare providers.** Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu), David Sykes (Acoust. Res. Council, Boston, MA), Wayne R. Triner (Emergency Medicine, Albany Medical College, Albany, NY), and Peter Dodds (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Karl Kryter's pioneering and encyclopedic effort begun over six decades ago to document "The Effects of Noise on Man" opened a door to trans-disciplinary, translational research. His approach continues to challenge many of us in acoustical science, medicine, and the development of codes and standards. The authors have emulated Kryter's trans-disciplinary approach by forming a unique and stable

partnership between doctors at a major medical center, scientist-engineers at a polytechnic institute, and a division of the American Hospital Association that promulgates evidence-based building criteria. This partnership enables detailed and insightful, quantitative research to be conducted in a variety of previously inaccessible clinical environments (e.g., hospitals) on the impacts of ambient noise on the cognitive performance of healthcare professionals. The authors couple psychoacoustic methods and soundscape analysis with cognitive performance tests to examine questions of major concern in the healthcare professions such as the extent to which medical errors are caused by environmental stresses such as noise. Several studies are currently being designed that highlight the challenges and opportunities of emulating Kryter's approach using 21st century methods.

2:15

2pAA4. Assessing hazardous noise pollution in the United States. Richard L. Neitzel (Dept. of Environ. Health Sci. and Risk Sci. Ctr., Univ. of Michigan, Ann Arbor, MI), Monica S. Hammer, and Tracy K. Swinburn (Risk Sci. Ctr., Univ. of Michigan, 1415 Washington Heights, Ann Arbor, MI 48109, monicashammer@gmail.com)

The risk of noise-induced hearing loss (NIHL) from noise exposure has been known for hundreds of years, but Karl Kryter's seminal work on NIHL represented a major step forward in addressing this disease. In recent years, a growing body of evidence has linked noise to additional non-auditory health effects, including cardiovascular disease, sleep disturbance, and stress. The United States has no national plan to address noise pollution, although standards designed to protect human health from noise do exist. National estimates of U.S. exposures were last created in the early 1980s. We have updated these estimates using current census and research data, and estimate that 104 million individuals in the United States had annual equivalent continuous average levels >70 dBA in 2013 and were at risk of NIHL and other effects. We describe approaches to the assessment of noise exposures at the level of the individual and the community, and outline ways in which noise can be more comprehensively assessed. Based on the number of exposed individuals in the United States and the impact of noise-related health effects, greater emphasis on noise reduction will likely improve public health and quality of life. This increased focus will allow us to continue and expand upon Kryter's outstanding work.

2:35

2pAA5. Karl Kryter—Soundscape pioneer. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooksaoustics.com)

Karl Kryter was a pioneer in the soundscape analysis field, even before the field had a name. Defined as the combination of the study of physical sound parameters with the perception of sound by listeners in an environmental context, soundscape analyses were conducted and reported by Kryter well before the term "Soundscape" was coined by R. Murray Schafer in 1977. These included several studies of community reactions to noise from subsonic aircraft while Kryter was with Bolt, Beranek and Newman in the 1950's. Those studies were followed by laboratory and field tests of listener reactions to aircraft noise in the 1960's. Kryter compiled and compared similar attitudinal and perceptual studies of transportation noise by other researchers in his 1970 book "The Effects of Noise on Man". This compilation has proven to be comprehensive and enduring in its presentation of noise perception data. Kryter's insights from those early studies provide valuable guidance to current and future soundscape researchers and soundscape design practitioners.

Contributed Papers

2:55

2pAA6. Contribution of floor treatment characteristics to background noise levels in health care facilities, Part 2. Adam C. Ackerman, Jonathan Zonshine, Eoin A. King, Robert D. Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu), and John J. LoVerde (Paul S. Veneklasen Res. Foundation, Santa Monica, CA)

Additional acoustical tests were conducted as Part 2 of a previous study [JASA 136(4), 2219–(2014)] on ten types of commercial grade flooring to assess their potential contribution to surface-generated noise within health-care facilities. These floor samples utilized an ECORE *ForestFX* sheet vinyl wear layer and tested with rubber-backing thicknesses ranging from 2 mm–10 mm, plus a composite backing composed of two 2 mm layers. Two types of ECORE International adhesives, *EGripIII* and *Evolve*, were tested as part of the adhesive used during manufacturing as well as with local mounting of all floors on concrete substrates. Sound power tests were conducted in compliance with ISO-3741-2010 for two source types, an impact tapping machine and a rolling hospital cart. The sample that achieved the lowest radiated sound power levels for both the rolling cart and tap tests was *ForestFX* sheet vinyl with 10 mm rubber backing and *Evolve* adhesive. Two trends were observed. First, measured sound power levels reduced as backing thickness rose. Second, the *Evolve* adhesive demonstrated lower sound

power levels for all tap/rolling tests when compared with those utilizing the *EGripIII* adhesive. The suitability of using thicker floor backings in health-care facilities are discussed. [Work supported by Paul S. Veneklasen Research Foundation.]

3:10

2pAA7. Effect of adhesives in structure-borne and impact sound measurements. Sharon Paley (ECORE Int., 715 Fountain Ave., Lancaster, PA 17601, sharon.paley@ecoreintl.com) and Bradley Hunt (ATI/Intertek, York, PA)

Based on a preliminary finding in "Contribution of Floor Treatment Characteristics to Noise Levels in Health Care Facilities Part 2," we investigate the effect that various adhesives may have in surface generated sound and impact sound transmission levels. The aforementioned study demonstrated that lower source room sound levels were generated with an acrylic adhesive assembly when compared to sound levels generated with a urethane adhesive assembly. These findings led to two follow-up questions: (1) What effect does adhesive have in IIC ratings? and (2) Is there a clear inverse relationship between sound levels within a source and receiving room, and how much does the adhesive affect this relationship? Results from tests conducted in a lab environment with multiple floor finishes and multiple assemblies will be discussed.

3:25–3:35 Break

Invited Papers

3:35

2pAA8. Karl Kryter: The evolution of the Articulation Index. Jont B. Allen (ECE, Univ. of IL, 1404 Sunny Acres Rd., Mahomet, IL 61853, jontallen@iee.org)

When one trained in speech perception hears the name Karl Kryter, they immediately think of the AI standard, which Kryter ferried through the standards process. Today, the AI has been studied to the point that we are beginning to understand both its beauty, and its fundamental limitations. The AI was created by Fletcher and Galt, as reported first in JASA in 1950, and in Fletcher's two books. It is based on two important intertwined observations: (1) that the average consonant error P_e is exponential in an average of critical band signal to noise ratios (i.e., $SP_e(\text{SNR}) = 0.015\{\text{AI}\}^S$) and (2) the total error is the product of independent critical band errors. While AI theory has been verified many times over, the real question is "Why does it work?" In this talk, I will answer this question, and in the process expose the AI's key weaknesses. AI theory has guided us to our modern understanding of speech perception in normal and hearing impaired ears. The publications supporting this work date from 1994–2014 and may be found at <http://hear.ai.uiuc.edu/Publications>.

3:55

2pAA9. The use of articulation theory and the speech intelligibility index in the design of clinical speech perception tests. Douglas Brungart and Kenneth W. Grant (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil)

For more than 50 years, the Articulation Index (AI) and its successor the Speech Intelligibility Index (SII) have been recognized as essential tools for predicting the impact that reduced bandwidth, background noise, and differences in speech materials will have on the speech perception performance of normal hearing listeners. However, one caveat about both AI and SII is that neither is considered to be valid for predicting the performance for hearing impaired listeners. Nevertheless, the insights provided by the AI and SII can be invaluable in the design and analysis of clinical speech tests designed to evaluate the relative performance of hearing impaired listeners. In this talk, we discuss role that the SII and AI played in the development of a speech-in-noise test based on the Modified Rhyme Test that is intended for use in the evaluation of Auditory Fitness-for-Duty in military personnel. In this case, the information drawn from the SII was essential both in the selection of appropriate speech materials and SNR values and in finding ways to analyze the results of normal and hearing impaired listeners. The results of the study provide evidence for a "pure" SNR loss in hearing impaired listeners that occurs even when all components of the speech signal are audible. [The views expressed in this article are those of the authors and do not necessarily reflect the official policy or position of the Department of Defense or U.S. Government.]

4:15

2pAA10. A challenge to Articulation Theory: Narrow-band acoustic signals and visual speech cues. Ken W. Grant (Audiol. and Speech Ctr., Walter Reed National Military Med Ctr., 301 Hamilton Ave., Silver Spring, MD 20901, ken.w.grant@gmail.com)

Given Kryter's extensive work on the physiological and subjective effects of noise on the human auditory system it is not surprising that at some point he would turn his attention to the impact of noise on speech intelligibility. In 1962, Kryter published "Methods for the Calculation and Use of the Articulation Index" greatly simplifying the methods originally proposed by Fletcher in the 20s and 30s and French and Steinberg in 1947, ultimately leading to the ANSI 1969 standard. Over the years, the AI and the revised Speech Intelligibility Index (SII) have become the leading metric for predicting speech recognition in noise and in quantifying the audibility provided by amplification. However, the scope of the AI/SII is still limited to communication conditions, which do not include multiple, narrow bands of speech or noise. Similarly, predictions for conditions of auditory-visual speech are only roughly approximated. Data from a variety of studies suggest that rather than ignore these communication conditions they should be studied closely for what they reveal. In this talk, I will show data that expose chinks in the armor of our understanding of speech perception in noise and under conditions where visual speech cues are present.

4:35

2pAA11. Predicting speech intelligibility for hearing-impaired listeners. James M. Kates (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

Karl Kryter was instrumental in developing the Articulation Index (AI) standard and demonstrating its effectiveness. The AI, and its successor, the Speech Intelligibility Index (SII), predict intelligibility based on the audibility of speech sounds. The concept of audibility is also relevant to hearing loss and the effectiveness of hearing aids, and extensions of the AI have been developed to predict hearing-aid benefit. However, audibility alone cannot explain the impact of nonlinear distortion and nonlinear hearing-aid processing on intelligibility. This talk will present a review of the how the AI and SII have been used for hearing aids, extensions to the SII targeted at predicting hearing-aid effectiveness, and more recent work that focuses on envelope fidelity rather than audibility in predicting intelligibility for hearing-aid users.

4:55

2pAA12. The effects of noise on learners of English as a second language. Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

In the 65 years since the publication of *The Effects of Noise on Man*, as the world has grown ever noisier, it has also grown more diverse and more connected than ever before. Today, nearly a fifth of U.S. residents speak a language other than English at home. Consequently, exploring the effects of noise on people communicating in a second language is a crucial component of understanding the effects of noise on mankind in our increasingly pluralistic society. This presentation will overview some of the challenges faced by second-language learners as speakers and listeners, particularly focusing on the extraction of phonetic information from a noisy or

impoverished signal. Results of ongoing research investigating vowel identification by native and non-native English-speaking listeners will be presented. Overall accuracy, the slope of the function relating signal-to-noise ratio and intelligibility, and effects of noise on listeners' ability to benefit from phonetic enhancement strategies will be compared across native English speakers and second-language learners with differing ages of immersion in an English-speaking environment.

5:15–5:30 Panel Discussion and Open Microphone

TUESDAY AFTERNOON, 19 MAY 2015

RIVERS, 1:00 P.M. TO 5:40 P.M.

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Acoustics of Fine Grained Sediments: Theory and Measurements

Mohsen Badiey, Cochair

College of Earth, Ocean, and Environment, University of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716

David P. Knobles, Cochair

Univ. of Texas, Austin, PO Box 8029, Austin, TX 78713

Charles W. Holland, Cochair

Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804

Chair's Introduction—1:00

Invited Papers

1:05

2pAO1. Modeling marine mud: A four phase system. Richard H. Bennett (SEAPROBE, Inc., 501 Pine St., Picayune, MS 39466, rhbenn_seaprobe1@bellsouth.net) and Matthew H. Hulbert (Res. Dynam., West Chester, PA)

Fine-grained marine sediment consists of up to four phases: (1) clay Minerals often with some silt and sand, (2) saline water, (3) free gas, and (4) organic matter (OM). Shallow-water, shelf, and slope muds often have all four phases present. The clay minerals are the "building blocks" of mud deposits; these consist of multi-plate face-to-face domains. There are only three possible modes of domain contact (termed signatures): (1) edge-to-face (EF), (2) edge-to-edge (EE), and (3) face-to-face (FF), usually offset. The domain faces are negatively charged and have a large surface area and the edges are both negative and positive charged. OM is usually expressed as total organic carbon (TOC) in percent total dry mass and can range from <0.5% to several percent and the OM can occupy a significant portion of pore volume. OM is usually attached to clay domain signatures and faces with the location partially dictated by electrostatic potential energy fields. When several percent TOC is present, the OM drives up water contents and porosity considerably. When free gas is present, it replaces part of the seawater in the pore space. Several published organo-clay fabric models for mud deposits may improve prediction of acoustic properties.

1:25

2pAO2. Frequency power-law attenuation and dispersion in marine sediments. Michael J. Buckingham (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Acoustic waves in marine sediments, including the fine-grained muds and clays, often exhibit an attenuation in the form of a frequency power law in which the exponent is close to unity. The frequency dependence of the wave speed, or dispersion, associated with such a power law has long been a subject of debate. Causality arguments, often in the form of the Kramers-Kronig dispersion relations, are usually applied to the problem, but these are not sufficient to characterize the dispersion fully. Following an alternative approach, a wave equation will be introduced which predicts the dispersion when the attenuation is of the frequency power law type. Based on this wave equation, it will be shown that only certain values of the frequency exponent are allowed, otherwise causality is violated, indicating that the associated attenuation and dispersion are not physically realizable. In effect, for these prohibited values of the frequency exponent, the Fourier components in the dispersion and attenuation cannot combine to give a zero response for negative times. Emerging from the theory is an expression for the complex wavenumber that is complete and exact, and from which all the properties of the dispersion and attenuation may be derived.

2pAO3. Geoacoustic modeling for acoustic propagation in mud ocean-bottom sediments. William L. Siegmann (Dept. of Mathematical Sci., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180-3590, siegmw@rpi.edu) and Allan D. Pierce (Retired, East Sandwich, MA)

This paper reviews research developed since 2007 at BU and RPI, with leadership by the late W.M. Carey, on acoustic properties of mud. Marine mud consists primarily of small clay mineral platelets, which are comprised of crystalline layers and usually carry charge because of isomorphous substitution. Because of resulting electrical forces, the physical nature of mud is considerably different from sand sediments. In particular, as platelets settle under gravity, electrical forces repel face-to-face contact while strong van der Waals forces permit edge-to-face attachment. This platelet aggregation results in card-house structures, for which a tentative quantitative model has been analyzed (J.O. Fayton, RPI Ph.D. thesis). The model preserves basic physics of platelet interactions and leads to low shear wave speed predictions that are consistent with observations. However, because compressional sound speed is independent of the electrostatic interactions, accurate sound speed estimates are available from the Mallock-Wood formula, which also incorporates bubble effects. The basic physical concepts and semi-empirical formulas derived for shear attenuation and its frequency dependence in sandy sediments do not apply for mud, nor do assumptions behind the often cited Biot theory for poro-elastic media. Consideration is given to incorporating geoacoustic models into equations for acoustic propagation in mud.

2pAO4. *In situ* measurements of sediment compressional and shear wave properties in Currituck Sound. Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese, Preston S. Wilson, Thomas G. Muir (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlt.utexas.edu), and R D. Costley (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

This paper reports *in situ* measurements of compressional and shear wave speed and attenuation collected below the sediment-water interface in Currituck Sound, North Carolina. Three measurement locations having distinctly different sediment types will be presented: (1) a near shore site with coarse sand, (2) a shallow-water site with silty fine-grained sand, and (3) an offshore site with loamy clay. At each site, grab samples were collected and later analyzed in the laboratory to quantify physical properties of the sediment, including grain size, density, and porosity. The *in situ* acoustic data were acquired using two measurement systems, each consisting of four probes mounted on a rigid frame. The shear wave system consisted of bimorph elements to generate and receive horizontally polarized shear waves; the compressional wave system was composed of cylindrically shaped piezoelectric elements. Both systems were manually inserted into the sediment, and waveforms were recorded by shipboard equipment. The measured wave speeds and attentions are compared to predicted values from one or more sediment models using the measured physical properties as inputs. The Biot-Stoll model, grain-shearing theory, Mallock-Wood equation, and card-house theory are considered for this analysis. [Work supported by ERDC, ARL:UT, and ONR.]

Contributed Papers

2pAO5. Tentative acoustic wave equation that includes attenuation processes for mud sediments in the ocean. Allan D. Pierce (Retired, PO Box 339, East Sandwich, MA 02537, allanpierce@verizon.net) and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Compressional wave attenuation in mud sediments is less than in sandy/silty sediments and comparable to that of sea water. Experimental results by Wood and Weston (Acustica, 1964) yield 0.07 dB/m for a frequency of 1 kHz, while the extensive data analysis of Holmes, Carey, Dediu, and Siegmann (JASA-EL, 2007) suggests a value of 0.33 dB/m for sandy/silty sediments. The linear frequency dependence over the range of 4 to 50 kHz reported by Wood and Weston suggests that the cause of the attenuation is relaxation processes rather than viscosity processes. The tentative appropriate relaxation process is suggested, in accord with previous ideas reported by Mahmood et al. (Colloids and Surfaces A, 2001), to be the breaking and reforming of van der Waals-force-related bonds between clay platelets that are connected edge to face. The present paper's theoretical development of a single wave equation with relaxation included parallels that discussed on pp. 550–553 of the text by Pierce (Acoustics, 1981) and follows the ideas of Liebermann (Phys. Rev., 1949). Predicted sound speed is in accord with the Mallock-Wood equation, and the attenuation term appears as an integral over past history, in accord with the envisioned frequency-dependent attenuation.

2pAO6. Fine-grained sediments from the poroelastic perspective. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, chotiros@arlt.utexas.edu)

Fine-grained sediments, such as silt and clay, are porous media, consisting of solid particles saturated by water; however, the skeletal structure of the solid component is significantly different than that of larger grained sediments such as coarse and fine sands. The modeling of fine grained sediments

from the poroelastic perspective, such as the Biot theory, can be challenging. A stumbling block is the frequency dependence of attenuation, which from experimental measurement is shown to be proportional to the first power of frequency, while the Biot theory generally predicts a second power of frequency dependence at low frequencies. One approach to reconcile this difference is based on the distribution of pore sizes [Yamamoto and Turgut, 1988]. When a broad distribution of pore sizes is taken into consideration, the frequency dependence approaches the first power of frequency within a limited frequency band. Another approach is to examine the relaxation process at the grain-grain contact, which is governed by a thin fluid film in the contact region. In fine-grained sediments, the contact region is extremely small and nano-fluid effects must be expected. In this study, a derivation of a poroelastic model, based on the relaxation process associated with the grain contact region will be discussed. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

2pAO7. Modeling transmission loss over stratified muddy sediments based on chirp sonar imagery. Dajun Tang, Brian T. Hefner, Jie Yang, and Darrell R. Jackson (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

The Office of Naval Research is sponsoring a shallow water acoustics experiment offshore New Jersey where the sediment consists nominally of a layer of mud over sand, and where the layer thickness is about 10 m. A preliminary chirp sonar survey shows that there exists fine layering within the mud layer. Nearby core data also reveal the existence of fine layers within the mud. One of the interesting acoustic questions is what impact such fine layering within the muddy sediments has on acoustic propagation. Based on the chirp sonar imagery, various plausible models of sediment layering and material composition are proposed and the transmission loss is estimated for those models for the purpose of assisting field experiment planning.

