Session 3aAAa

Architectural Acoustics: Separating Spaces: Adventures in Acoustic Isolation of Acoustically Sensitive Spaces

Shane J. Kanter, Cochair
Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604

Stephanie Hoeman, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

3aAAa1. An examination of the minimum construction necessary to improve the sound isolation in a university's existing music building. Ashley Masoner (Acoust. Dimensions, 120 N Pelham Rd., Apt. 1L, New Rochelle, NY 10805, amasoner@gmail.com)

The National Association of Schools of Music (NASM) is the organization responsible for accrediting both degree- and non-degree granting music schools in the United States. Typically, NASM accreditation is based on the musical program itself, but what happens when an existing building's sound isolation is so poor it threatens a school's accreditation status? As is often the case in the Arts, the University of Wyoming has a limited budget to use for upgrades in their existing Music building. Through a series of mock-up tests, Acoustic Dimensions was able to identify the most critical noise-flanking paths and recommend the most cost effective means of raising the NIC values between practice rooms to levels that better accommodate simultaneous use of adjacent spaces.

8:25

3aAAa2. Night club skylight vs. hotel—A case study. Shane J. Kanter, Carl Giegold, Constance Walker, and John Strong (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

The Paris Club, a popular night club in Chicago, Illinois, is enclosed by an enormous, operable skylight. This feature spans across the entire space, acting as the only "barrier" between the club and the directly adjacent hotel and high end condominium buildings. From within the Paris Club, the condominium balconies and hotel windows are in direct line-of-site. Although the skylight is a wonderful architectural feature, it is devastating to the isolation of thumping music played within the club. Through various testing methods, the isolation performance was measured and required performance was determined. Calculations and a laboratory tested mockup yielded a viable solution.

8:45

3aAAa3. The isolation of Lookingglass Theatre, or how an active municipal pumping station became the site for a successful theater. Gregory A. Miller (Threshold Acoust., LLC, 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com and Richard H. Talaske (TALASKE/Sound Thinking, Oak Park, IL)

In search of a new permanent home, the Lookingglass Theatre Company was offered the chance to transform a former steam generation plant within the Chicago Avenue Pumping Station into a multi-venue theater. The Pumping Station is a historic structure—one of the few to survive the 1871 Chicago Fire—but remains an active part of the municipal water supply, pumping water to the near north side of the city. The pump room is less than 5 ft. from the site of the main theater, a 270-seat reconfigurable space specified to achieve a background noise level of RC-25. This paper will describe design challenges identified for achieving this level of acoustic separation, the construction challenges encountered building within a 130-year-old historically protected structure, and the successful result achieved by overcoming these challenges.

9:05

3aAAa4. Design and implementation of an open-plan art gallery with heavily amplified audio/video works. John T. Strong and Carl P. Giegold (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, jstrong@thresholdacoustics.com)

An exhibit consisting of short film and video works by internationally recognized artist and film director Steve McQueen was installed in a large open-plan gallery at the Art Institute of Chicago. Several of the works included a heavily amplified audio component with very high levels of low-frequency energy, such as machine noise and hip-hop music. The layout of the gallery dictated that these works be placed in close proximity to each other as well as to works requiring a very low noise floor, while maintaining an open-plan

format within an existing gallery space. Temporary enclosures were designed and constructed to allow these works to be displayed concurrently without significant disturbance to each other. Providing favorable room acoustics for the heavily amplified program within each space was also of primary importance. Panel absorbers tuned to the modal characteristics of each space as well as the unique acoustic characteristics of each work were integrated into the construction along with diffusive and absorptive surfaces to provide a comfortable listening environment for listeners while maintaining a uniform appearance.

9:25

3aAAa5. Two case histories relating to the transmission of noise from mechanical equipment rooms to adjacent noise sensitive spaces. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, coffeen@ku.edu)

Two recent noise control projects related to the reduction of noise from mechanical equipment rooms to an adjacent conference room and to an adjacent office. For both of these situations, the noise control procedures to be implemented involved the determination of the most significant noise transmission path; air borne noise transmitted through a concrete floor slab or structurally transmitted noise due to lack of proper chiller and pipe vibration isolation in the conference room case, or for the office case airborne noise transmitted through a wall constructed of gypsum wall board and flanking through common floor slabs or structurally transmitted due to lack of suitable pump and piping vibration isolation. A method for determining the most significant noise transmission path and the resulting noise reduction for these two cases will be described.