3:25

2pAO8. High-frequency scattering from a sand sediment with an overlying mud layer. Brian T. Hefner, Anatoliy Ivakin, and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

In the spring of 2014, multibeam echo sounder time series data were collected in St. Andrew's Bay, FL, in an area of the bay where the sand sediment was covered by a mud layer. As part of the environmental characterization at the experiment site, the *In-Situ* Measurement of Porosity (IMP2) system collected conductivity probe data 25 cm into the seabed along 3 m tracks. The mud layer appears clearly in the conductivity probe data and had a mean thickness of 13 cm. The roughness power spectrum of the sand/mud interface was 10–20 dB higher than that of the mud/water interface and, more significantly, was 5–10 dB higher than that of sand/water interfaces measured in the Gulf of Mexico during the Target and Reverberation Experiment 2013. The mud layer appears to be preserving the roughness of the sand interface, an effect observed during the Sediment Acoustics Experiment in 2004 following the passage of Hurricane Ivan. The impact of both the increased roughness and the presence of the mud on 180–420 kHz scattering will be assessed through data/model comparisons using the sediment properties measured at the experiment site. [Work supported by ONR and SERDP.]

3:40

2pAO9. Seabed attenuation inversion from broadband reverberation measurements in the Yellow Sea. Lin Wan and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., 104 Robinson Hall, Newark, DE 19716, wan@udel.edu)

The Yellow Sea 1996 experiment was the first joint China-U.S. shallow water experiment conducted in August, 1996. The water depth is 75 m with a deviation of ± 1 m at the experimental area. The seabed sound speed and attenuation in this experiment have been successfully estimated using the sound propagation measurements [Wan, Zhou, and Rogers, *J. Acoust. Soc. Am.* **128**(2), 2010]. During this experiment, the monostatic reverberation data produced by 1-kg TNT shots provide a spatial sampling of the acoustic field and contains the information of seabed acoustic properties. Thus, the present paper uses the inverted seabed sound speed from the aforementioned propagation measurements as a constraint condition and analyzes the reverberation data for seabed attenuation inversion. The broadband normal mode filter is applied to the reverberation data received by a 32-element vertical line array. The reverberation level of the first three modes is obtained as a function of range to extract the modal attenuation coefficients. The seabed attenuation is inferred by minimizing the difference between the theoretical and measured modal attenuation coefficients. The inverted seabed attenuation over a frequency range of 400–1000 Hz shows non-linear frequency dependence.

3:55

2pAO10. The angle of intromission in muddy sediments: Measurements and inferences. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Jan Dettmer, and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Fine-grained sediments, e.g., mud, often exhibit sound speeds less than that of the interstitial fluid. This leads to an angle of intromission (AI), or Brewster angle, at which there is nearly complete transmission of acoustic energy from the water column into the sediment. It is generally difficult to observe this angle directly inasmuch as (1) fine-grained sediments typically have a very low attenuation and (2) underlying granular sediments have a higher sound speed and thus the total seabed reflected field tends to be dominated by the underlying more highly reflective sediments. In some cases, the AI can be directly observed by time-gating the reflected field such that only the fine-grained sediments are insonified. Nominally, the AI is independent of frequency. However, reflection measurements from hundreds of hertz to several kilohertz show a frequency dependence. This dependence is due to sound speed and density gradients (which thus can be estimated from the reflection data). Reflection measurements and core data indicate that geoacoustic gradients depend upon location on the continental shelf. In the

Italian littoral, density gradients are always observed to be positive and decrease seaward. Curiously, sound speed gradients can be positive or negative and have a less obvious correlation with position. [Research supported by ONR Ocean Acoustics.]

4:10

2pAO11. A semi-empirical predictive model for the attenuation properties of fine-grained sediments. David P. Knobles, Steven A. Stotts, and Robert A. Koch (ARL, Univ. of Texas, PO Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu)

It is of interest to be able to predict the acoustic properties (especially the compressional sound speed and attenuation) of sediments that can be classified as mud. An inverse problem is constructed to infer estimates of the surface sediment attenuation, the attenuation gradient, and the frequency exponent and applied to acoustic data in the 10–400 Hz band collected in the Gulf of Oman basin. The results indicate that, unlike a sandy seabed, the surface attenuation is about 0.005 dB/m-kHz and the frequency exponent is close to unity. These estimates are compared to those of an inversion analyses for a shallow-water site in the Gulf of Mexico, which is an environment with similar sediment properties. Further, the inferred attenuation values from both experiments are compared to calculated values from the Buckingham grain-shearing model. The geological histories of the shallow- and deep-water seabeds are compared with the intent of constructing a semi-empirical predictive model for the attenuation properties of fine-grained sediments.

4:25

2pAO12. Shear wave attenuation estimates from Scholte wave data. Gopu R. Potty, James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

Scholte waves are a valuable tool for estimating the shear properties of the ocean bottom sediments. Previously estimates of the shear wave speed were obtained using interface wave data from a small scale experiment conducted in very shallow water in coastal Rhode Island. The University of Rhode Island's shear measurement system consisting of vertical axis and three-axis geophones were used to collect data in 3 m of water. Interface waves were excited by dropping a weight from a research vessel. In this study, we use the geophone data to estimate the depth-averaged shear attenuation using spectral amplitude ratios. The estimated sediment properties will be compared with historic core data from the field test location and other published results for similar type of sediments. The correlation of the shear speed and attenuation to sediment properties such as porosity, grain size and bulk density will be discussed. [Work supported by Office of Naval Research.]

4:40

2pAO13. Wave propagation in muddy sediments using time domain finite difference approach. Holly C. Clark (R&D Ocean Eng., Appl. Physical Sci., 475 Bridge St., Groton, CT 06340, hclark@aphysci.com), Ralph A. Stephen (Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

There is a renewed interest in understanding wave propagation in muddy sediments. In this study, we are using the time domain finite difference approach to investigate wave propagation in a muddy environment representative of the New England Mud Patch south of Martha's Vineyard. The model uses the two-dimensional full-wave time-domain finite-difference code developed at WHOI over the past 35 years. In order to simplify the computation we consider perfectly elastic, isotropic media with uniform step sizes in time and space. The source waveform is a Ricker wavelet with a peak frequency in pressure of 200 Hz. This study considers a 100 m water waveguide ($V_p = 1500$ m/s) overlying 12 m of mud ($V_p = 1530$ m/s, $V_s = 100$ m/s) above a faster bottom ($V_p = 1700$ m/s, $V_s = 500$ m/s). The goal of the modeling was to understand the propagation of compressional, shear and interface waves in the presence of a soft muddy layer. [Work supported by ONR Code 3220A.]

2pAO14. Efficient geoacoustic inversion of spherical-wave reflection coefficients for muddy seabeds. Jorge E. Quijano, Stan E. Dosso, Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, British Columbia V8P 5C2, Canada, jorgeq@uvic.ca), and Charles W. Holland (ARL, Penn State Univ., Reston, VA)

This paper presents efficient geoacoustic inversion of high-resolution reflectivity data for a seabed site consisting of a thick mud layer (~10 m) over multi-layered sandy sediments. Wide-angle, broadband reflection-coefficient data were collected using a towed source and a recording hydrophone located a few meters above the seabed. Trans-dimensional Bayesian inversion is applied to estimate sound-speed, density, and attenuation profiles for an unknown number of sediment layers. Due to the experiment geometry, the data should be modeled as spherical-wave reflection coefficients, which require numerical integration of rapidly oscillating functions and can be computationally intensive. Here, a speed-up of two to three orders of magnitude is achieved by introducing a fast Levin-type numerical integration implemented on a graphics processing unit. The new fast integration algorithm/implementation alleviates time constraints for the spherical-wave inversion, which would otherwise require weeks of computation time, and precludes the use of fast but less-accurate plane-wave theory. Inversion results are presented for simulated data and for experimental data collected on the western Malta Plateau. The analysis of muddy sediments in this work is expected to provide insight into the geoacoustics of the New England mud patch, location of the upcoming ONR Seabed Characterization Experiment 2016 (SCAE16).

5:10

2pAO15. Measurement of shear wave velocity in marine sediments. Aaron S. Bradshaw (Civil and Environ. Eng., Univ. of Rhode Island, 207A Bliss Hall, Kingston, RI 02818, bradshaw@egr.uri.edu) and Christopher D. Baxter (Ocean Eng. and Civil Eng., Univ. of Rhode Island, Narragansett, RI)

This presentation describes the shear wave velocity measurement techniques that have been utilized over that past decade at the Marine Geomechanics Laboratory (MGL) at the University of Rhode Island (URI). The shear wave velocity of marine sediments is of interest to both the underwater acoustics and marine geotechnical engineering communities. In

underwater acoustics, sediment shear wave velocity plays an important role in understanding compressional wave attenuation. Geotechnical engineers utilize shear wave velocity, for example, for the analysis of seismic site response and evaluation of soil liquefaction potential. The shear wave measurement techniques that will be covered include laboratory bender element tests made within the triaxial apparatus that allows for the application of situ stress states on the test specimen. Bender elements have also been used in the lab to make shear wave velocity measurements in reconstituted sand fills at very low stress levels, as well as on intact clay sediments directly within a piston core. More recently, field measurements of shear wave velocity have been made using a system developed by colleagues at URI that involves the measurement and inversion of Scholte waves. The presentation aims to stimulate dialog between the fields of underwater acoustics and marine geomechanics on the topic of shear wave velocity.

5:25

2pAO16. Laboratory and field measurements of Rayleigh waves. Christopher J. Norton (Dept. of Civil Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, cnorton@my.uri.edu), James H. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Aaron Bradshaw, and Christopher D. Baxter (Dept. of Civil Eng., Univ. of Rhode Island, Kingston, RI)

There is a need for rapid and nondestructive sensing of near-surface shear properties of the ground and seafloor. Our approach on land is to measure Rayleigh wave dispersion and invert these measurements to extract a shear wave speed profile. A portable surface wave inversion system has been developed that employs an array of six accelerometers to measure Rayleigh waves excited by simple impulsive sources in the laboratory and in the field. To verify the shear wave speed estimated by the Rayleigh waves, a calibrated bender element system was also developed to provide independent measurements of the shear wave speed profile at selected depths. The bender element system was placed in a test tank initially filled with mason sand and the Rayleigh wave inversion system was deployed on the surface of the sand in the tank. The estimated shear wave speeds were then compared to the results of the bender element system as well measurements by a dynamic cone penetrometer and laser vibrometer. The approach allows for the measurement of shear speeds in for various sediment and soil arrangements in the well-controlled laboratory setting as well in the field. [Work supported by Army Research Office and ONR.]

Session 2pBA

Biomedical Acoustics: Ultrasonic Characterization of Bone

James G. Miller, Chair

*Physics, Washington U Saint Louis, Box 1105, 1 Brookings Drive, Saint Louis, MO 63130**Invited Papers*

1:00

2pBA1. Application of ultrasound transit time spectroscopy to human cancellous bone for derivation of bone volume fraction *in-vitro*. Christian M. Langton and Marie-Luise Wille (Inst. of Health & Biomedical Innovation, Queensland Univ. of Technol., 60 Musk Ave., Brisbane, Queensland 4049, Australia, christian.langton@qut.edu.au)

We have previously demonstrated that ultrasound propagation in complex composite media may be described as an array of parallel sonic rays. The transit time of each sonic ray is determined by the proportion of solid (bone) and fluid (marrow) traversed, the received ultrasound signal being a superposition of all sonic rays. An Ultrasound Transit Time Spectrum (UTTS) for a test sample may be obtained via digital deconvolution of input and output ultrasound signals, describing the proportion of sonic rays having a particular transit time, from which the bone volume fraction (BVF) of the sample may be estimated. In a recent *in-vitro* study, 21 cancellous bone samples, extracted from 5 human femoral heads following total hip replacement, were measured with microCT to derive the true BVF value. Transmission ultrasound signals of 1 MHz were recorded and UTTS-derived BVF calculated. A coefficient of determination (R^2) of 82% was achieved between ultrasound and microCT derived BVF values. Current work is clinically implementing UTTS, noting its potential to estimate bone mineral density, and hence, a means to diagnose osteopenia and osteoporosis using WHO T-score criteria.

1:20

2pBA2. A review of basic to clinical studies of quantitative ultrasound of cortical bone. Pascal Laugier, Simon Bernard, Quentin Vallet, Jean-Gabriel Minonzio, and Quentin Grimal (Laboratoire d'Imagerie Biomedicale, Universite Pierre et Marie Curie-Paris6, CNRS, INSERM, 15 rue de l'ecole de medecine, Paris 75017, France, laugierp@gmail.fr)

The use of quantitative ultrasound (QUS) for bone has increased sharply over the last two decades. QUS offers potential benefits over other diagnostic modalities which include (1) *ex vivo* non-destructive testing, (2) *in vivo* non-ionizing testing, and (3) the ability to assess important cortical bone quality factors which cannot easily be captured with X-ray techniques. These advantages have stimulated widespread interest in basic through clinical studies. For instance, resonant ultrasonic spectroscopy (RUS) has provided gains in *ex vivo* assessment of the anisotropic stiffness and viscoelasticity of bone despite strong damping. RUS is prone to provide answers to questions that remain open regarding the determinants of cortical bone elastic properties. *In vivo* QUS technologies using guided waves (GW) have the potential to reveal cortical bone strength-related factors such as the cortical thickness and porosity. These properties can be estimated by comparing the measured dispersion curves with an appropriate waveguide model of the cortical bone. This presentation reviews the *ex vivo* (RUS) and clinical (GW) studies of cortical bone based on major experimental studies particularly within the past decade. These gains still constitute a prelude to what is to come, given the incessant developments of better instrumentation and signal processing techniques.

1:40

2pBA3. A novel ultrasound device for estimating bone mineral density at the 1/3 radius. Jonathan J. Kaufman (CyberLogic, Inc. & The Mount Sinai School of Medicine, 611 Broadway Ste. 707, New York, NY 10012, jkkaufman@cyberlogic.org), Emily Stein (Columbia Univ. College of Physicians and Surgeons, New York, NY), Gangming Luo (CyberLogic, Inc., New York, NY), Elizabeth Shane (Columbia Univ. College of Physicians and Surgeons, New York, NY), and Robert S. Siffert (OrthopeDC, The Mount Sinai School of Medicine, New York, NY)

This study evaluated a new ultrasound device for estimating bone mineral density (BMD) at the 1/3 radius (1/3R), as well as investigated its ability to discriminate fracture (fx) and non-fx cases. The device measures two net time delay parameters, NTD_{DW} and NTD_{CW} . NTD_{DW} is the difference between transit time of an ultrasound pulse through soft-tissue, cortex and medullary cavity, and transit time through soft tissue of equal distance. NTD_{CW} is the difference between transit time of an ultrasound pulse through soft-tissue and cortex only, and transit time through soft tissue of equal distance. The square root of the product of these two parameters is a measure of BMD at the 1/3R as measured by DXA. A clinical IRB-approved study measured 77 adults using ultrasound and DXA. An age and sex-matched subset of these subjects was used to determine the capability of DXA and ultrasound to discriminate between fx and non-fx cases. A linear regression showed that $BMD_{US} = 0.19 * \{NTD_{DW} * NTD_{CW}\}^{1/2} + 0.28$ and that the linear correlation between BMD_{US} and BMD_{DXA} was 0.93 ($P < 0.001$). We found that ultrasound measurements yield results that are closely associated with those from DXA. In the case-control fx study, we found a small decrease in mean BMD_{DXA} (1/3R) for the fx cases ($P = 0.20$) but a somewhat greater decrease in mean BMD_{US} ($P = 0.05$). In conclusion, the new ultrasound device should enable significant expansion of the identification of bone loss as occurs in osteoporosis and also may ultimately be able to provide additional data on fx risk.

2:00

2pBA4. Piezoelectric response of bone in the MHz range. Mami Matsukawa, Sayaka Matsukawa, and Hiroko Tsuneda (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp)

The healing mechanism of bone fractures by low intensity ultrasound is not yet to be fully understood. As one possible initial process of this mechanism, we focus on the piezoelectricity of bone and demonstrate that bone can generate electrical potentials by ultrasound irradiation in the MHz range. We have fabricated ultrasonic bone transducers using bovine cortical bone as the piezoelectric device. Electrical potentials induced by the ultrasound irradiation were obtained from all transducers. The electrical potentials changed as a function of time during immersed ultrasonic measurements and became stable when the bone was fully wet. The sensitivities of bone transducers were around 1/1000 of a poly(vinylidene fluoride) ultrasonic transducer and did not depend on the magnitude and alignment of hydroxyapatite crystallites in bone. In addition, the magnitude of the induced electrical potentials changed owing to the microstructure in the cortical bone. The potentials of transducers with haversian structure bone were higher than those of plexiform structure bone, which informs about the effects of bone microstructure on the piezoelectricity.

2:20

2pBA5. Relationships among ultrasonic and mechanical properties of cancellous bone. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov), Saghi Sadoughi (Univ. of California, Berkeley, CA), Srinidhi Nagaraja, Maureen Dreher (Ctr. for Devices and Radiological Health, Food and Drug Administration, Silver Spring, MD), and Tony M. Keaveny (Univ. of California, Berkeley, CA)

Most clinical bone sonometers measure broadband ultrasound attenuation (BUA) and speed of sound (SOS) in calcaneus. In addition, backscatter coefficient (BC) has been shown to have clinical utility. The objective of this work was to assess the extent to which ultrasonic measurements convey mechanical properties of cancellous bone. Twenty-five defatted human calcaneus samples were investigated *in vitro*. Normalized BUA (nBUA), SOS, and BC were measured using 500 kHz focused ultrasound transducers. Finite Element Analysis, based on micro-computed tomography images (Scanco microCT 100), was used to estimate stiffness and apparent modulus of the samples. Correlation coefficients from linear regressions were as follows: nBUA vs. stiffness—0.80 ($p < 0.0001$), nBUA vs. apparent modulus—0.81 ($p < 0.0001$), SOS vs. stiffness—0.80 ($p < 0.0001$), SOS vs. apparent modulus—0.84 ($p < 0.0001$), BC vs. stiffness—0.75 ($p < 0.0001$), and BC vs. apparent modulus—0.69 ($p < 0.001$). In conclusion, ultrasonic measurements are very sensitive to mechanical properties of cancellous bone. The mention of commercial products, their sources, or their use in connection with material reported herein is not to be construed as either an actual or implied endorsement of such products by the Department of Health and Human Services.

2:40–3:05 Break

3:05

2pBA6. Sample thickness dependence of Bayesian and modified least squares Prony's analysis methods on systematically shortened bovine cancellous bone. Amber Groopman (Phys., Washington Univ. in St. Louis, 1 Brookings Dr., Compton Hall, Campus Box 1105, St. Louis, MO 63130, nelsonam@wustl.edu), Keith Wear (U.S. Food and Drug Administration, Silver Spring, MD), Yoshiki Nagatani (Electronics, Kobe City College of Technol., Kobe, Japan), Katsunori Mizuno (Underwater Technol. Collaborative Res. Ctr., Univ. of Tokyo, Tokyo, Japan), Mami Matsukawa (Lab. of Ultrasonic Electronics, Doshisha Univ., Kyoto, Japan), Hirofumi Taki (Communications and Comput. Eng., Kyoto Univ., Kyoto, Japan), Jonathan Katz (Phys., Washington Univ. in St. Louis, St. Louis, MO), Mark Holland (Radiology & Imaging Sci., Indiana Univ. School of Medicine, Indianapolis, IN), and James Miller (Phys., Washington Univ. in St. Louis, St. Louis, MO)

The goal of this study was to compare the results from two proposed methods for separating fast and slow waves from mixed-mode signals, one using Bayesian probability theory and one using modified least-squares Prony's (MLSP) [Wear, *J. Acoust. Soc. Am.* **133**, 2490–2501 (2013)], on measurements of cancellous bone. Ultrasonic through-transmission data were acquired on a bovine femoral head specimen for thicknesses ranging from 15 mm to 6 mm. The thickness was reduced in 1 mm increments and measurements were acquired at each sample thickness [Nagatani *et al.*, *Ultrasonics* **48**, 607–612 (2008)]. A Bayesian parameter estimation analysis was performed on the experimentally acquired signals to isolate the fast and slow waves and to obtain estimates of the fast and slow wave's ultrasonic properties. Results for the corresponding ultrasonic properties were estimated using the modified least squares Prony's method plus curve fitting (MLSP + CF) on the same bovine sample by Wear *et al.* (*J. Acoust. Soc. Am.* **136**, 2015–2024). The results show good agreement between the phase velocities estimated by Bayesian and MLSP + CF analysis methods for both the slow wave and the fast wave. Both analysis methods yielded fast and slow wave phase velocities that depended on sample thickness. This may imply that the propagation model, which is used in both methods, may be incomplete.

3:25

2pBA7. Multiscale assessment of cortical bone properties with quantitative ultrasound. Johannes Schneider (Charité Universitätsmedizin Berlin, Berlin, Germany), Simon Bernard, Jean-Gabriel Minonzio (Laboratoire d'Imagerie Biomédicale, Paris, France), Peter Varga (AO Foundation, Davos, Switzerland), Robert Wendlandt (Biomechatronics, Lübeck, Germany), Quentin Grimal, Pascal Laugier (Laboratoire d'Imagerie Biomédicale, Paris, France), and Kay Raum (Charité Universitätsmedizin Berlin, Augustenburger Platz 1, Berlin 13353, Germany, kay.raum@charite.de)

Clinical bone quality assessment still relies strongly on bone mineral density (BMD). It is now well accepted that BMD predicts only around 60% of the individual fracture risk because it depends on geometrical dimensions of the bone and mechanical properties of its cortical shell. The tibia mid-shaft is a clinically relevant site for ultrasound axial transmission (AT) measurements, which allow simultaneous assessment of cortical thickness (Ct.Th) and tissue elasticity. This *ex-vivo* study investigated 19 representative human tibiae (native samples, mid-shaft, age range: 69–94 yrs) using AT with the aim to develop novel ultrasound biomarkers of cortical bone loss. Ct.Th_{AT} was in good agreement with corresponding data from micro-computed tomography ($R^2=0.90$). Resonant ultrasound

spectroscopy was used to measure the transverse isotropic elasticity tensor of extracted bone cubes. Strong linear correlation between axial stiffness (c_{33}) and mass density ($R^2=0.86$) was found. Mechanical testing based strength was moderately correlated to c_{33} ($R^2=0.72$) and mass density ($R^2=0.65$). These findings indicate the potential of cortical elasticity to be an adequate biomarker of bone quality which combined with information about bone geometry (e.g., Ct.Th) could improve clinical fracture risk assessment.

3:45

2pBA8. Backscatter difference techniques for bone assessment using an ultrasonic imaging system. Brent K. Hoffmeister, Morgan R. Smathers, Catherine J. Miller, Joseph A. McPherson, Cameron R. Thurston (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, hoffmeister@rhodes.edu), and Sang-Rok Lee (Health and Sport Sci., Univ. of Memphis, Memphis, TN)

Background: Backscatter difference techniques are being developed to detect changes in bone caused by osteoporosis. Backscatter difference techniques compare the power in one portion of an ultrasonic backscatter signal to the power in a different portion of the same signal. Goal: Evaluate the feasibility of using an ultrasonic imaging system to perform backscatter difference measurements of bone. Procedure: Ultrasonic signals and images were acquired from 24 specimens of bone using an ultrasonic imaging system (Terason) with a 5 MHz linear array transducer. The signals were analyzed to determine the normalized mean backscatter difference (nMBD) between two gated portions of each signal. The images were analyzed to determine the normalized pixel value difference (nPVD) between regions of interest (ROIs) positioned at two different depths. Results: nMBD demonstrated strong linear correlations with bone mineral density that ranged from 0.83 to 0.87. nPVD performed less well, yielding correlations that ranged from 0.42 to 0.81, depending on ROI separation. Conclusions: It is feasible to apply the backscatter difference technique to ultrasonic imaging systems for the purpose of bone assessment. Better results are obtained by analyzing the signals rather than the images.

4:05

2pBA9. Ultrasound backscattering from cancellous bone *in vivo*. Dean Ta, Tao Tang, Chengcheng Liu, and Weiqi Wang (Dept. of Electron. Eng., Fudan Univ., 220 Handan Rd., Shanghai 200433, China, tda@fudan.edu.cn)

Osteoporosis is a musculoskeletal disease characterized by a loss of bone mass and a deterioration of trabecular bone microarchitecture. In recent years, significant progress has been made in the quantitative ultrasound technique, and it has become well established as a noninvasive tool for the assessment of bone status. This report will show the ultrasonic backscatter theory in cancellous bone, analyze the ultrasonic backscattering signals, and to investigate correlations among backscatter parameters and bone mineral density (BMD) *in vivo*. Ultrasonic backscattering measurements were performed on 1226 subjects (601 males and 625 females) at the right calcaneus *in vivo* using the novel ultrasonic backscatter bone system. Then, the subjects underwent DXA for the BMD at sites located on the lumbar spine (sBMD) and left hip (hBMD). Spectral centroid shift (SCS), mean trabecular bone spacing (Tb.Sp), backscatter coefficient (BC), and apparent integral backscatter coefficient (AIB) were calculated at the central frequencies of 3.5 and 5.0 MHz. Linear regression showed that the SCS at 3.5 MHz exhibited negative correlations with the sBMD ($R=-0.66$) and hBMD ($R=-0.64$). The SCS at 5.0 MHz was also found to be closely related to the sBMD ($R=-0.68$) and hBMD ($R=-0.66$). Tb.Sp at 3.5 MHz and 5.0 MHz exhibited negative correlations with the sBMD ($R=-0.74$) and hBMD ($R=-0.66$). The correlations between backscatter parameters and BMD were found to be statistically significant in both the male and female groups.

4:25

2pBA10. Quantitative ultrasound imaging reconstruction in peripheral skeletal assessment. Yi-Xian Qin and Jesse Muir (Biomedical Eng., Stony Brook Univ., BioEng. Bldg., Rm. 215, Stony Brook, NY 11794, yi-xian.qin@stonybrook.edu)

Disuse osteopenia affect mineral density, microstructure and integrity of bone, which lead to increased risk of osteoporosis and fracture during long term space mission. In order to provide a non-ionizing, repeatable method of skeletal imaging, a novel hand-held scanning confocal quantitative ultrasound (QUS) device has been developed. A mobile scanning ultrasound images were collected using ten sheep tibia and eight human volunteers at the wrist, forearm, elbow, and humerus with the arm submerged in water. A custom MATLAB software used a least-error algorithm to adjust adjacent lines and remove unwanted image waver from hand movement. Ultrasound attenuation (ATT) values of 42.4 ± 0.6 dB and 41.5 ± 1.0 dB were found in water and gel coupling, respectively. Scans of the human subject revealed detailed anatomy of the humerus, elbow, forearm, and hand. Repeat measures of the distal radius found an attenuation of 37.2 ± 3.3 dB, with negligible changes cause by wrist rotation of $\pm 10^\circ$ (36.4 to 37.3 dB) indicating small changes in scan angle would not significantly affect bone quality measurements. The hand-held QUS device allows for rapid skeletal imaging without ionizing radiation. Information gathered from ultrasonic attenuation and velocity can provide detailed information on localized bone fragility as well as identify fractures.

Contributed Paper

4:45

2pBA11. Transcranial time-of-flight estimation using backscattered infrared data. Qi Wang, Mark Howell (Cleveland Clinic, 9500 Euclid Ave./ND 20, Cleveland, OH 44195, qiqiawang83@gmail.com), Namratha Reganti (Biomedical Eng., Case Western Reserve Univ., Cleveland, OH), and Gregory T. Clement (Cleveland Clinic, Cleveland, OH)

Recently, we reported on our observation that diffuse infrared (IR) light transmitted through human skull bone exhibits an intensity that correlates positively with acoustic (1 MHz) time-of-flight data acquired at equivalent locations [POMA 22, 020002 (2015)]. Presently, we investigate the potential to exploit this correlation as a means of correcting for skull-induced

aberration. For measurement, a dual ultrasonic/infrared array capable of illuminating a localized region of the skull bone and recording the backscattered light was constructed. The array design utilized 940 nm IR emitters (TSAL4400, 3-mm-diameter) and a single detector (TEFD4300-3 mm) configured to transmit and record under the face of an ultrasonic PZT ring transducer. Initial tests consisted of a three-emitter configuration, with fiberglass-reinforced foil used to isolate the direct infrared signal between the emitters and the receiver. Effects of skin were considered by attaching a skin-mimicking layer to the skull and comparing results with and without the layer. Data acquired on a set of eight skull samples will be presented. Preliminary results show good correlation between ultrasound and IR intensity. [Work supported by NIH award R01EB014296.]

Invited Paper

2pBA12. Numerical investigation of fast and slow longitudinal waves backscattered from various depths inside cancellous bone. Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, 679-3 Nishioka, Uozumi, Akashi 674-8501, Japan, hosokawa@akashi.ac.jp)

Ultrasound backscattering in cancellous bone was numerically investigated using three-dimensional (3D) finite-difference time-domain (FDTD) simulations with microcomputed tomographic (μ CT) models of the bone. The cancellous bone models with various thicknesses, in which artificial absorbing layers were set at the surfaces opposite to the ultrasound-transmitted surfaces, were prepared. The backscattered waveform inside cancellous bone was isolated by deriving the difference between two simulated waveforms obtained using the bone models with different thicknesses, and the backscatter properties from various depths of the bone were investigated. When an ultrasound pulse wave was transmitted in the direction parallel to the main orientation of the trabecular network, the backscattered waves from the deep bone depths could clearly separate into two waves, which could correspond to the backscattered waves of the fast and slow longitudinal waves propagating mainly in the trabecular elements and the pore spaces. For each of the fast and slow waves, the backscatter coefficient, which was the ratio of the backscattered fast/slow wave amplitude to the incident fast/slow wave amplitude, was obtained. The results showed that both backscatter coefficients were weakly correlated with the bone depth.

Contributed Papers

2pBA13. Broadband ultrasound scanning and frequency sweep measurement for trabecular bone with novel wideband single crystal transducer. Jian Jiao, Xiaofei Li, Liangjun Lin (Biomedical Eng., Stony Brook Univ., 100 Nicolls Rd., 212 BioEng. Bldg., Stony Brook, NY 11794, jian.jiao@stonybrook.edu), Raffi Sahul, Ed Nesvijski (TRS Technologies Inc., State College, PA), and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., Stony Brook, NY)

Quantitative ultrasound has been developed as a radiation-free technique to evaluate both bone density and strength. The ultrasound parameters, such as velocity (UV), attenuation (ATT), and broadband attenuation (nBUA), have been widely used as indicators for bone health status. In our study, a novel spiral-wrapped wideband ultrasound transducer is developed by TRS. Bovine trabecular bone samples (15 mm thickness) were prepared for frequency scanning. In each group, TRS transducer was used to emit the frequency from 0.5 to 7 MHz to propagate through samples; and pulses were received by the hydrophone. Frequency sweep response mostly correlated to bone density at frequency under 3 MHz, and barely showed any correlation with bone density at a frequency above 3 MHz, as higher frequency ultrasound energy attenuated significantly in trabecular bone. With controlled bone density reduction by 34.71%, UV decreased from 2575.03 ± 34.38 m/s to 1717.15 ± 48.19 m/s, ATT decreased from 3.66 ± 0.71 dB to 2.21 ± 0.76 dB, and nBUA decreased from 2.09 ± 0.96 dB/MHz/cm to 1.21 ± 0.47 dB/MHz/cm. Generally, bone density decrease resulted in UV, ATT, and nBUA decreases, but were sensitive to the change of the wideband frequency from 0.5 MHz to 0.8 MHz. This new wideband transducer offers more information than the conventional ultrasound transducer, and can thus be used as a modality to evaluate bone health status.

2pBA14. Ultrasonic wave velocities in radial direction of bovine cortical bone. Yuma Nishimura (Faculty of Sci. and Eng., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, duo0347@mail4.doshisha.ac.jp), Satoshi Kawasaki (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

The ultrasonic axial transmission (AT) techniques have been proposed to evaluate the cortical bone of arms and legs. For the mode analysis of propagating waves, the information of anisotropy and heterogeneity of the cortical bone is important. Yamato has pointed that longitudinal wave velocity in the

axial direction of bovine cortical bone changes due to the measurement positions and microstructures. In this study, we have investigated longitudinal wave velocities in radial direction of cortical bone. The cuboid samples were obtained from the cortical bone of bovine femur. During ultrasonic pulse immersion experiments, a sinusoidal ultrasonic wave in the range from 3 to 10 MHz was transmitted from a PVDF transducer, which passed through the sample in the radial direction. By reducing the sample thickness gradually, wave velocities were measured. The average velocities in radial direction of samples were around 3450 m/s and depended on the positions of the samples in the femur. They variation of the velocities due to the thickness was about 7%. These data give us the information of heterogeneity of cortical bone in radial direction which should be considered in the AT techniques. [1] Y. Yamato, *et al.*, *Jpn. J. Appl. Phys.*, **47**, 4096–4100 (2008).

2pBA15. FDTD simulations of ultrasonic wave propagation in the cortical bone with heterogeneity. Toshiho Hata (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tatara, Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, dmo1008@mail4.doshisha.ac.jp), Yoshiki Nagatani (Kobe City College of Technol., Kobe, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

The ultrasonic axial transmission (AT) technique is known to assess the cortical bone of femur and tibia. After the first application of the finite difference time domain (FDTD) method [1], the wave propagation in bone has been discussed by both simulation and experimental studies. The wave propagation depends on the wave velocity, bone geometry (curvature, cortical thickness), anisotropy, heterogeneity, etc. In this study, we have investigated the ultrasonic wave propagation in the cortical bone with heterogeneity (stiffness distribution) by FDTD method. Using the stiffness distribution of bovine cortical bone, experimentally measured by Yamato and Nakatsuji [2], [3], wave propagation of one cycle of sinusoidal wave at 0.5~1 MHz was investigated, assuming cuboid samples. The wave velocities inside the bone were higher than those of the bone surfaces. The complex propagating waves were slightly concentrated at the bone surface. The results indicate that the stiffness distribution is also an important factor to understand the wave propagation. [1] E. Bossy *et al.*, *J. Acoust. Soc. Am.* **115**, 2314 (2004). [2] Y. Yamato *et al.*, *Jpn. J. Appl. Phys.* **47**, 4096 (2008). [3] T. Nakatsuji *et al.*, *Jpn. J. Appl. Phys.* **50**, 07HF18 (2011).

Session 2pED

Education in Acoustics: Topics in Acoustics Education

Andrew A. Piacsek, Chair

Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Contributed Papers

1:30

2pED1. Acoustic engineering by design—Example of a service learning course for undergraduate engineering students. Eoin A. King, Philip P. Faraci, and Robert D. Celmer (Acoust. Prog. and Lab, Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

The University of Hartford has two ABET-accredited undergraduate engineering programs in the area of acoustics—a Bachelor of Science in Mechanical Engineering with acoustics concentration and a Bachelor of Science in Engineering with a major in acoustical engineering and music. Students participating in both of these programs take an acoustic engineering design course with a focus on service learning during their sophomore year. Service learning is an alternative to the traditional teaching model and offers students the opportunity to apply their classroom studies to local issues in the community; thus, students achieve academic objectives by meeting real community needs. Each year members of the community approach the University to request assistance with rooms exhibiting poor acoustic characteristics due to improper reverberation times and/or excessive background noise levels. Students are required to design and communicate solutions to these issues in a manner similar to a professional consultation. Recent examples include a science center exhibit hall, a restaurant, an architecture studio classroom, and a retirement community auditorium. This presentation will provide examples of such projects completed as part of this class. Challenges and benefits associated with this service learning course for acoustic engineering undergraduate students will be discussed.

1:45

2pED2. Development of an acoustics outreach program for the deaf. Cameron T. Vongsawad, Mark L. Berardi, Kent L. Gee, Tracianne B. Neilson, and M J. Lawler (Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604, cvongsawad@byu.net)

The Hear and Seemethodology (Groppe, 2011) has often been used as a means of enhancing pedagogy by focusing on the two strongest learning senses, but this naturally does not apply to deaf or hard of hearing students. Because deaf students' prior nonaural experiences with sound will vary significantly from those of students with typical hearing, different methods must be used to build understanding. However, the sensory-focused pedagogical principle can be applied in a different way for the Deaf by utilizing the senses of touch and sight, called here the "See and Feel" method. This presentation will provide several examples of how acoustics demonstrations have been adapted to create an outreach program for a group of junior high students from a school for the Deaf and discuss challenges encountered.

2:00

2pED3. Using acoustics and vibrations to teach experimental concepts and technical communications. Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Diego Turo, Joseph F. Vignola, and John A. Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work describes an experience-based conceptual course in vibrations and acoustics. This one-hour course is designed with the goal of framing general experimental issues, interpreting experimental results, and developing technical communication skills in engineering by way of acoustics and vibrations experiments. Like other laboratory experiences, the goal is to support the skill of making the connection between experimental results and the physical system. Unlike other courses offered at East Carolina University and The Catholic University of America, the primary learning objectives are those connections and technical communication skills rather than the acoustics and vibrations content. Experimental issues explored include error, repeatability, linearity, uncertainty, and general experimental design. This particular set of acoustics and vibrations experiments is well suited to this purpose because they do not require extensive laboratory infrastructure, they are inherently safe, and they use commonplace tools. These tools include a speaker, microphone, and accelerometer, which the students already have as part of their laptop and/or smartphone. These tools, along with a copy of MATLAB, provide the opportunity to perform a wide array of experiments, which can illustrate sophisticated concepts using these basic, familiar items.