9:45

3aAAa6. Case study: Isolation of a television production studio from airborne noise and ground-borne vibration. Eric L. Reuter (Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

This project involved design of a 650-s.f. television and audio production studio in an existing historic building situated along a busy urban street with a subway tunnel below. Low background noise in the space was the primary concern, but the possibility of noise from music performance in the space interfering with nearby editing rooms and on-air studios was also considered. In order to satisfy all of the isolation requirements, the studio was constructed as a free-standing structure, supported by spring isolators on a new steel structure, within the existing building shell.

Contributed Papers

10:05

3aAAa7. Upgrading secret military facilities—What is more important, acoustic design standards or acoustical performance? Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

The U.S. Department of Defense has developed acoustical performance standards that are to be achieved in the design and construction of meeting and conference rooms where sensitive and secret information will be discussed. These performance standards rely on published acoustical industry design data, which are readily available. The intention of these standards is to prevent sensitive and secret information from being heard, understood, or otherwise obtained by persons or devices that are not authorized to have access to such information. This paper presents design and field performance test results for new and renovated secret rooms that initially passed the acoustical design criteria and acoustical standard field tests, but failed to provide the desired secret level acoustical performance. Further investigations and research into partition, component, and building composite performance indicated that floors, walls, ceilings, doors, windows, and perimeter penetrations by conduit and HVAC ducting, which individually met the design standards and when installed meet the design standards, but as a composite did not provide the intended acoustical performance that would prevent unauthorized access to sensitive and secret information by persons or devices outside the designated perimeter. Reasons for certain performance failures are discussed and specific successful remedies are presented.

10:20

3aAAa8. Development of a low-frequency impact noise metric. David W. Dong and John LoVerde (Veneklasen Associates, 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Low frequency footfall noise (“thudding”) is a common source of complaints in lightweight joist-framed multifamily projects. Previous work by the authors has indicated that the low frequency impact sound pressure levels (LFISPL) from a standard ISO tapping machine are highly correlated with occupant reaction [LoVerde and Dong, J. Acoust. Soc. Am. **112**, 2201 (2002), LoVerde and Dong, INCE Inter-Noise 2004 Proceedings, **167** (2004)]. In order to be useful, the raw LFISPL must be translated into a single number that maintains the high correlation with subjective reaction while providing adequate dynamic range to distinguish the performance of different assemblies. Candidate metrics are evaluated.

10:35

3aAAa9. Acoustical characteristics of restorative space on a university campus. Abigail Bristow (School of Civil and Bldg. Eng., Loughborough Univ., Loughborough LE11 3TU, United Kingdom, a.l.bristow@lboro.ac.uk) and Kirill V. Horoshenkov (School of Eng., Univ. of Bradford, Bradford, United Kingdom)

This paper explores the nature, sound environment, and value of restorative space on University campuses in the United Kingdom that is separated from standard teaching venues. Questionnaire surveys were undertaken in the Atrium and Student Central at the University of Bradford and in the Library at Loughborough University to explore expectations and perceptions of restorative quality, preferences, values, and use on non-teaching space. These spaces are used routinely by the students for social, relaxation, individual, and group studies. Contemporaneous, continuous measures of noise were undertaken at the survey locations. Noise levels in a social space on campus can be as high as 85 dB, which affects the restorative quality of the space. The results of these surveys enabled us to explore the link between the acoustical characteristics of a space and its restorative quality, and to identify key sounds which contribute to the value of a restorative space. The value of such spaces is reflected in the willingness to pay for quieter and greener spaces.