2:15

2pED4. A simply constructed condenser microphone as a teaching aid. Randall A. Ali (Dept. of Elec. and Comput. Eng., Univ. of the West Indies, Eastern Main Rd., St. Augustine na, Trinidad and Tobago, randall.ali@sta.uwi.edu) and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

To gain a deeper appreciation and a better understanding of the operation of condenser microphones, it is instructive to have a tangible device, which can undergo a fair amount of experimentation. Low cost condenser microphones used in laptops, mobile phones, and other similar devices can be purchased, but not necessarily deconstructed to serve as an effective learning tool. On the other hand, condenser microphones used in recording studios can be taken apart, but can become quite expensive and making any elaborate modifications to the design may prove challenging. As an alternative, we present a functional and cost-effective condenser microphone that can be simply constructed from readily available materials. The device enables students to make changes to the diaphragm material, back plate, electrostatic gap spacing, and the electronic circuit for connection to a lower impedance device. A SPICE model of the condenser microphone is also presented, which was used to guide the design and can provide further insight for the student.

2:30

2pED5. Impedance tube experiments as a tool to teach acoustics of porous media. Diego Turo, Aldo A. Glean, Chelsea E. Good, Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington DC, 20064, diegoturo@gmail.com), Teresa Ryan (Eng., East Carolina Univ., Greenville, NC), and John A. Judge (Mech. Eng., The Catholic Univ. of America, Washington DC, DC)

As a practical matter, sound propagation in many everyday situations includes interactions with porous materials. Such materials include textiles, foams, gravel, grass-covered ground, asphalt, and trees. Modeling sound propagation in complex environments, such as passenger cabins or urban spaces, with porous materials like these is essential to understanding realistic acoustics problems. The theory behind the sound absorption in porous material is not typically included in introductory acoustics courses. This is, at least in part, attributable to the mathematical complexity of the topic. Despite this barrier, acoustic properties of porous material can be easily introduced to typical undergraduate students using simple experiments to demonstrate the acoustic effects of different materials. This presentation describes acoustic property measurements of rigid-frame porous material with straight cylindrical pores as well as foam and soil included in an "Introduction to Acoustics" class at the Catholic University of America. The apparatus is composed of a LabVIEW measurement interface, a pair of microphones, and a simple aluminum impedance tube. The modular design allows students to adapt the software or hardware for other experiments. These hands-on experiences are intended to impart a conceptual understanding of the theory of sound absorption and the acoustic performance of sound absorbing (porous) media.

2:45

2pED6. Energy flux streamlines versus acoustic rays for modeling an acoustic lens: Testing and refinement. Cleon E. Dean (Phys., Georgia Southern Univ., PO Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu) and James P. Braselton (Mathematical Sci., Georgia Southern Univ., Statesboro, GA)

As an extension of recently published experimental work [Cleon E. Dean and Kendez Parker, "A ray model of sound focusing with a balloon

lens: An experiment for high school students," J. Acoust. Soc. Am. **131**, 2459–2462 (2012)], preliminary results comparing energy flux streamlines [David M. F. Chapman, "Using streamlines to visualize acoustic energy flow across boundaries," J. Acoust. Soc. Am. **124**, 48–56 (2008)] versus acoustic rays for visualizing the energy flow inside and in the focal region of an acoustic lens in the form of a carbon dioxide filled balloon in air were presented. The sound field was expanded in the usual Legendre polynomials and spherical Bessel functions, and the energy flux vectors at points throughout the regions of interest were calculated [J. Adin Mann III, et al., "Instantaneous and time-averaged energy transfer in acoustic fields," J. Acoust. Soc. Am. **82**, 17–30 (1987)]. Deficiencies in the streamline plotting routines used in this earlier version of Mathematica and subtle differences between acoustic rays and acoustic energy flux streamlines lent itself to an inaccurate perception of the results. This talk uses Mathematica 10 routines to revisit these results and concentrates on testing and verification of the conclusions from the previous work.

3:00

2pED7. Development of boundary element method to analyze acoustic models with comparison to finite element model. Mahdi Farahikia and Ronald N. Miles (Mech. Eng., SUNY Binghamton, 13 Andrea Dr. Apt. A, Vestal, NY 13850, mfarahi1@binghamton.edu)

Numerical and computational models provide valuable information about engineering designs. One of their desirable benefits is reduction of expenses and feasibility of observing the change in behavior of a model with varying design parameters. In this paper, a boundary element method that was developed in Matlab is described. The approach in developing this method is based on the Green's function for an unbounded acoustical domain. This model allows reading of computer aided designs to further analyze them in acoustic domain. The results obtained from this method are compared with equivalent finite element models carried out in ANSYS and exact analytical solutions to check for accuracy. Steps in analyzing acoustic models in ANSYS and Matlab are also explained in this paper.

TUESDAY AFTERNOON, 19 MAY 2015

KINGS 5, 1:30 P.M. TO 2:55 P.M.

Session 2pNSa

Noise and Psychological and Physiological Acoustics: Application of Psychoacoustics in Noise

Klaus Genuit, Cochair

HEAD Acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Brigitte Schulte-Fortkamp, Cochair

Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Invited Papers

1:30

2pNSa1. Environmental noise assessment by means of psychoacoustics and its relation to *in-situ* annoyance judgments of urban noise. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

It is obvious that environmental noise assessment only based on the consideration of L_{Aeq} values cannot determine the character of environmental noise and its specific reactions and responses evoked. Noise is not only perceived in terms of lower or higher (averaged) sound pressure levels; noise leads to different auditory sensations and perceptions, which are influenced by context. The introduction of

psychoacoustics allows for describing the character of noise more in detail and for predicting human responses to noise more reliably. In order to investigate the link between psychoacoustic parameters and annoyance judgments, soundwalk data including *in-situ* noise judgments from several measurement campaigns were analyzed. It turned out that by taking into account psychoacoustic parameters more variance in noise annoyance data is explained. Based on the analysis and results of the soundwalk data, the questions are addressed about how long must be measured to encompass all relevant acoustical situations and how the measurement of short-term reactions are related to long-term noise effects. Based on the presented case studies, it is intended to gain insight into the benefits and limitations of psychoacoustics with respect to environmental noise assessment.

1:50

2pNSa2. Status of psychoacoustics in noise analysis. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

Psychoacoustics provides knowledge about the relationship between acoustical stimuli and provoked hearing sensations for specific “one-dimensional” sensations like loudness, sharpness, roughness, or fluctuation strength. Psychoacoustic measures are often applied in the context of sound quality investigations. Sound quality has to be understood as a multi-dimensional phenomenon related to the perception of sensory pleasantness (sound character) and suitability of sound in context. It is widely accepted that psychoacoustic measures offer better agreement with auditory sensation than conventional A-weighted sound pressure levels and spectra do. In particular, the psychoacoustic parameter loudness gains extensively in significance, because it shows a higher correspondence with the sensation of volume (loudness) than any sound level indicator. Thus, loudness represents a dominant feature for sound quality evaluation and is frequently applied in numerous applications. In the field of *environmental noise* the main focus lies on the exact measurement and description of the acoustical situation in a pure physical sense, whereas the community noise perspective tries to bridge acoustical exposure and the human assessment of noise in sense of annoyance level. In contrast to it, the disciplines of psychoacoustics, sound quality, and soundscape put more emphasis on the perceiving human being. This paper gives an overview about the status of psychoacoustics with respect to application in noise.

Contributed Paper

2:10

2pNSa3. Tuning the cognitive environment: Sound masking with “natural” sounds in open-plan offices. Alana G. DeLoach (School of Architecture (Architectural Acoustics), Rensselaer Polytechnic Inst., PO Box 413, Nantucket, MA 02554, deloaa3@rpi.edu), Jeff P. Carter, and Jonas Braasch (School of Architecture (Architectural Acoustics), Rensselaer Polytechnic Inst., Troy, New York)

With the gain in popularity of open-plan office design and the engineering efforts to achieve acoustical comfort for building occupants, a majority of workers still report dissatisfaction in their workplace environment. Office acoustics influence organizational effectiveness, efficiency, and satisfaction through meeting appropriate requirements for speech privacy and ambient sound levels. Implementing a sound masking system is one tried-and-true

method of achieving privacy goals. Although each sound masking system is tuned for its specific environment, the signal—random steady state electronic noise, has remained the same for decades. This session explores how “natural” sounds may be used as an alternative to this standard masking signal employed so ubiquitously in sound masking systems in the contemporary office environment. As an unobtrusive background sound, possessing the appropriate spectral characteristics, this proposed use of “natural” sounds for masking challenges the convention that masking sounds should be as meaningless as possible. Based on psychophysical data and a sound-field analysis through an auditory model, we hypothesize that “natural” sounds as masking sounds have the ability (with equal success as conventional masking sounds) to meet standards and criteria for speech privacy while enhancing cognitive functioning, optimizing the ability to concentrate, and increasing overall worker satisfaction.

2:30–2:55 Panel Discussion

Session 2pNSb**Noise and Psychological and Physiological Acoustics: Mobile Technology Solutions for Hearing Loss Prevention**

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

Michael G. Heinz, Cochair

*Speech, Language, and Hearing Sciences & Biomedical Engineering, Purdue University, 500 Oval Drive, West Lafayette, IN 47907***Chair's Introduction—3:30*****Invited Papers*****3:35****2pNSb1. Accurate mobile noise measurements: Challenges for the app developer.** Benjamin Faber (Faber Acoust., LLC, 277 S 2035 W, Lehi, UT 84043, ben@faberacoustical.com)

The ubiquity of mobile computing devices, their connection to the cloud, and their media-centric nature makes them attractive for wide scale acquisition of acoustical information, including human noise exposure in various environments and at various times. One concern in harnessing the potential of such a network of devices is ensuring that each device is able to acquire the desired acoustical information with sufficient accuracy. It is possible to make accurate acoustical measurements, as long as care is taken to ensure that the hardware, software, and users all work together to ensure success. System software, signal acquisition hardware, microphones, the mobile devices and the people who use them each present certain issues and challenges to the process of making accurate and reliable acoustical measurements. Various of these issues and challenges will be presented and discussed from an app developer's perspective.

3:55**2pNSb2. Use of smartphone sound measurement apps for occupational noise assessments.** Chucri A. Kardous (National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati, OH 45226, ckardous@cdc.gov) and Metod Celestina (EA LAB, Ljubljana, Slovenia)

NIOSH conducted two studies to examine the accuracy of smartphone sound measurement applications (*apps*). The first study examined 192 sound measurement *apps* on the Apple (iOS) and Google (Android) platforms. Only 10 iOS *apps* met our selection criteria for functionality, measurement metrics, and calibration capability. The studies compared the performance of the *apps* with a reference microphone and with a professional type 1 sound level meter and a type 2 noise dosimeter. The results showed 4 iOS *apps* with means of differences within ± 2 dB(A) of the reference microphone. The Android-based *apps* lacked the features and functionalities found in iOS *apps* and showed a wide variance between the same *app* measurements on different devices. A follow-up study of the 4 iOS *apps* using calibrated external microphones (MicW i436 and Dayton Audio IMM-6), showed an even closer agreement with professional meters. Overall, the studies suggest that certain *apps* may be used for some occupational noise assessments but only if properly calibrated and used within the hardware limits of the mobile devices. NIOSH and EA LAB are collaborating to develop an occupational sound measurement *app* for iOS devices in an effort to improve awareness of the noise hazards in the workplace.

4:15**2pNSb3. Acoustic requirements for audiometric testing and hearing protector fit-testing with mobile platforms.** 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)

Mobile technology platforms present new challenges for implementing hearing loss prevention programs. Laptop computers, smartphones, and tablets have capabilities to generate 16-bit and sometimes 24-bit audio stimuli for hearing tests. For testing hearing at threshold, the noise floor of the sound card of the computer can pose a limitation. During the development of sound compression standards (e.g., MPEG, AAC, MP3), 24-bit resolution and 96 kHz sampling is necessary to encompass the requirements for dynamic level the frequency range of human hearing. However, hearing screening and fit-testing can be accomplished with 44 kHz sampling rates. However, the bit resolution deserves careful treatment to ensure that output is accurate. This paper will describe features for mobile applications for audiometric testing and hearing protector fit-testing. Several external DAC platforms on Windows 7 based laptops along with a small sample of iOS and Android devices have been tested and demonstrate differing levels of performance. Linearity of the systems and output levels for a standard audiometric headphone will be presented.

2pNSb4. Comparison of headphones for audiometric screening and hearing protector fit-testing. David C. Byrne (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati, OH 45226-1998, zne2@cdc.gov), Laura Schmitt (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Accurate hearing screening in hearing loss prevention programs is entirely dependent upon sufficient isolation of the subject from any surrounding noise as well as adequate acoustical performance of the headphones that are used. Assessment of hearing protector attenuation by newly developed individual fit-test systems uses many elements borrowed from standard audiometric testing. To assess the suitability for this type of testing, the National Institute for Occupational Safety and Health evaluated the attenuation characteristics, maximum output levels, linearity, and total harmonic distortion of three different circumaural headphones: the FitCheck headphone with a TDH-39 driver, the Sennheiser HDA-200, and Sennheiser HDA-300 headphones. Attenuation measurements were obtained with a laboratory-based real-ear attenuation at threshold system, and the electroacoustic measurements were conducted on a standard IEC318 coupler. The long term goal of this effort is to determine whether attenuation differences or the performance characteristics of the different styles of headphones affect the resulting earplug attenuation values when measured with an earplug fit-test system.

Contributed Paper

4:55

2pNSb5. Characterization of the smart-phone radiated noise using near-field acoustic holography. Donghyun Kim, Taeyoung Park, and Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A391, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, kimdh88@yonsei.ac.kr)

Smart phones have dramatically evolved on the functional and aesthetic fronts in recent years. However, as smart phones get slimmer, problems arise where the noise radiated from the smart phone, albeit small in SPL, interferes with the conversation and degrades the call quality. The noise

radiation is primarily due to the vibration of the internal electronics components and its transmission through the circuit board and the casing. In this paper the nearfield acoustic holography (NAH) is used to characterize the 3-D sound field of the smart-phone radiated noise, which is tailored specifically to deal with the small size and the low SPL of the sound radiator. Quantities such as acoustic pressure, particle velocity, and intensity at the smart phone surface are obtained and utilized to identify parts that are responsible for noise generation. In this respect, the NAH proves to be an effective tool for the characterization and control of the smart phone acoustics.

2p TUE. PM

TUESDAY AFTERNOON, 19 MAY 2015

KINGS 1, 1:00 P.M. TO 4:00 P.M.

Session 2pPA

Physical Acoustics: General Topics in Physical Acoustics II

Kevin M. Lee, Chair

Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

1:00

2pPA1. Ultrasound propagation in concentrated random dispersions of spherical particles: Thermal- and shear-mediated contributions to multiple scattering. Valerie Pinfield and D Michael Forrester (Chemical Eng. Dept., Loughborough Univ., Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk)

Ultrasonic techniques offer advantages for process monitoring for dispersions of colloidal or nano particles; such materials occur in a wide variety of process industries. However, the application of ultrasonic techniques has been limited by the inaccuracy of ultrasonic propagation models used to interpret the measurements (typically attenuation spectra). Multiple scattering models, such as the Lloyd and Berry model [Proc. Phys. Soc. London

91, 678 (1967)], have been used with great success in relatively dilute colloidal dispersions, but fail for higher concentrations, smaller particles, and low frequencies, primarily due to the neglect of thermal- and shear-mediated effects. We present a modified multiple scattering model that includes these thermal- and shear-wave contributions and explore their significance. The model develops work by Luppé, Conoir and Norris [J. Acoust. Soc. Am. 2012 (131) 1113] for compressional, thermal and shear wave propagation. We identify the dominant scattering contributions for emulsions [Pinfield, J. Acoust. Soc. Am. **136**, 3008 (2014)] and suspensions and develop analytical forms for them. Numerical calculations demonstrate the contribution of the additional multiple scattering effects to the compressional wave speed and attenuation through the emulsion or suspension. The calculations are compared with previously published experimental data.

1:15

2pPA2. Direct visualization of shear waves in viscoelastic fluid using microspheres. Cecille Labuda (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38677, cpembert@olemiss.edu), Connor M. Tierney, E. G. Sunethra K. Dayavansha (Phys. and Astronomy, Univ. of MS, University, MS), and Joseph R. Gladden (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

Wormlike micellar fluids are viscoelastic and can thus support shear waves. Shear waves in 500 mM CTAB-NaSal micellar were visualized by seeding the fluid with microspheres. This method was compared to visualization by observation of birefringence patterns induced by fluid strain in response to shear stresses. Shear speeds measured using both techniques were consistent. Particle displacement was observed to be a sinusoidal function of time and displacement amplitude decreased quadratically with distance from the source. This supports the possibility of using particle amplitude measurements as a measure of attenuation even at low fluid concentration where birefringence visualization techniques fail.

1:30

2pPA3. Determination of the phase speed in a bubbly liquid at the single bubble resonance in an impedance tube using a transfer function technique. Stanley A. Cheyne, Hugh O. Thurman, Walter C. McDermott, and Charles G. Kelley (Dept. of Phys. & Astronomy, Hampden-Sydney College, Hampden-Sydney, VA 23943, scheyne@hsc.edu)

The Transfer function method was used to determine the phase speed in a bubbly liquid at the single bubble resonant frequency. The Transfer function technique is widely used to determine the impedance of an absorbing sample. Once the impedance of the sample is found, the phase speed can be determined. White noise was generated and allowed to propagate upward in a vertical, steel, impedance tube. The top was terminated by a bubble cloud generated by pressurized air forced through hypodermic needles. The sound spectrum was measured at two different points in the pure water by carefully moving a single hydrophone. The transfer function was then determined and used to calculate the phase speed in the bubbly liquid.

1:45

2pPA4. Acoustic measurements and modeling of air-filled underwater resonator cavities. Laura M. Tseng, Kevin M. Lee (Appl. Res. Laboratories: The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, ltseng@utexas.edu), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This project investigated the near-resonance acoustical properties of submerged air-filled resonator cavities intended for use in underwater noise abatement systems, exploring a potential alternative to encapsulated bubbles. The underwater resonator cavities are air-filled containers resembling Helmholtz resonators, inverted and submerged in water, creating a direct interface between the volume of air trapped inside the cavity and the surrounding water. Experiments were performed with a prototype resonator cavity in a closed water-filled laboratory tank operated in the long wavelength limit (under 500 Hz) and the resonance frequencies and Q-factors were measured for various air-fill volumes. A finite-element simulation was used to examine the behavior of the resonator cavities within the experimental tank apparatus, yielding good agreement with measurements. Finally, free-field models were also developed (a finite-element numerical model and a Helmholtz-resonator-based analytical model), both of which yielded good agreement with the tank measurements. The measurements and models presented here indicate that inexpensive and convenient sub-wavelength laboratory tank measurements, along with simple analytical models can be used to accurately design and verify the free-field behavior of low frequency underwater noise abatement resonators.

2:00

2pPA5. Acoustic emissions of ice fracture. Melisa Yashinski, Rintaro Hayashi, Tyler Heei-Wai, and John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

Acoustic signals of ice movement and breaking has been recorded in previous field studies, but the specific role of salinity on ice fracture noise remains an open research topic. In lab studies, salinity has been shown to alter the mechanical properties but the associated emissions are not well understood. In this study, high-speed optical visualization is done in conjunction of acoustic measurements of ice fracture for three salinity values in air and underwater. Fracture strength is determined through three point bending and the role of columnar structure representative of Arctic ice is explored. The elastic-plastic behavior is characterized through time-frequency analysis.

2:15

2pPA6. Direct computation of acoustic scattering and difference frequency generation. Chrisna Nguon (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu), Olo-lade Mudasiru, Sameera Mogulla, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MA)

A simulation of the scattered field at the difference-frequency resulting from the nonlinear interaction of two beams at differing frequencies incident on a fluid scatterer is performed. The total wavefield response is modeled using a finite-difference time-domain approximation and a perfectly matched layer to estimate the spatial scattering features of the medium. A pulsed ultrasonic transducer beam is modeled and its interaction with a fluid scatterer with spatially varying compressibility contrast is visualized. The second-order pressure field is determined as a response to the difference-frequency source field constructed from the first order-response of the scatterer to the two incident beams as well as a spatial distribution of the nonlinear parameter. The performance and accuracy of the numerical simulation is examined.

2:30–2:45 Break

2:45

2pPA7. Evaluation of the spatial impulse response of planar ultrasonic radiators. Ayse Kalkan-Savoy, Nicholas Misiunas, J. Cory Minter, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, University of Massachusetts Lowell, One University Ave., CACT FA203, Lowell, MA 01854, ayse.k.savoy@gmail.com)

A method for evaluating the spatial impulse response for a planar source having arbitrary velocity and delay distribution is presented. The time-domain approach is based on the Rayleigh integral formulation for the velocity potential. To obtain the velocity potential one must evaluate and superpose the contribution from each elemental element of the radiator. This step entails the accumulation of the amplitude contribution from each elemental source arriving at the observation point at a prescribed time. The problem of sorting of these arrivals is examined in this work and procedures for mapping the algorithm for implementation on general-purpose graphics processing units are presented.

3:00

2pPA8. The weak sensitivity of acoustic resonance frequencies to temperature gradients. Keith A. Gillis, Michael R. Moldover (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov), James B. Mehl (Dept. of Phys. and Astronomy, Univ. of Delaware, Orkas, WA), and James W. Schmidt (Sensor Sci. Div., National Inst. of Standards and Technol., Gaithersburg, MD)

We determined the mass of argon gas contained within an un-thermostatted, commercially-manufactured 300 L pressure vessel (tank) with an uncertainty of 0.16 % at 0.6 MPa by combining measurements of the argon pressure, the frequencies of microwave and acoustic resonances within the tank, and an equation of state for argon. After correction for the

thermoacoustic boundary layer and for the tank's center-of-mass motion, the measured acoustic resonance frequencies f_{ac} determined the average speed of sound, and therefore the average temperature, of the argon in the tank. We show that, consistent with first-order perturbation theory, f_{ac} for the 3 lowest-frequency longitudinal gas modes gave the correct average temperature even when we imposed a 13 K temperature difference ΔT across the tank's diameter. However for the nearly degenerate doublet modes, we observed a linear dependence on ΔT for f_{ac} , which the theory does not predict. Using the thermal expansion deduced from the microwave measurements, we show that the linear dependence on ΔT was consistent with anisotropic changes in the tank's shape in response to the applied temperature gradient. We will discuss the predictions from perturbation theory of the changes in f_{ac} due to temperature gradients in cylindrical and spherical cavities. From these results, we argue that resonance frequencies can be used to "weigh" a compressed gas in much larger tanks in un-thermostatted environments and at high pressures.

3:15

2pPA9. Absorption and dispersion in Venus' lower and middle atmosphere. Mathbar S. Raut and Andi Petculescu (Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504, msr0706@louisiana.edu)

Vertical profiles for the sound speed and attenuation coefficient are calculated in the troposphere and mesosphere of Venus. The predictions are based on ambient quantities (pressure, temperature, density, and composition) obtained by measurements and global circulation models. The three major species considered for composition are carbon dioxide, nitrogen, and sulfur dioxide. The thermophysical parameters—specific heats, viscosity, and thermal conductivity—for the principal constituents were obtained by interpolation from the NIST Chemistry Webbook, at the pressure/temperature values of each altitude. A real-gas equation of state was used to account for the dense environment of the Venusian troposphere.

3:30

2pPA10. Investigation of acoustic scattering using fast-multipole Padé approximant methods. Hui Zhou, Elaheh Noursadeghi, Aydin Sadeqi, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, One University Ave., CACT FA203, Lowell, MA 01854, hui_zhou@student.uml.edu)

The scattering of acoustic waves from three-dimensional compressible fluid scatterers is considered. Particular attention is paid to cases where the

scatterers have moderate magnitude in compressibility contrast and nondimensional wave number. The scattered field is cast in terms of asymptotic series valid in the limit of small compressibility contrast. It has been shown that pointwise application Padé Approximants in the scattering volume may be used to expand the range of validity of the result. However pointwise divergence in the scattering volume can result as the magnitude of the compressibility contrasted is increased. In this work the utility of low-frequency Fast-multipole method in evaluating the Born series coefficients is examined.

3:45

2pPA11. Accurate fast field formulation for a uniformly moving source traveling above an impedance plane. Bao N. Tong and Kai Ming Li (School of Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2099, bntong@purdue.edu)

The Lorentz transform, which can be applied to a uniformly moving source, effectively converts the time-domain wave equation to an equivalent problem in frequency-domain due to a stationary source. Standard fast field program (FFP) implementations apply the property of conjugate symmetry in the wavenumber domain for improved efficiency. Recent literature [D. Dagna *et al.*, AIAA **52**, 1928–1939 (2014)] suggests the use of a Dopplerized frequency-dependent impedance model to account for the effects of source motion. Consequently, the FFP kernel function is no longer identical for the positive and negative wavenumbers. Additional complications are introduced by the necessity to compute the positive and negative horizontal separation distances in the Lorentz frame to obtain a complete time history of sound pressures for the source approaching and receding from the receiver. Further development of the FFP algorithm in the Lorentz frame is explored such that a frequency-dependent impedance model is developed. Both moving line and point monopole sources are considered in the current investigation. Results are validated against direct numerical integration schemes and with the published time-domain solutions. Applications for an aircraft operating in cruise condition is also examined in the present study. [Work Sponsored by the Federal Aviation Administration.]

Session 2pPP

Psychological and Physiological Acoustics, Biomedical Acoustics, Speech Communication, and Signal Processing in Acoustics: Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention II

Blake S. Wilson, Cochair
Duke University, 2410 Wrightwood Ave., Durham, NC 27705

Michael Dorman, Cochair
Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85258

Invited Papers

1:00

2pPP1. Lessons learned from multichannel cochlear implants that are relevant to a visual prosthesis. Jim Patrick (Cochlear Ltd., Macquarie Univ., 1 University Ave., New South Wales 2109, Australia, jpatrick@cochlear.com)

The clinical success of the multichannel cochlear implant was built on the successful outcomes of the multidisciplinary research program led by Professor Graeme Clark at the University of Melbourne and many of the outcomes from these studies can also be applied to other sensory prostheses, including the visual prosthesis. A primary requirement of any neural prosthesis is that it is safe, and this presentation will include studies that define stimulus parameters that allow stable long term responses, with no evidence of metabolic overload or neural degeneration. Any device that is chronically implanted for the life of a patient must be reliable, with a low likelihood of need for replacement because of device failure or infection, and the presentation will include both the results from Clark's infection studies and long term cochlear implant reliability observations. The presentation will describe the design of an experimental retinal prosthesis that uses an electrode array that is based on an array that was used to stimulate the cochlear nucleus of profoundly deaf patients. It will include preliminary results for volunteer patients, and approaches that may improve these pilot study outcomes.

1:20

2pPP2. Lessons learned from cochlear implants that are relevant to a vestibular prosthesis. Jay T. Rubinstein, Chris Phillips, Kai-bao Nie, Leo Ling, and James O. Phillips (Otolaryngol., Univ. of Washington, Box 357923, Seattle, WA 98195, rubinj@uw.edu)

In October, 2010, we began human studies of a vestibular prosthesis. Our goal is to develop a device for the management of several vestibular disorders that defy existing treatment modalities. We have long-term data on four human subjects and ten non-human primates and are awaiting regulatory approval of our second-generation device, modified based on that data. We have been guided by a wealth of information based on lessons learned from cochlear implants. These lessons include first and foremost the otherwise hubristic idea that an effective vestibular prosthesis is technically feasible. Indeed the extraordinary compensation mechanisms of the central vestibular system convince us that such a prosthesis could be even more effective than for hearing. Second, the ability to leverage CI engineering expertise and industry relationships has been critical to our success so far. Third, the regulatory comfort with the success and favorable risk profile of cochlear implants has greatly speeded up human translation. Fourth, surgical familiarity with these devices has greatly enabled our research and facilitates future multicenter studies. Lastly, experience with "soft-surgery", hearing preservation, and single-sided deafness cochlear implantation holds forth alluring possibilities for the range of vestibular disorders that may eventually succumb to this closely related technology.

1:40

2pPP3. Neuroplasticity with cochlear implants: Underlying neuronal mechanisms. Andrej Kral (Dept. of Experimental Otolology, Inst. of AudioNeuro Technol., Medical Univ. Hannover, Feodor-Lynen-Str. 35, Hannover 30625, Germany, a.kral@uke.de)

Chronic cochlear implant (CI) stimulation in congenitally deaf cats (CDCs) leads to cortical maturation matching imaging data from humans (Kral and Sharma, 2012, TINS). One possible measure of auditory plasticity is the reorganization of aural preference following monaural CIs (Kral *et al.*, 2013, Brain), showing a sensitive period of <4.2 months in cats. A substantial reduction of binaural information following developmental unilateral hearing was found in cortical neurons (Kral *et al.*, 2015, Audiol Neurootol). Consequently, auditory maturation requires experience. Additional to reduced synaptic plasticity, loss of acuity in feature representation, deficits in integrative function of the cortical column and deficits in corticocortical, particularly top-down, interactions, close the sensitive periods (Kral 2013, Neuroscience). Cross-modal plasticity recruits auditory resources for non-auditory tasks. Despite of cross-modal reorganization in some auditory areas the extend of the underlying reorganization of corticocortical connections (Barone *et al.*, 2013, PLoS One) indicates that this only moderately limits auditory processing capacity. Electrophysiological recordings in the dorsal auditory cortex of CDCs demonstrate that the cross-modally reorganized secondary auditory areas maintain a predominance of dormant auditory inputs additional to moderate cross-modal reorganization. [Supported by Deutsche Forschungsgemeinschaft (Cluster of Excellence Hearing4all).]

2:00

2pPP4. Longitudinal results in the childhood development after cochlear implantation study. John K. Niparko (Dept. of Otolaryngology-Head & Neck Surgery, Keck School of Medicine, Univ. of Southern California, Los Angeles, CA 90033, niparko@med.usc.edu)

This NIDCD-funded study has aimed to develop a longitudinal, multivariate model of spoken language outcomes after early cochlear implantation. Our national cohort accrued subjects in their infant and toddler stages. As the study enters its 12th year, the study offers a model of post-implant language development, now at a stage when participants are able to self-assess their interaction with peers, school performance, and quality of life. Here, we will present our primary outcome of interest-spoken language acquisition-discussing modifying variables related to auditory function, cognition, device, and environmental variables.

2:20

2pPP5. Importance of cochlear health for cochlear-implant function. Bryan E. Pflugst (Dept. of Otolaryngol., Univ. of Michigan, Kresge Hearing Res. Inst., 1150 West Medical Ctr. Dr., Ann Arbor, MI 48109-5616, bpfingst@umich.edu), Ning Zhou (Dept. of Commun. Sci. and Disord., East Carolina Univ., Greenville, NC), Deborah J. Colesa, Melissa M. Watts (Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI), Stefan B. Strahl (MED-EL GmbH, Innsbruck, Austria), Soha N. Garadat (Dept. of Hearing and Speech Sci., Univ. of Jordan, Amman, Jordan), Kara C. Schwartz-Leyzac, Yehoash Raphael, and Teresa A. Zwolan (Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI)

In humans with cochlear implants, functional measures show considerable variation from one stimulation site to another along the electrode array. Our research has demonstrated that (1) the across-site patterns of the functional data are stable over time but differ across subjects; and (2) the across-site patterns are measure specific. These observations are consistent with the hypotheses that implant performance at a given stimulation site is dependent on specific conditions near the site, and that the various functional measures do not all depend on the same conditions. However, we lack direct evidence as to the specific conditions leading to the site-specific differences in performance in humans. Studies in our guinea pig laboratory and elsewhere have demonstrated highly significant correlations between psychophysical or electrophysiological measures of implant function and anatomical measures of cochlear health. Furthermore, the correlated anatomical features differ across functional measures. Finally, some functional measures that are correlated with anatomical measures of cochlear health in animals are predictive of speech recognition ability in human implant users. The data support efforts to preserve and/or restore the health of the implanted cochlea. [This work was supported by NIH/NIDCD grants R01 DC010786, R01 DC010412, and P30 DC05188, and a contract from MED-EL.]

2:40–2:55 Break

2:55

2pPP6. Engineering contributions to the development of the cochlear implant. Fan-Gang Zeng (Otolaryngol. - Head and Neck Surgery, Univ. of California Irvine, 110 Med Sci E, Irvine, CA 92697, fzen@guci.edu)

The cochlear implant is a product of deliberate systems engineering, starting with a clear design goal by physicians to restore human hearing by means of electrical stimulation. The engineering development of the cochlear implant has been a dynamic interplay between a perfectionist's approach attempting to replicate the intricate sensing and processing in a normal biological system and a reductionist's approach building an artificial system with minimal components and complexity. The reductionist's approach won the initial battle as the first FDA-approved single-channel cochlear implant consisted of simply a microphone, an amplifier, and a pair of coils. Borrowing techniques and knowhow's from diverse engineering areas in acoustics, aerospace, and electronics, modern multi-channel cochlear implants have shifted towards the perfectionist's approach. Acoustic researchers, particularly speech scientists, have played an important role in the engineering development of the cochlear implant. From an engineering perspective, I will highlight development milestones, evaluate their contributions to actual implant performance, and delineate the complicated relationships between artificial and natural hearing.

3:15

2pPP7. Anatomical considerations for the design and applications of cochlear implants. Helge E. Rask-Andersen, Wei Liu (Dept. ORL, Uppsala Univ. Hospital, Uppsala, Uppsala 75241, Sweden, helge.raskandersen@gmail.com), Annelies Schrott-Fischer, and Rudolf Glueckert (Dept. ORL, Medical Univ. of Innsbruck, Innsbruck, Austria)

This presentation aims to display some important anatomical characteristics and variations of the human cochlea that may influence outcomes with cochlear implants. I will describe the complexity of the "hook" region and how it may challenge atraumatic insertion of an electrode array. In addition, I will discuss advantages and disadvantages of various cochleostomy approaches and of various trajectories for insertions. These anatomical considerations are informed by micro dissections of human temporal bones and plastic casts of different parts of the bones. In addition, the studied microanatomy of the basilar membrane may inform insertions and insertion depths needed to preserve cochlear structures. In particular, the studies suggest that the vulnerability of the membrane may be greatest at the apex and this possibility has implications for electrode designs and insertions. Also, the apical region may be most susceptible to inflammation and fibrosis due to its anatomy. The anatomical considerations for safe insertions of electrodes, including preservation of the remaining neural tissue, will be discussed. This work was supported by grants from Uppsala University Hospital and Uppsala University; the Tysta Skolan Foundation; the Swedish Deafness Foundation; the European Community 7th Framework Programme; and kind private donations from Börje Runögård.

2p TUE. PM

2pPP8. Tissue engineering and pharmaceutical interventions. Josef M. Miller, Yehoash Raphael, Keith R. Duncan, Richard A. Altschuler, and Gabriel Corfas (Kresge Hearing Res. Inst., Univ. of Michigan, Ann Arbor, MI 48109, josef@umich.edu)

Hearing performance with modern cochlear implants is dependent on remaining sensory and neural elements. CNS plasticity and individual capacity for outstanding performance with meager peripheral neural survival provide some exceptions; however, both human and animal studies support a direct relationship of performance with structural integrity/function of remaining sensory and neural elements in the implanted ear. This dependency will increase with next generation implants. This has driven interventions based on increased understanding of molecular mechanisms underlying cell death and survival to enhance sensory and neural survival. It has driven tissue engineering of remaining elements with neurotrophic and other factors to induce regeneration, neurite outgrowth, and stem cell replacements of lost elements. Delivery to the cochlea is being accomplished via middle ear or transtympanic application, direct delivery in bolus or continuous administration via osmotic pumps, as coatings or channels in implants, gene transfer via viral vectors, factors incorporated in nano- and micro-particles, and via systemic administration. Current results provide a base for optimism for a future integrated system of cochlear implant-drug/gene delivery system. Each agent and delivery has risks and benefits. Challenges to select the best strategy for translation and application in humans will be discussed. [This work was supported by NIH/NIDCD grants R01 DC010412, DC 004820, DC011294, P30 DC005188, and DOD W81XWH-12-10492.]

TUESDAY AFTERNOON, 19 MAY 2015

KINGS 3, 1:30 P.M. TO 2:45 P.M.

Session 2pSAa

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

Christina J. Naify, Cochair

Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Michael R. Haberman, Cochair

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

Contributed Papers

1:30

2pSAa1. A transformation-based formulation of an airfoil cloak using Joukowski mapping. Saliou Telly (Mech. Eng., Univ. of Maryland College Park, 14359 Long Channel Dr., Germantown, MD 20874, stelly@umd.edu) and Balakumar Balachandran (Mech. Eng., Univ. of Maryland College Park, College Park, MD)

Since its introduction in 2006, the transformation approach to cloaking has been extensively explored for electromagnetic and acoustic fields, due to its intuitive nature as well as the methodology. In this approach, the construction of invisibility cloaks consists of exploiting a change of coordinates to create a void region within a finite space while keeping the outer boundary of the considered space unchanged. Such changes in physical coordinates can be interpreted as a transformation of material properties related to electromagnetics and acoustics, given the invariance of the respective field equations to coordinate change. Based on the transformation, properties of an invisibility cloak can be inferred using available formulae. To this date, this approach has been successfully applied to various two-dimensional geometries including circular and square cylinders. In this work, as an extension, the authors present initial results obtained for the cloaking of a two-dimensional airfoil section. They take advantage of the mapping properties of the well-known Joukowski transformation, a complex mapping that is used in aerodynamics applications.