Session 3aAAb

Architectural Acoustics and Noise: Restaurant Acoustics

Eric L. Reuter, Cochair
Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801

Steven D. Pettyjohn, Cochair
The Acoustics & Vibration Group, Inc., 5700 Broadway, Sacramento, CA 95820

Chair's Introduction—10:55

Invited Papers

11:00

3aAAb1. Design of a new restaurant and remedial design of an existing café: Reducing noise of patron activities. Steve Pettyjohn (The Acoustics & Vibration Group, Inc., 5765 9th Ave., Sacramento, CA, spettyjohn@acousticsandvibration.com)

R. Selands is an experienced restauranteur with an understanding of the importance of controlling excess sound, as was his architect from the Netherlands. When designing a new restaurant in Sacramento called Ella, acoustics was an important item. Both sound from the HVAC system and patron noise were evaluated. Acoustical treatment of patron noise had to fit in with the aesthetic plan by the architect. HVAC sound was controlled using specific wall and floor/ceiling design and spring isolation. The result was very successful with continuous compliments from patrons. Some issues still exist where a decision was made to not use acoustical treatment. When designing a new market and café, acoustics was not high on the list until the day the facility was opened. Within three days, calls were made requesting assistance to reduce or eliminate excess sound because of complaints from patrons. Site visits, limited field tests, and drawings were used to evaluate the architectural acoustics. Options were provided to modify room finishes to reduce the reverberation time within the space. The most difficult task was modifying the tin-type ceiling to allow sound absorptive material to be placed above.

11:20

3aAAb2. Disco dining: Where DJ culture meets high-end restaurants. Tyler Adams (JBA Consulting Engineers, 36 Technol. Dr. Ste. 200, Irvine, CA 92618, tadams@jbace.com) and Michael Schwob (JBA Consulting Engineers, Las Vegas, Nevada)

Live DJs have quickly become a popular feature in the contemporary landscape of high-end restaurants. Many new establishments are sacrificing prime dining floor area so they may prominently feature a DJ booth. The Restaurateur's goal is not simply to entertain patrons and create ambiance but to use the name-recognition of DJs to draw customers and impart a sense of luxury and ”cool” cachet. Along with this cultural phenomenon comes an assortment of sound isolation and room acoustics issues that must be addressed to provide patrons with an enjoyable dining and listening experience and ensure the thumping dance music results in minimal impact to adjacent spaces and properties. In this talk, a number of such issues will be discussed using real-world problems and solutions from current high-end restaurants.

11:40

3aAAb3. Examining the relationship between room acoustics parameters and the cafe effect in restaurants. Eric L. Reuter (Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com) and Christopher O'Connor (Berklee College of Music, Boston, MA)

It is well known that the so-called café effect, the progressive increase in speech effort as an enclosed space becomes more crowded with talkers, is related to the reverberant characteristics of the room. This study attempts to correlate specific room acoustics parameters (total absorption, reverberation time, etc.) to the café effect through analysis of several existing restaurants. At each establishment, noise monitoring was conducted over the course of a busy weekend, followed by corresponding impulse response measurements in the empty room. Results and conclusions from the study will be presented.

Session 3aAB

Animal Bioacoustics, Signal Processing in Acoustics, Psychological and Physiological Acoustics, and Speech Communication: Neural Mechanisms of Complex Sound Discrimination I

Andrea Simmons, Cochair
Brown Univ., Box 1821, Providence, RI 02912

Hiroshi Riquimaroux, Cochair
Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan

Invited Papers

8:00

3aAB1. Hierarchical effects of attention on amplitude modulation encoding in auditory cortex. Mitchell Sutter, Kevin N. O'Connor, Joshua Downer, Jeffrey Johnson, and Mamiko Niwa (Ctr. for Neurosci., Univ. of California Davis, 1544 Newton Cr, Davis, CA 95618, mlsutter@ucdavis.edu)

How attention influences single neuron responses in the auditory system remains unresolved. We found that when monkeys actively discriminated temporally amplitude modulated (AM) from unmodulated sounds, primary auditory (A1) and middle lateral belt (ML) cortical neurons better discriminated those sounds than when the monkeys were passively listening. This was true for both rate and temporal codes. Differences in AM responses and effects of attentional modulation on those responses suggest: (1) attention improves neurons' ability to temporally follow modulation (2) non-synchronized responses play an important role in AM discrimination (3) ML attention-related increases in activity are stronger and longer-lasting for more difficult stimuli consistent with stimulus specific attention, whereas the results in A1 are more consistent with multiplicative nonlinearity, and (4) A1 and ML code AM differently; ML uses both increases and decreases in firing rate to encode modulation, while A1 primarily uses activity increases. These findings provide a crucial step to understanding both how the auditory system encodes temporal modulation and how attention impacts this code. Further, our findings support a model where rate and temporal coding work in parallel, permitting a multiplexed code for temporal modulation. [Work supported by NIDCD RO1 DC-02514.]