1:45

2pSAa2. An impedance-mobility model of stacked membrane-type acoustic metamaterials. Matthew G. Blevins (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 909 S 70th plz #4, Omaha, NE 68106, mblevins@huskers.unl.edu), Siu-Kit Lau (Armstrong (China) Investment Co. Ltd., Shanghai, China), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

Membrane-type acoustic metamaterials have been proven to exhibit high low-frequency transmission loss despite their small thickness and light weight. To date, analysis has focused primarily on experimental studies in plane-wave tubes and numerical modeling using finite element methods. These methods are inefficient when used for applications that require iterative changes to the structure of the material. In addition, high sound transmission loss with a single layer of such metamaterial only occurs in a narrow frequency range. To facilitate design and optimization of stacked membrane-type acoustic metamaterials, a computationally efficient dynamic model based on the impedance-mobility approach is proposed. Results are verified against a finite element model. Single and double layer transmission loss characteristics are compared. Wide-band high-transmission-loss acoustic metamaterials can be achieved by double layer membranes and using the proposed approach for optimization. The impedance-mobility approach is shown to be very efficient for modeling and optimization of such materials, compared against the conventional finite element approach.

2:00

2pSAa3. Acoustoelectric admittance of two-dimensional ferroelectric metamaterial under dispersion curve branching. Ola H. Nusierat (Dept. of Phys. and Astronomy, Univ. of MS, 108 Lewis Hall, P.O. Box 1848, University, MS 38677, ohnusierat@go.olemiss.edu), Lucien Cremaldi, and Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, Oxford, MS)

The acoustoelectric admittance Y of the ZX-cut periodically poled LiNbO₃ plate is investigated experimentally and theoretically computed. The sample consists of 88 inversely poled domains of 0.45 mm-long each. The vector voltmeter, digital oscilloscope, and function generator were used to measure the frequency dependencies $Y(F)$ at room temperature. Double peaks were observed in the admittance measurements near the lower edge of a stop-band at frequencies 3.268 MHz and 3.287 MHz. Another double peak exists near upper edge of the acoustic stop band at frequencies 3.651 MHz and 3.663 MHz. The double peak in Y can be explained as follows. Ultrasound in this ferroelectric acoustic metamaterial (FAM) has a so-called stop band when an acoustic wavelength is close to a double-length of ferroelectric domain within the inversely poled structure. The dispersion curves computed by the finite element method reveal an effect of decoupling of two acoustic displacements in a zero-antisymmetric mode. The two displacements A_x and A_z along the X and Z axes become decoupled near the boundaries of the acoustic Brillouin zone. This can be explained by different diffraction losses in two orthogonal displacements within the FAM. Computations are in a good agreement with experiments.

2:15

2pSAa4. Acoustic Poisson-like effect. Alexey Titovich and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey.titovich@rutgers.edu)

A sonic crystal with a particular local resonance is capable of transferring incident acoustic energy to a perpendicular direction. This is done with

a square array of cylindrical scatterers vibrating with a non-axisymmetric mode which couples to the anti-symmetric mode (deaf mode) of the crystal via an evanescent band unlocking it from the bandgap boundaries and resulting in perpendicular propagation. The scatterer is an elastic shell designed to be acoustically transparent with impedance and index matched to water, which is vibrating with the $n=2$ in-plane bending mode. The shells' collective motion resembles the quasi-static Poisson effect in elastic solids and results in a quadrupole scattering pattern which causes a purely perpendicular mode to be excited at normal incidence. Thus, the acoustic Poisson-like effect is non-refractive and also highly efficient. Variations of this novel effect will also be discussed such as different local resonances, lattices and crystals. [Support from ONR is gratefully acknowledged.]

2:30

2pSAa5. Acoustic ground cloaks revisited. Peter Kerrian (Graduate Program in Acoust., The Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, pak215@psu.edu), Amanda Hanford, Dean Capone, and Scott Miller (Appl. Res. Lab Penn State, University Park, PA)

The unique material properties now obtainable with acoustic metamaterials have led to unprecedented control of acoustic wave propagation, resulting in many applications including acoustic cloaking. The two fundamental approaches in the development of a ground cloak are quasiconformal mapping [Li *et al.* Phys. Rev. Lett. **101**, 203901 (2008)] and coordinate transformations [Popa *et al.* Phys. Rev. B. **83**, 224304 (2011)]. The differences in the required material properties prescribed by these two approaches lie in the amount of anisotropy and inhomogeneity, as well as the size of the cloak relative to object. The coordinate transformation approach has been used to produce a realizable anisotropic homogeneous ground cloak in the acoustic domain. This presentation will highlight the findings of work that examined how advances in metamaterial development could lead to the realization of required material properties for ground cloaks, and explore alternative transformations to expand the applications for acoustic ground cloaks.

2p TUE. PM

TUESDAY AFTERNOON, 19 MAY 2015

COMMONWEALTH 2, 3:30 P.M. TO 4:50 P.M.

Session 2pSAb

Structural Acoustics and Vibration and Signal Processing in Acoustics: Time-Domain Methods

Micah R. Shepherd, Cochair

Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801

John B. Fahline, Cochair

ARL / Penn State, P.O. Box 30, State College, PA 16804-0030

Invited Papers

3:30

2pSAb1. Inverse problems in transient structural acoustics. Timothy F. Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov) and Wilkins Aquino (Civil and Environ. Eng., Duke Univ., Durham, NC)

Inverse problems are frequently encountered in transient structural acoustics. Given measured accelerometer or microphone time histories, one may need to estimate material properties in a structure, impedance conditions on the boundaries, shape or topology of an object, or characterize acoustic and structural sources that produced the measured data. Applications include model calibration, acoustic testing of aerospace structures, military surveillance, and underwater acoustics. Typically, accelerometer or microphone pressures are

measured experimentally, and it is desired to characterize the material parameters, boundary conditions, topology, or acoustic sources that produced these accelerations or microphone pressures. In this talk, we will present a set of example inverse problems in structural acoustics, and a framework for their solution using an operator-based partial differential equation (PDE) constrained optimization approach. This abstract framework enables one to reduce any inverse problem to a set of fundamental operations and enables re-use of several software constructs. Formulations will be presented in both the time and frequency domain, and the merits and drawbacks of both approaches will be discussed. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed Martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL85000.]

3:50

2pSAb2. Simulation of coupled structural-acoustic response with dynamic damage evolution. Jonathan S. Pitt (Appl. Res. Lab., The Penn State Univ., Appl. Res. Lab, PO Box 30, Mailstop 3320B, State College, PA 16804, jonathan.pitt@psu.edu)

A novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system is presented. The system is derived via the theory of continuum damage mechanics, and incorporates standard damage evolution models. The overall solution method is staggered, solving for the dynamic damage evolution first with an explicit step, and then using the new values in the coupled computation of the structural-acoustic system. The spatial domain is discretized using a mixed finite element method, and the temporal space is discretized with a higher-order implicit time discretization scheme. Efforts toward fully coupled verification of the solution algorithm are presented, as are validation studies for cases without evolving damage. Applications with evolving damage are presented, and present a first principles study of changes in the structural acoustic response to dynamically evolving damage in the structure. Examples of downstream usage of the evolving structural response are discussed in the concluding remarks.

4:10

2pSAb3. Time windowed comparisons between models and measurements. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu)

The present work proposes and demonstrates a method for improving the accuracy of large finite element models using measurements. Finite element analysis routinely allows for the construction of dynamic models with millions of degrees of freedom. The models often involve a large number of parameters, such as material properties and dimensions, that are not precisely known. For example, one may measure the material properties of a homogeneous sample but those properties will vary from sample to sample. Moreover, material properties may change considerably during construction as the material is prestressed, bonded, or welded. Therefore, it is often the case that the finite element model does not agree with measurements and one is faced with the task of modifying model parameters to yield agreement. This is made difficult by the large number of parameters as well as the computational cost of evaluating the model for each set of parameter choices. The present work proposes a time windowing method that localizes the spatial volume of response in the model and experiment and therefore isolates a small number of model parameters that may be varied to bring agreement. Numerical examples are presented to illustrate the method.

4:30

2pSAb4. A coupled finite element/equivalent source formulation for transient structural-acoustic problems. John B. Fahline (ARL / Penn State, P.O. Box 30, State College, PA 16804-0030, jbf103@arl.psu.edu)

Recent research suggests that instability problems that occur in the time domain boundary element formulations are manifestations of nonuniqueness difficulties, and that remedies similar to those used in frequency domain formulations can address the difficulties and improve solution stability. For frequency domain equivalent source formulations, the difficulties can be addressed by combining simple and dipole sources together to form "tripole sources". The basic goals of the current research are to develop an analogous equivalent source formulation for transient acoustic boundary value problems and to combine it with structural finite element analyses to solve transient coupled structural-acoustic problems. A brief description of the equivalent source formulation is given along with several validation cases for acoustic boundary value problems, including one with a closed boundary surface where nonexistence difficulties should occur. The formulation requires convolution summations, which are independent of each other and a brief discussion is given of how the computations can be parallelized. A brief description is also given of the coupled finite element/equivalent source formulation for transient structural-acoustic boundary value problems. Several examples are given to validate the formulation and to demonstrate that the tripole source formulation for the equivalent sources is more stable than the simple source version.

Session 2pSC

Speech Communication: Speech Methods, Models and Technology (Poster Session)

Robert Hagiwara, Chair

Department of Linguistics, University of Manitoba, Winnipeg, MB R3T 5V5, Canada

Posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 1:15 p.m. to 2:45 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m. There will be a 15-minute break from 2:45 p.m. to 3:00 p.m.

Contributed Papers

2pSC1. Channel and noise robustness of articulatory features in a deep neural net based speech recognition system. Vikramjit Mitra (Speech Technol. and Res. Lab, SRI Int., 333 Ravenswood Ave., EJ133, Menlo Park, CA 94025, vmitra@speech.sri.com), Ganesh Sivaraman (Univ. of Maryland, College Park, MD), Hosung Nam (Haskins Labs, New Haven, CT), Carol Y. Espy-Wilson (Univ. of Maryland, College Park, MD), and Elliot Saltzman (Boston Univ., Boston, MA)

Articulatory features (AFs) are known to provide an invariant representation of speech, which is expected to be robust against channel and noise degradations. This work presents a deep neural network (DNN)—hidden Markov model (HMM) based acoustic model where articulatory features are used in addition to mel-frequency cepstral coefficients (MFCC) for the Aurora-4 speech recognition task. AFs were generated using a DNN trained layer-by-layer using synthetic speech data. Comparison between baseline mel-filterbank energy (MFB) features, MFCCs and fusion of articulatory feature with MFCCs show that articulatory features helped to increase the noise and channel robustness of the DNN-HMM acoustic model, indicating that articulatory representation does provide an invariant representation of speech.

2pSC2. A method for estimating lingual cavity volume in click consonants. Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, miller.5592@osu.edu)

A method for estimating lingual cavity volume in click consonants is presented. The upper edge of the lingual cavity is estimated from stone palate casts, which were scanned using a Polhemus 3D Digitizer. The floor of the midline of the lingual cavity was estimated using 114 fps mid-sagittal ultrasound traces of the tongue collected with the CHAUSA method (Miller and Finch 2011). The width of the cavity from front to back was estimated by measurements of the cavity width at six different locations in linguograms. Changes in lingual cavity volume for the four coronal click types in Mangetti Dune !Xung are estimated by changing the mid-sagittal ultrasound trace that corresponds to one of five stages in click production. 3-D images of the changes in cavity volume for each of the four coronal click types recognized by the IPA (2006) are presented.

2pSC3. Quantal biomechanics in an embodied phonetics. Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, British Columbia V6T1Z4, Canada, gick@mail.ubc.ca) and Scott R. Moisiak (The Max Planck Inst. for PsychoLinguist, Nijmegen, Netherlands)

Quantal regions were described by Stevens [e.g., 1989, *J. Phon.* **17**, 3–45] to identify nonlinear stabilities in the relationship between articulation

and acoustics. Classic cases of quantal effects show how tongue posture may vary within one region of the vocal tract with little acoustic change, while in other regions very small movements can have large effects on acoustic output. Such effects can be thought of as attractors to speech behavior in those regions of the phonetic space that allow greater noise. Quantal-like stabilities have been suggested to operate not just in articulatory-acoustic space, but in biomechanical-articulatory space as well [e.g., Schwartz *et al.*, 1997, *J. Phon.* **25**, 255–286]. It is argued here that such quantal-like stabilities are a hallmark of speech modules [Gick & Stavness, 2013, *Front. Psych.* **4**, 977], providing the basis for robust, feed-forward control. Computer simulations in the ArtiSynth platform (www.artisynth.org) are used to demonstrate quantal effects in speech biomechanics at multiple vocal tract loci, including the lips, oropharyngeal isthmus, and larynx. Moving into the future, quantal work will integrate observations about nonlinear stabilities cutting across the many physical and sensory domains that figure in speech. [Research funded by NSERC.]

2pSC4. Measurement of child speech complexity using acoustic landmark detection. Marisha Speights, Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267, speighma@mail.uc.edu), Joel MacAuslan (Speech Technol. and Appl. Res., Boston, MA), and Harriet Fell (College of Comput. Information Sci., Northeastern Univ., Boston, MA)

An important measure of intelligibility in children is the ability to articulate complex syllables. At the same time, the transcription and analysis of young children's speech data is laborious and time-intensive, requiring specialized training and expert perceptual abilities on the part of researchers. Researchers have called for automatized methods to quantify these measures of articulatory complexity. One such method, Automatic Syllabic Cluster Analysis, may allow for more efficient analysis by using acoustic landmarks [Stevens *et al.*, 1992] to systematically detect abrupt and maximal events in the speech signal, group them into syllable patterns and further group the syllable patterns into utterances. Statistics derived from these groupings are used to determine the complexity of utterances. Automatic Syllabic Cluster Analysis does not require transcription for analysis of the speech because it is not lexically driven, instead employing measurements of acoustic parameters that characterize changes in articulatory precision. To identify a potential application for this automated approach, this pilot research examines syllabic complexity in children to differentiate between children who are typically developing and those with diagnosed speech disorders. Preliminary results indicate that Automatic Syllabic Cluster Analysis can identify articulatory differences between the groups.

2pSC5. Augmenting acoustic phonetics with articulatory features for phone recognition. Ganesh Sivaraman (Elec. & Comput. Eng., Univ. of Maryland College Park, 5002 Seminole St., Berwyn Heights, MD 20740, ganesa90@umd.edu), Vikramjit Mitra (SRI Int., Menlo Park, CA), Hosung Nam (Korea Univ., Seoul, South Korea), Elliot Saltzman (Dept. of Physical Therapy and Athletic Training, Boston Univ., Boston, MA), and Carol Espy-Wilson (Elec. & Comput. Eng., Univ. of Maryland College Park, College Park, MD)

In articulatory phonetics, a phoneme's identity is specified by its articulator-free (manner) and articulator-bound (place) features. Previous studies have shown that acoustic-phonetic features (APs) can be used to segment speech into broad classes determined by the manner of articulation of speech sounds; compared to MFCCs, however, APs perform poorly in determining place of articulation. This study explores the combination of APs with vocal Tract constriction Variables (TVs) to distinguish phonemes according to their place of articulation for stops, fricatives and nasals. TVs were estimated from acoustics using speech inversion systems trained on the XRMB database with pellet trajectories converted into TVs. TIMIT corpus sentences were first segmented into broad classes using a landmark based broad class segmentation algorithm. Each stop, fricative and nasal speech segment was further classified according to its place of articulation: stops were classified as bilabial (/P/, /B/), alveolar (/T/, /D/) or velar (/K/, /G/); fricatives were classified as labiodental (/F/, /V/), alveolar (/TH/, /DH/, /S/, /Z/), palatal (/SH/, /ZH/) or glottal (/HH/); and nasals were classified as bilabial (/M/), alveolar (/N/) or velar (/NG/). Polynomial kernel support vector machines were trained on APs concatenated with vocal tract constriction features. Results showed that combining acoustic and articulatory features leads to reliable recognition of manner and place of articulation, and improves phone recognition.

2pSC6. Non-distorting method of vowel normalization by transposition in the semitone scale. Danila Gomulkin (Povarskoy per. 3, kv. 26, Saint Petersburg 191025, Russian Federation, gomulkin@yahoo.co.uk)

Converting formant values from Hertz into semitones provides a remarkable "area agreement" between vowel systems in the F1/F2 plane without distorting intervals between the formants. Further simple transposition of a Sample system in relation to a Reference system by the difference between the mean of all converted F₁s and F₂s for the Sample speaker/group and the mean of those for the Reference speaker/group efficiently eliminates physiological mismatch between the speakers/groups and allows for convenient comparison of the vowel systems in a meaningful scale of pitch intervals. To convert formant *n* of vowel *V* from Hertz to semitones (expressed in MIDI keyboard numbers) use formula: [1] $F_{n[V]midi} = \text{midi}(F_{n[V]Hz})$, where [2] $\text{midi}(x) = 12 * \log_2(x/440) + 69$. To normalize converted formants of a Sample speaker/group against those of a Reference speaker/group, deduct from the converted Sample formants ($F_{n[V]midi}(\text{Sample})$) the difference between the mean of all converted F₁s and F₂s for the Sample speaker/group and the mean of those for the Reference speaker/group ($\Delta \text{MEAN}_{\text{midi}(\text{Sample-Ref})}$): [3] $F_{n[V]midi}(\text{Sample}) = F_{n[V]midi}(\text{Sample}) - \Delta \text{MEAN}_{\text{midi}(\text{Sample-Ref})}$

2pSC7. Realtime voice activity and pitch modulation for laryngectomy transducers using head and facial gestures. Gautam Mohan, Katherine Hamilton, Andrew Grasberger, Adam C. Lammert, and Jason Waterman (Comput. Sci. Dept., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, lammert@cs.swarthmore.edu)

Individuals who have undergone laryngectomy often rely on handheld transducers (i.e., the electrolarynx) to excite the vocal tract and produce speech. Widely used electrolarynx designs are limited, in that they require manual control of voice activity and pitch modulation. It would be advantageous to have an interface that requires less training, perhaps using the remaining, intact speech production system as a scaffold. Strong evidence exists that aspects of head motion and facial gestures are highly correlated with gestures of voicing and pitch. Therefore, the goal of project MANATEE is to develop an electrolarynx control interface which takes advantage of those correlations. The focus of the current study is to determine the feasibility of using head and facial features to accurately and efficiently modulate the pitch of speaker's electrolarynx in real time on a mobile platform using the built-in video camera. A prototype interface, capable of running

on desktop machines and compatible Android devices, is implemented using OpenCV for video feature extraction and statistical prediction of the electrolarynx control signal. Initial performance evaluation is promising, showing pitch prediction accuracies at double the chance-level baseline, and prediction delays well below the perceptually-relevant, ~50 ms threshold.

2pSC8. Dual electromagnetic articulometer observation of head movements coordinated with articulatory gestures for interacting talkers in synchronized speech tasks. Mark Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, tiede@haskins.yale.edu) and Dolly Goldenberg (Linguist, Yale Univ., New Haven, CT)

Previous research has demonstrated that speakers readily entrain to one another in synchronized speech tasks (e.g., Cummins 2002, Vatikiotis-Bateson *et al.* 2014, Natif *et al.* 2014), but the mixture of auditory and visual cues they use to achieve such alignment remains unclear. In this work, we extend the dual-EMA paradigm of Tiede *et al.* (2012) to observe the speech and coordinated head movements of speaker pairs interacting face-to-face during synchronized production in three experimental tasks: the "Grandfather" passage, repetition of short rhythmically consistent sentences, and competing alternating word pairs (e.g., "topper-cop" vs. "copper-top"). The first task was read with no eye contact, the second was read and then produced with eye contact, and the third required continuous eye contact. Head movement was characterized using the tracked position of the upper incisor reference sensor. Prosodic prominence was identified using F0 and amplitude contours from the acoustics, and gestural stiffness on articulator trajectories. Preliminary results show that frequency and amplitude of synchronized head movement increased with task eye contact, and that this was coordinated systematically with both acoustic and articulatory prosodic prominence. [Work supported by NIH.]

2pSC9. Benefits of using polar coordinates for working with ultrasound midsagittal tongue contours. Matthias Heyne (Linguist, Univ. of Canterbury, Private Bag 4800, Christchurch, Canterbury 8140, New Zealand, matthias.heyne@pg.canterbury.ac.nz) and Donald Derrick (New Zealand Inst. of Lang. Brain and Behaviour, Univ. of Canterbury, Christchurch, New Zealand)

Calculating SSANOVA average curves of tongue contours is a technique widely used in the field of speech research to facilitate the comparison of articulatory data (Davidson, 2006). Even though this technique is visually easier to understand and offers better statistical tests than using a concentric grid, Mielke (JASA, in press) has recently shown problems arising from generating SSANOVA curves in the Cartesian plane and has instead suggested the use of polar coordinates. Transferring Cartesian coordinates to the polar plane requires choosing an origin for the polar coordinate system and we hereby propose an alternative way of doing so by estimating the ultrasound transducer position through plotting two lines on an ultrasound image. The resulting average curves are very similar to Mielke's results but facilitate further analysis of data due to being relative to the estimated transducer position. Such possibilities include cutting off individual tokens at the edges to avoid wiggly ends of average contours, rotating (by adding/subtracting to/from the theta coordinate) and shifting data (by changing the x- and y-coordinates of the estimated origin in the Cartesian plane) to improve the (visual) comparability of ultrasound data across subjects. Some of these techniques were used in Heyne and Derrick (SST, 2014).

2pSC10. On the use of respiratory masks for speech measurements. Hao Zhang, Xiao Chen, and Stephen Zahorian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, Binghamton University, Binghamton, NY 13902, hzhang20@binghamton.edu)

Ideally, respiratory masks, when used to make certain measurements of speech and singing, should not interfere with what is being measured. Unfortunately this is not always the case. In this paper, two masks intended for speech measurements, are experimentally compared. One is a hard-walled mask manufactured and marketed by Kay-Pentax. The other is the circumferentially-vented pneumotachograph (CV) mask manufactured and marketed by Glottal (Syracuse, NY). Distortion measures, as quantified by formant frequencies, and muffling measures, as quantified by estimated

mask transfer functions, indicate that the hard-walled mask interferes with the speech characteristics much more than does the CV mask. It is hypothesized that SPL measurements are also likely to be inaccurate if taken when using the hard-walled mask.

2pSC11. Audio dilation in real time speech communication. John S. Novak (Dept. of Comput. Sci., Eng. Res. Facility (ERF), Univ. of Illinois at Chicago, 842 W Taylor, Rm. 2032, M/C 152, Chicago, IL 60607, john.novak@gmail.com), Jason Archer (Dept. of Commun., Univ. of Illinois at Chicago, Chicago, IL), Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ., Chicago, IL), and Robert V. Kenyon (Dept. of Comput. Sci., Univ. of Illinois at Chicago, Chicago, IL)

Algorithmically decreasing speech tempo, or “audio dilation,” can improve speech perception in attentionally demanding tasks [Gygi & Shafiro (2014), *Hear. Res.* **310**, pp. 76–86]. On-line audio dilation [Novak, *et al.* (2013), Interspeech 1869–1871] is a recently developed technique that decreases the tempo of audio signals as they are generated. This pilot study investigated effects of on-line audio dilation on performance in interactive problem solving tasks. We used a Diapix task [Baker & Hazan (2011), *Behav. Res. Methods* **43**(3), 761–770] to elicit and record spontaneous speech from pairs of participants under various dilation conditions: participants, seated in different rooms, were asked to find ten differences on two similar pictures, while their speech was either transmitted as spoken or diluted. Conditions tested include stretching one, both, or neither audio signal by 40%. Subsequent analysis shows that the technique, even using this substantial increase, did not interfere with interactive problem solving tasks, and did not lead to changes in speech production rate, measured as number of syllables per second. The lack of negative effects of on-line speech dilation provides a preliminary basis for further assessment of this method in speech perception tasks with high attentional and memory load.

2pSC12. Human-system turn taking analysis for the let’s go bus information system. Tiancheng Zhao and Maxine Eskenazi (Lang. Technol. Inst., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, tianchez@andrew.cmu.edu)

We examined turn-taking in the Let’s Go Bus spoken dialog data from two aspects: study the consequences of system barge-in when users are not finished speaking (false system barge-in, FSB); determine whether using a partial recognition result other than the final one produces better results. The consequence of FSBs is a less user-friendly system coupled with poor recognition caused by barge-ins, which divide one user turn into several fragments (UFs). We observed that UFs result in longer dialogs because the dialog manager has to recover from misrecognized utterances. Dialogs with UFs have 34 turns on average those without have 27. Poor recognition and long dialogs together cause lower task success rate. Dialogs with UFs have a success rate of 62% versus 84% for dialogs without. Moreover, we annotated the number of correct and incorrect slots for all partial recognitions. For 51% of the utterances, there exists a partial that contains more correct slots than the final recognition result. This will lead us to develop an algorithm to find the best partial. We conclude that systems that avoid FSB will have more efficient dialogs. They will also have better recognition by using the best partial instead of only the final one.

2pSC13. Robust automatic speech recognition in reverberation: onset enhancement versus binaural source separation. Hyung-Min Park (Dept. of Electron. Eng., Sogang Univ., Seoul, South Korea), Matthew Maciejewski (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Chanwoo Kim (Google, Mountain View, CA), and Richard M. Stern (Dept. of Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA, rms@cs.cmu.edu)

The precedence effect describes the auditory system’s ability to suppress later-arriving components of sound in a reverberant environment, maintaining the perceived arrival azimuth of a sound in the direction of the actual

source, even though the later reverberant components may arrive from other directions. It is also widely believed that precedence-like processing can also improve speech intelligibility for humans and the accuracy of speech recognition systems in reverberant environments. While the mechanisms underlying the precedence effect have traditionally been assumed to be binaural in nature, it is also possible that the suppression of later-arriving components may take place monaurally, and that the suppression of the corresponding components of the spatial image may be a consequence of this more peripheral processing. This paper compares potential contributions of onset enhancement (and consequent steady-state suppression) of the envelopes of subband components of speech at the monaural and binaural levels. Experimental results indicate that substantial improvement in recognition accuracy can be obtained in reverberant environments if feature extraction includes both onset enhancement and binaural interaction. Recognition accuracy appears to be relatively unaffected by which stage in the binaural processing is the site of the suppression mechanism. [Work supported by the LG Yonam Foundation and Cisco.]

2pSC14. Improving the accuracy of speech emotion recognition using acoustic landmarks and Teager energy operator features. Reza Asadi (College of Comput. and Information Sci., Northeastern Univ., 93 Chester St., Apt. 4, Allston, MA 02134, asadi@ccs.neu.edu) and Harriet Fell (College of Comput. and Information Sci., Northeastern Univ., Boston, MA)

Affective computing can help us achieve more intelligent user interfaces by adding the ability to recognize users’ emotions. Human speech contains information about the emotional state of the speaker and can be used in emotion recognition systems. In this paper, we present a machine learning approach using acoustic features which improves the accuracy of speech emotion recognition. We used 698 speech samples from “Emotional Prosody Speech and transcripts” corpus to train and test the classifiers. The emotions used were happy, sadness, hot anger, panic, and neutral. Mel-frequency Cepstral Coefficients (MFCC), Teager Energy Operator (TEO) features, and acoustic landmark features were extracted from speech samples. Models were trained using multinomial logistic regression, k-Nearest Neighbors(k-NN) and Support Vector Machine(SVM) classifiers. The results show that adding landmark and TEO features to MFCC features improves the accuracy of classification. SVM classifiers with a Gaussian kernel had the best performance with an average accuracy of 90.43%. We achieved significant improvement in the accuracy of the classification compared to a previous study using the same dataset.

2pSC15. Using the speech recognition virtual kitchen infrastructure for reproducible cross-disciplinary speech research exchange. Andrew R. Plummer (Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, plummer.321@osu.edu)

Computational models and methods of analysis have become a mainstay in speech research over the last seventy years, but the means for sharing software systems is often left to personal communication between model developers. As a result, the sharing of systems is typically complicated, error-prone, or simply not done at all, making it difficult (in some cases impossible) to verify model performance, engage with developers directly using their own models, or bridge gaps between research communities that have diverged over time. Moreover, the learning curve for new students or tech consumers entering these communities is quite steep, limiting the use of the models to those initiated in a given area. Over the last few years a number of computing infrastructures have taken shape that aim to address the difficulties encountered in the exchange of large software systems (e.g., the Berkeley Computational Environment, <http://collaboratool.berkeley.edu/>). We present the Speech Recognition Virtual Kitchen (www.speechkitchen.org)—a computing infrastructure that provides tools and communication channels for the exchange of speech-related software systems in a manner that facilitates the sharing of reproducible research models and pedagogical material—together with several demonstrations of the infrastructure and its potential uses.

2pSC16. Documenting sound change with smartphone apps. Adrian Leemann (Dept. of Theor. and Appl. Linguist, Univ. of Cambridge, Sidgwick Ave., Cambridge, Cambridgeshire CB3 9DA, United Kingdom, a1764@cam.ac.uk), Marie-José Kolly (LIMSI, Université de Paris-Sud, Paris, France), David Britain (Dept. of English Lang. and Linguist, Univ. of Bern, Bern, Switzerland), Ross Purves (Dept. of Geography, Univ. of Zurich, Zurich, Switzerland), and Elvira Glaser (Dept. of German, Univ. of Zurich, Zurich, Switzerland)

Crowdsourcing linguistic phenomena with smartphone applications is relatively new. Apps have been used to train acoustic models for automatic speech recognition (de Vries *et al.* 2014) and to archive endangered languages (Iwaidja Inyaman Team 2012). Leemann and Kolly (2013) developed a free app for iOS—*Dialläkt Äpp* (DÄ) (>78k downloads)—to document language change in Swiss German. Here, we present results of sound change based on DÄ data. DÄ predicts the users' dialects: for 16 variables, users select their dialectal variant. DÄ then tells users which dialect they speak. Underlying this prediction are maps from the *Linguistic Atlas of German-speaking Switzerland* (SDS, 1962-2003), which documents the linguistic situation around 1950. If predicted wrongly, users indicate their actual dialect. With this information, the 16 variables can be assessed for language change. Results revealed robustness of phonetic variables; lexical and morphological variables were more prone to change. Phonetic variables like *to lift* (variants: /lupfə, lypfə, lipfə/) revealed SDS agreement scores of nearly 85%, i.e., little sound change. Not all phonetic variables are equally robust: *ladde* (variants: /xælə, xællə, xæuə, xæfə, xæfə/) exhibited significant sound change. We will illustrate the results using maps that show details of the sound changes at hand.

2pSC17. The recorder's paradox: Balancing high-quality recordings with spontaneous speech in noisy recording environments. Paul De Decker (Linguist, Memorial Univ. of NF, Sci. Bldg. 3050B, St. John's, Newfoundland A1B 3X9, Canada, pauldd@mun.ca)

While sociophonetic analysis requires high-quality sound recordings, sociolinguistic interviews (Labov 1984) are often conducted in uncontrolled, natural environments to elicit casual speech (Tagliamonte 2006). The effects of room acoustics and background noise on formant measurements, however, have never been systematically examined. To empirically investigate how ambient noise affects recording quality and measurements, a male speaker of English was simultaneously recorded to multiple devices reading 260 carrier phrases. Three naturally occurring background noise conditions (+20dB, +10dB, and +0dB SNR) were created in Praat 5.4 (Boersma and Weenink 2014) and mixed with the original audio recordings. 10,028 measurements of F1 and F2 were taken at the temporal midpoint of each vowel in each of the four recordings using LPC analysis in Praat. Pearson's *r* tests in R (R Core Team, 2014) assessed the correlation between measurements from each recording. Main results reveal positive correlations between "noiseless" and +20dB, and +10dB SNR conditions for each device, all vowels and both formants. When the signal was not appreciably louder than the background noise (i.e. +0 dB SNR) correlations significantly weakened. These findings are discussed as they relate to sociolinguistic investigations that need to balance high-fidelity recordings with noisier speaking environments.

2pSC18. A coder with adaptive filters and non-uniform techniques for simple speech communication. Seonggeon Bae, Myungsook Kim, and Myungjin Bae (Soongsil Univ., 1118-408, Mokdong Apt. 11dangi, Yangcheon Gu, Seoul 148-771, South Korea, sgbae123@gmail.com)

This paper proposes a new method of non-uniform sampling coding that utilizes a modified zero-crossing rate filter and a variable filter. In speech signals, the modified zero-crossing rate refers to the number of times in which the signal crosses zero in a single frame; the rate is high in noisy sections but low in sections of vocal sounds that are quasi-periodic. And the modified zero-crossing rates of unvoiced signals are similar to those of voiced signals. In speech signals, the non-uniform sampling coding method utilizes the peaks and valleys in a frame. In noisy sections, there are a lot of zero crossings and thus creating many peaks and valleys, while sections with voice sounds have zero-crossings numbering only 2–3 times of their fundamental frequency. The method proposed in this paper utilizes the

number of peaks and valleys in determining the validity of a sample; the less we have peaks and valleys, the higher compression rates we get. Therefore, by decreasing the number of samples from sections with noise and voiceless sounds which have a high number of peaks and valleys, and by creating a signal that has little effect on the recognized signals, we may acquire a better compression rate. As examined and proposed in this paper, these features can be used to design a filter coefficient for a better performance in order to improve compression rates of non-uniform sampling coding and to maintain high quality signals.

2pSC19. Vowel nasalization might affect the envelop of the vowel signal by reducing the magnitude of the rising and falling slope amplitude. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu), Mohammad Mehdi Alemi (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), and Mohammad Eshghi (Inst. of TeleCommun. Systems, Technische Univ. Berlin, Berlin, Germany)

Electrical analog study of the vocal tract has long time ago shown that nasalization can cause a drop of 5–10 dB in the overall vowel amplitude (House and Stevens, 1956). In this preliminary study, the magnitude of rising and falling slope amplitudes from the vowel signal was introduced as a new index to discriminate oral vowels from nasalized vowels. Two speech samples of /iti/ and /iki/ were produced by two normal children and two age-matched children with cleft lip and palate and hypernasal speech. Speech samples were digitally recorded using a microphone and CSL software. The PRAAT software was used to extract the text file of the vowel segment (i.e., from the initiation of the vowel periodicity to the initiation of the constriction interval of the following plosive). Then, the text files were analyzed by MATLAB to measure amplitude rising and falling slopes of the vowel signal. Results indicated that mean falling and rising slopes of the amplitude in the nasalized vowel are smaller than those of the oral vowel. This technique might be considered as an objective index to measure vowel quality in disordered speech studies as well as speech synthesis.

2pSC20. Smiled speech in a context-invariant model of coarticulation. Samuel Akinbo, Thomas J. Heins, Megan Keough, Elise K. McClay, Avery Ozburn, Michael D. Schwan, Murray Schellenberg, Jonathan de Vries, and Bryan Gick (Univ. of Br. Columbia, 2613 West Mall, Vancouver, British Columbia V6T 1Z4, Canada, mkeough@alumni.ubc.ca)

Smiling during speech requires concurrent and often conflicting demands on the articulators. Thus, speaking while smiling may be modeled as a type of coarticulation. This study explores whether a context-invariant or a context-sensitive model of coarticulation better accounts for the variation seen in smiled versus neutral speech. While context-sensitive models assume some mechanism for planning of coarticulatory interactions [see Munhall *et al.*, 2000, *Lab Phon.* V, 9–28], the simplest context-invariant models treat coarticulation as superposition [e.g., Joos, 1948, *Language* 24, 5–136]. In such a model, the intrinsic biomechanics of the body have been argued to account for many of the complex kinematic interactions associated with coarticulation [Gick *et al.*, 2013, *POMA* 19, 060207]. Largely following the methods described in Fagel [2010, *Dev. Multimod. Interf.* 5967, 294–303], we examine articulatory variation in smiled versus neutral speech to test whether the local interactions of smiling and speech can be resolved in a context-invariant superposition model. Production results will be modeled using the ArtiSynth simulation platform (www.artisynth.org). Implications for theories of coarticulation will be discussed. [Research funded by NSERC.]

2pSC21. Languages across the world are efficiently coded by the auditory system. Christian Stilp and Ashley Assgari (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Independent Component Analysis (ICA) is a powerful method for uncovering statistical structure in natural stimuli. Lewicki (2002 *Nature Neuroscience*) used ICA to examine statistical properties of human speech. Filters that optimally encoded speech were an excellent match for frequency tuning in the cat auditory nerve, leading to suggestions that speech makes

efficient use of coding properties in the mammalian auditory system. However, Lewicki only examined American English, which is neither normative nor representative of the world's languages. Here, fourteen languages were examined (Dutch, Flemish, Greek, Javanese, Jul'hoan, Norwegian, Swedish, Tagalog, Tahitian, Urhobo, Vietnamese, Wari', Xhosa, and Yeyi). Each recording contained speech tokens from a native speaker without any background noise for at least one minute. Maximum likelihood ICA was used to create statistically optimal filters for encoding sounds from each language. These filters were then compared to the same physiological measures analyzed in Lewicki (2002). Languages produced a range of ICA solutions, as expected, but were highly consistent with both statistically optimal filters for American English and physiological measures. Results significantly extend Lewicki (2002) by revealing agreement between response properties of the auditory system and speech sounds from a wide range of languages.

2pSC22. High fidelity analysis of vowel acoustic space. Michael H. Coen (Biostatistics and Medical Informatics, Univ. of Wisconsin-Madison, Madison, WI), Hourii K. Vorperian, and Raymond D. Kent (Waisman Ctr., Univ. of Wisconsin-Madison, Waisman Ctr., 1500 Highland Ave., # 427, Madison, WI 53705, vorperian@waisman.wisc.edu)

Vowel acoustic space is often characterized by polygons, whose vertices are determined by summary statistics such as mean values of the formant frequencies of distinct phonemes. The F1-F2 quadrilateral is the most familiar of these. However, using summary statistics to represent formant-frequency data presents fundamental limitations. These data are inherently lossy—summarizing large amounts of data with single values; *mean* itself is a non-robust statistic, highly sensitive to outliers; and even robust statistics ignore *distributional information* within the data, which can vary markedly among different phonemes and age groups. We introduce a new approach characterizing and measuring change in formant spaces statically and developmentally. This approach treats acoustic spaces as *point clouds* of data, in which no information is abstracted or lost. Within this framework, we measure the *spatial overlap* of sets of formant data using an approach combining optimization theory and computational statistics. This provides highly sensitive measures of both similarity and extent of temporal change. This novel approach is robust with respect to outliers, noise, and missing values. It also has a strong intuitive and rigorous mathematical foundation and is easily visualized. Finally, it enables detailed examination of individual phonemes and *clustering* of speakers identifying shared developmental patterns. [Work was supported by NIH grants # R01-DC 006282 & P30-HD03352.]

2pSC23. Shannon entropy predicts the sonority status of natural classes in English. Fernando Llanos (School of Lang. and Cultures, Purdue Univ., 640 Oval Dr., West Lafayette, IN 47907, fllanos@purdue.edu), Joshua M. Alexander (Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN), and Christian E. Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Speech sounds tend to co-occur in the speech stream according to specific combinatory patterns predicted from their sonority status [Parker, S. G. (2002). *Quantifying the sonority hierarchy*. Unpublished doctoral dissertation, University of Massachusetts, Amherst, MA]. This study introduces a measure of spectral complexity, inspired by Shannon entropy, that ranks American English phonemes into a minimal version of the sonority hierarchy: vowels > approximants > nasals > fricatives > affricates > stops. Spectral complexity for every consonant and vowel in the TIMIT database was calculated by first parsing the phonemes into 20-ms segments and computing an FFT. For each short-term FFT, Shannon entropy was computed using the distribution of relative amplitudes (dB) across frequency. Average entropy across the FFTs was used to index spectral complexity for the phonemes, which were then sorted by sonority status. Results of a between-group comparison with spectral complexity as the independent variable and natural class as the dependent variable revealed the existence of six significantly different groups with spectral complexity ranking according to the sonority hierarchy. These findings suggest that Shannon entropy is a reliable acoustic correlate of sonority and may account for the combinatory patterns of co-occurrence of speech sounds in the speech stream.

2pSC24. Acoustic modeling of the perception of place information in incomplete stops. Megan Willi and Brad Story (Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721, mkittles@email.arizona.edu)

Previous research on stop consonant production found that less than 60% of the stops sampled from a connected speech corpus contained a clearly defined hold duration followed by a plosive release [Crystal & House, JASA 1988]. How listeners perceive the remaining portion of incomplete stop consonants is not well understood. The purpose of the current study was to investigate whether relative formant deflection patterns, a potential model of acoustic invariance proposed by Story and Bunton (2010), is capable of predicting listeners' perceptions of acoustically continuous, voiced stop consonants lacking a canonical hold duration. Listeners were randomly presented a total of 60 voiced stop-consonant VCV stimuli, each 100 ms in duration, synthesized using a computational model of speech production. Stimuli were created using a continuum of 20 equal step constrictions along the length of the vocal tract in three vowel-to-vowel contexts [see Story & Bunton, JSLHR 2010]. Participants listened to the stimuli and performed a forced choice test (i.e., /b-d-g/). The phonetic boundaries predicted by the relative formant deflection patterns and phonetic boundaries obtained by the forced choice test were compared to determine the ability of the acoustic model to predict participants' perceptions. The acoustic and perceptual results are reported. [Work supported by NIH R01-DC011275.]

2pSC25. A spectral filtering method for tracking formants in children's speech. Brad H. Story and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Children's speech is typically characterized by high fundamental frequencies (e.g., 300–600 Hz) that create widely spaced harmonic components, producing an apparent undersampling of the vocal tract transfer function. The purpose of this study is to describe a formant measurement technique based on cepstral analysis that does not require modification of the cepstrum itself or transformation back to the spectral domain. Instead, the spectrum is low-pass filtered with a cutoff point (i.e., cutoff "quefrequency" in the terminology of cepstral analysis) to preserve only the spectral envelope. To test the method, speech representative of a 2 to 3 year-old child was simulated with an airway modulation model of speech production. The model includes physiologically-scaled vocal folds, vocal tract, and trachea and generates sound output analogous to a microphone signal. The true formant frequencies can be calculated independently of the output signal and thus provide test cases that allow for assessing the accuracy of the formant tracking algorithm. Formant analysis will also be applied to children's natural speech samples to demonstrate the method. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

2pSC26. Measures of tone difficulty. Chilin Shih (East Asian Lang. and Cultures, Univ. of Illinois at Urbana-Champaign, 707 S. Mathews Ave., 2090 Foreign Lang. Bldg., Urbana, IL 61801, cls@illinois.edu)

Measures of difficulty are needed in many real life as well as computer-simulated applications. Such measures for text, math and science have long received academic and industrial attention due to the demands for k-12 instruction and assessment. In recent years, the demands for comparable studies of speech are on the rise given the popularity of on-line second language teaching software and games. The goal of this project is to explore whether the acoustic attributes of Mandarin lexical tones obtained from individual sound files can explain their level of difficulty experienced by second language learners. The study uses monosyllabic tones, thus isolating the task from measurements of complexity in order to focus on acoustics. We recorded sound files that are rich in natural variation using different levels of talker-to-listener distance, used quadratic decomposition to obtain three coefficients that represent each tonal contour, and analyzed their relationship with learners' performance. We will report the difference between native and second-language tone perception. The results have potential applications in speech synthesis: to generate tone tokens with different levels of difficulty for language learners, and different levels of talker-to-listener distance.

2pSC27. Sources of variability in consonant perception and their auditory correlates. Johannes Zaar and Torsten Dau (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørstedes Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, jzaar@elektro.dtu.dk)

Responses obtained in consonant perception experiments typically show a large variability across stimuli of the same phonetic identity. The present study investigated the influence of different potential sources of this response variability. It was distinguished between source-induced variability, referring to perceptual differences caused by acoustical differences in the speech tokens and/or the masking noise tokens, and receiver-related variability, referring to perceptual differences caused by within- and across-listener uncertainty. Two experiments were conducted with normal-hearing listeners using consonant-vowel combinations (CVs) in white noise. The responses were analyzed with respect to the different sources of variability based on a measure of perceptual distance. The speech-induced variability across and within talkers and the across-listener variability were substantial and of similar magnitude. The noise-induced variability was smaller than the above-mentioned contributions but significantly larger than the amount of within-listener variability, which represented the smallest effect. To determine how well the source-induced variability is reflected in different auditory-inspired internal representations (IRs), the corresponding perceptual distances were compared to the distances between the IRs of the stimuli. Several variants of an auditory-spectrogram based IR and a modulation-

spectrogram based IR were considered and the importance of the different domains for consonant perception was evaluated.

2pSC28. Double entendre: Embedding a secondary message in pointillistic speech. Gerald Kidd and Christine R. Mason (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, gkidd@bu.edu)

A dual-purpose application for pointillistic speech (Kidd *et al.*, JASA **126**, EL186–201) will be described. The speech was represented by a time-frequency matrix of pure-tone “points” derived from the original stimulus. The pattern of frequencies and intensities of the points coded the primary message while the phases of the points coded an independent secondary message. For example, the audible and intelligible word “shoes” may also convey the (inaudible, unintelligible) ASCII characters of the phrase “The road goes ever on and on...” by the binary phase-coded bit pattern of the pointillistic representation of “shoes.” The success in recovering the secondary message via signal processing was examined while the intelligibility of the primary message in speech recognition experiments was measured. Tradeoffs for accuracy in primary and secondary message transfer/reception were explored for both pointillistic speech and hybrid speech comprising natural and pointillistic representations. Some initial observations about the possible uses and limitations of this approach will be considered. [Supported by AFOSR award FA9550-12-1-0171.]

TUESDAY AFTERNOON, 19 MAY 2015

BALLROOM 4, 1:00 P.M. TO 5:20 P.M.

Session 2pUW

Underwater Acoustics: Historical Perspectives on the Origins of Underwater Acoustics II

David L. Bradley, Cochair

Penn State University, PO Box 30, State College, PA 16870

Thomas G. Muir, Cochair

Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713

Invited Papers

1:00

2pUW1. The Sound Lab at Fort Trumbull, New London, Connecticut, 1945–1996. Cathy Ann Clark (Sensors & Sonar Systems, NUWC DIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

The Navy Underwater Sound Laboratory was formed in 1945 when the U.S. Navy merged on-going efforts of Columbia and Harvard Universities to combat the German U-boat threat in the North Atlantic. Prior to that time, the Division of War Research of Columbia, housed in a single building at Fort Trumbull, was sponsored by the National Defense Research Committee (NDRC) while the Harvard Underwater Sound Laboratory was doing similar work in Cambridge, Massachusetts. For the next 50 years, until it was formally closed in 1996, the “Sound Lab” continued to support virtually all aspects of Naval Warfare technology. This talk will attempt to describe the rich scientific culture and technological contributions of the New London Sound Lab.

1:20

2pUW2. Naval oceanography contributions to underwater acoustics—The Cold War era. Robert S. Winokur (None, 15201 Redgate Dr., Silver Spring, MD 20905, robwinok@aol.com)

Underwater acoustics is an integral part of the Navy and is the major enabler for its capabilities in anti-submarine warfare. The importance of oceanography to naval warfare in general and anti-submarine warfare in particular was clearly demonstrated during World War II. Anti-submarine warfare became high technology in the post World War II era, especially during the Cold War. The period from

1960 to the mid 1980s was an important time in the history and evolution of underwater acoustics, building on the experience and important discoveries and research made shortly before, during and immediately after World War II. Naval oceanography programs, large-scale measurement programs and Navy support for research and development were key elements in the rapid expansion of knowledge in the full spectrum of underwater acoustics and understanding the effect of the ocean environment on the development and operation of anti-submarine warfare systems. This paper will provide an overview of the relationship of Naval oceanography to the rapid expansion of knowledge of underwater acoustics during an important period of the Cold War era.

1:40

2pUW3. Contributions to underwater acoustics by the Naval Ordnance Laboratory (NOL). Ira Blatstein (Johns Hopkins Univ., 2190 Chesapeake Harbour Dr. East, Annapolis, MD 21403, blatstein@jhu.edu) and John Tino (none, Adelphi, MD)

This presentation documents the contributions to underwater acoustics by NOL, built in White Oak, MD, in 1946 and operated until closing (as the White Oak Laboratory of NSWC) in 1997. Initial work on mine systems led to R&D on a variety of more sophisticated passive/active acoustic weapon systems for mine warfare, submarines, and ASW aircraft. The increasingly complex mission requirements led to research in a wide range of underwater acoustic disciplines and included development of some key measurement systems, platforms, and facilities across a broad range of frequencies. The measurements obtained by NOL personnel advanced the state of knowledge in underwater acoustic short and long range propagation, surface and bottom acoustic reverberation, sound speed variability with depth, as well as a variety of other research topics. This research led to the development of sensors, materials, target detection systems, and other systems that were applied to a wide variety of acoustic systems. The evolution of this work, and its wide variety of applications, will be discussed.

2:00

2pUW4. Underwater acoustics at the Naval Air Development Center. Thomas B. Gabrielson (Penn State Univ., PO Box 30, State College, PA 16804, tb3@psu.edu)

Much of the progress in underwater-acoustic research from World War II to the present has been funded by government agencies for development of sonar systems. Although sonar is often considered the domain of surface ships and submarines, the naval aviation community has made significant contributions to basic underwater-acoustic measurement and to acoustic-system development. In the 1940s, the U.S. Navy established the Naval Air Development Center (NADC) in Warminster, Pennsylvania, as the lead laboratory for naval aviation research and development and a substantial part of the work of that laboratory supported development of air-deployed sonar systems. Partnerships between NADC, other laboratories both domestic and foreign, and manufacturers produced a stream of innovative, inexpensive, and expendable devices to support the missions of marine patrol aircraft; these same devices were used extensively for ocean-acoustic measurements. Basic bottom-reflection loss, ambient-noise level and directivity, and reverberation measurements were made using air-deployed sonobuoys and acoustic sources. While lacking the precision of ship-based measurements, the cost of airborne surveys was low, deployment was rapid, and the coverage was ultimately global.

2:20

2pUW5. Pioneers in side scan sonar: Julius Hageman and the shadowgraph. Kerry Commander and Daniel Sternlicht (Sci. and Technol. Dept., Naval Surface Warfare Ctr. Panama City Div., NSWC PCD, Code X, 110 Vernon Ave., Panama City, FL 32407-7001, kerry.commander@navy.mil)

The concept of the side scan sonar was developed during the early 1950s at the U.S. Navy Mine Defense Laboratory, Panama City Florida—now known as the Naval Surface Warfare Center Panama City Division. In technical reports and laboratory notebooks, Dr. Julius Hageman, a German scientist who relocated to the Laboratory after World War II and worked there until his death in 1964, outlined the proposed “short-range high-definition mine location sonar” that would eventually become the C-MK-1 mine classification sonar system, more commonly known as “Shadowgraph.” Hageman’s patent for the concept (US Patent 4,197,591) was first disclosed in 1958, but remained classified until finally issued in 1980. The Shadowgraph was contracted by the U.S. Navy in 1957 and towed primarily from Ocean-going Mine Sweepers. It was operated as an undersea search and survey tool for more than 25 years before decommissioning in 1991. The Shadowgraph was a 1.5 MHz dual-sided side scan sonar, with range of 100 feet, and imaging resolution of approximately 3 in. square at a range of 75 feet. During its service life, it located numerous lost objects and aircraft on the sea floor and to this day has influenced the development of commercial and military sonars.

2:40

2pUW6. A few Canadian contributions to underwater acoustics. Harold M. Merklinger and John C. Osler (DRDC - Atlantic Res. Ctr., PO Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada) (john.osler@drdc-rddc.gc.ca)

An historical perspective on four Canadian achievements in underwater acoustics will be presented: Towed Variable Depth Sonar (TVDS); Chapman-Harris model for reverberation; seabed-interaction in shallow water, and uninhabited underwater vehicles (UUVs) for Arctic acoustics. The poor performance of hull-mounted sonars in Canadian waters during WWII prompted TVDS development in 1947. The resulting prototype (CAST/1X) was tested against UK and US developmental systems in 1958. Subsequently, the UK purchased the Canadian system and the US “re-packaged” their sonars according to the Canadian design. An understanding of the effects of back-scattering from the sea surface and bottom was required to model sonar system performance. Experiments began circa 1960 leading to a new model for sea surface back-scattering, and revealing the ocean volume as an additional factor. The conditions governing propagation and noise in shallow waters remained mysterious in the 1970s, until the significant role of the seabed emerged through theoretical developments (e.g., loss due to shear-waves) and experiments conducted over a variety of seabed types. Political, sovereignty and energy issues prompted Canada to investigate under-ice acoustics in the Arctic beginning circa 1957. The efforts ultimately resulted in the ability to undertake under-ice acoustic and geophysical surveys using long endurance UUVs.

3:00–3:20 Break

2p TUE. PM

3:20

2pUW7. Early acoustics research at the Navy's Pacific research and development laboratory. C. D. Rees (S&T Forecasting, Assessment, and Transition (72120), SSC Pacific, 53560 Hull St., Code 72120, San Diego, CA 92152-5001, Dave.Rees@navy.mil)

Few organizations have a more essential connection to underwater acoustics than the US Navy; the Navy's mission success has been critically dependent on understanding and exploiting underwater acoustics since the advent of submarines. In its several manifestations since WW II, the Navy's San Diego-based R&D laboratory (currently SPAWAR Systems Center Pacific) has played an important role in the development of the underwater acoustics field. This progress has been in conjunction with other San Diego institutions, such as Scripps Institute of Oceanography and the Marine Physics Laboratory, and other Navy laboratories. We provide a historical overview of acoustics research at SSC Pacific and predecessors in the era following WW II, including the impact of the environment, Arctic acoustics, fielded systems, sonar and active acoustics, and marine mammal acoustics, and the scientists and researchers such as Homer Bucker, Sam Ridgeway, Shelby Sullivan and others who worked in developing our modern understanding of underwater acoustics on the edge of the Pacific.

3:40

2pUW8. History of underwater electroacoustic transducer standards, calibration methods, facilities, and some early contributors. David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net) and Anthony Paolero (Code 153, Naval Underwater Warfare Ctr., Newport, RI)

The practice of realizing underwater acoustic standards and methods for the systematic calibration of the wide variety of electroacoustic transducers is rich in tradition and has given rise to some unique facilities. In about 1940, the US Navy Office of Scientific Research and Development established a center for Underwater Acoustic Metrology under the expertise of Columbia University in collaboration with Bell Laboratories, which subsequently resulted in the establishment of the Underwater Sound Research Laboratory (USRL) in Orlando Florida with unique acoustic lake testing facilities. That facility later became a detachment of the Naval Research Laboratory (USRD) and was the home for a concerted effort in transducer development and calibration for many years. The National Bureau of Standards and later the National Institute for Standards (NIST) deferred the activity of establishing and maintaining underwater acoustic standards and calibration methods to the USRD Navy facility. This paper summarizes some of the transducers standards, methods and facilities and many of the important contributions of early pioneers in this field including Ira Groves, Robert Bobber, Joseph Blue, and others. USRD continues today as a division at NUWC, Newport.

4:00

2pUW9. Early history of underwater acoustics research at SACLANTCEN, La Spezia, Italy. Finn B. Jensen (NATO CMRE, Viale San Bartolomeo 400, La Spezia 19126, Italy, finn.jensen@cmre.nato.int)

The SACLANT ASW Research Centre was established in 1959 to provide scientific and technical advice to the Supreme Allied Commander Atlantic (SACLANT) in the field of antisubmarine warfare and to respond to the needs of NATO nations and maritime commands. Hence, it was a NATO-sponsored research center placed in the Italian Navy compound in La Spezia, Italy, and it was internationally staffed (total about 230) with 50 rotational scientists coming from NATO nations on both sides of the Atlantic. This review covers three decades of achievements in ocean acoustics research from the foundation in 1959 to the end of the Cold War in 1989. Both basic and area-related experimental studies of propagation in deep and shallow waters, ambient noise, signal coherence, and reverberation were conducted over the years. From the early 1970s, the Centre also initiated the development of a set of high-fidelity acoustic models, which subsequently were made available to researchers within the NATO community. Finally, the Centre served as a focus for international cooperation by organizing scientific conferences and workshops on a yearly basis and by conducting multinational measurement programs in the Mediterranean, the Black Sea, and the Eastern North Atlantic from Gibraltar up to the Barents Sea.

4:20–5:20 Panel Discussion

TUESDAY AFTERNOON, 19 MAY 2015

DUQUESNE, 1:45 P.M. TO 2:45 P.M.

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

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

D. S. Houser, Vice Chair ASC S12
National Marine Mammal Foundation, 2240 Shelter Island Drive Suite 200, San Diego, CA 92106

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

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note-those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

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

2p TUE. PM

TUESDAY AFTERNOON, 19 MAY 2015

DUQUESNE, 3:00 P.M. TO 4:15 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

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

P. B. Nelson, Vice Chair ASC S3
Department of SLHS, University of Minnesota, 115 Shevlin, 164 Pillsbury Drive S.E., Minneapolis, MN 55455

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note-those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

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

Meeting of Accredited Standards Committee (ASC) S12 Noise

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

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

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

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note-this meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

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

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	Brigade
Acoustical Oceanography	Rivers
Animal Bioacoustics	Brigade
Architectural Acoustics	Ballroom 3
Physical Acoustics	Ballroom 4
Psychological and Physiological Acoustics	Commonwealth 2
Structural Acoustics and Vibration	Commonwealth 1