8:20

3aAB2. Forebrain processing of complex sounds: From mice to humans. Christoph E. Schreiner, Craig Atencio, Jonathan Shih, Patrick Hullett (Otolaryngol., UCSF, 675 Nesln Rising Ln., San Francisco, CA 94143-0444, chris@phy.ucsf.edu), and Edward Chang (Neurological Surg., UCSF, San Francisco, CA)

The decomposition and re-integration of complex sounds, such as speech, are at the core of auditory cortical processing. We will discuss the transformation of this process between subcortical and cortical stations from single-filter to multiple-filter models and its relationship to the structural organization of auditory cortex in animal models. We will compare findings of recordings from human superior temporal gyrus to speech sounds with the findings obtained from animal models and discuss potential implications for speech representation in primary and non-primary cortical areas.

8:40

3aAB3. Individual identification of Japanese macaques by coo-calls: Pitch or vocal tract characteristics? Takafumi Furuyama, Kohta I. Kobayasi, and Hiroshi Riquimaroux (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

Japanese macaques, *Macaca fuscata*, utter the Smooth Early High (SEH), one of harmonically structured coo-calls, for greeting and locating other individuals. The purpose of this study was to examine acoustical features of SEH for them to identify the caller. Two male Japanese macaques were trained to discriminate SEHs of Macaque A (SEHa) from those of B (SEHb). The fundamental frequencies (pitch) of SEHa and SEHb were different. Those calls were recorded from monkeys unfamiliar to subjects. GO-NOGO paradigm was used for training the animals. SEHa served GO stimuli (S+) while SEHb served NOGO stimuli (S-). A "Hit" was reinforced with about 1 ml of orange juice. Unfamiliar SEHa and SEHb, which were never used for training sessions, were used as test stimuli. A series of morphed SEHs between SEHa (S+) and SEHb (S-) with only pitch or vocal tract characteristics varied were also used for test stimuli. Results showed that reaction times (RTs) for unfamiliar SEHa were significantly shorter than those for unfamiliar SEHb. RTs to morphed stimuli shortened as the distance from SEHb (S-) increased in both pitch and vocal tract dimensions. Data suggested that both pitch and vocal-tract characteristics were important for Japanese macaques to identify individuality.

3a WED. AM

9:00

3aAB4. Early experience improves neural discrimination/recognition of natural complex sounds. Shaowen Bao (Helen Wills Neurosci. Inst., Univ. of California, 210X Barker Hall, Berkeley, CA 94720, sbao@berkeley.edu)

In natural environments, behaviorally relevant complex sounds are often produced with "speaker" variability and contaminated with environmental noises. To efficiently discriminate/recognize natural complex sounds, the auditory system has to tune to defining acoustic features and filter out random, meaningless features. In humans, for example, efficient speech recognition is achieved by enhancing perceptual contrast for native speech sounds, as well as reducing perceptual contrast for non-native speech sounds. The neural mechanisms underlying this perceptual transformation are still not well understood. We exposed juvenile rats to heterospecific vocalizations recorded in a natural environment, and subsequently examined their cortical complex sound representations. Cortical neurons became more responsive to dynamic and complex features of the complex sounds. In addition, more neurons were involved in representing the whole set of complex sounds, but fewer neurons actually responded to each individual sound. Cortical responses to different renderings of the same song motif were more similar, and responses to sounds of different motifs became more distinctive, indicating that cortical neurons were more selective to the defining features of the experienced sounds. These effects lead to better neural discrimination/recognition of the experienced complex sounds.

9:20

3aAB5. From sounds to meaning: Neural representation of calls in the avian auditory cortex. Julie E. Elie and Frederic E. Theunissen (Psych. Dept. & Helen Wills Neurosci. Inst., Univ. of California, Berkeley, 3200 Tolman Hall, Berkeley, CA 94720, julie.elie@berkeley.edu)

Understanding how the brain extracts meaning from communication sounds is a central question in auditory research. Communication sounds distinguish themselves not only by their acoustical properties but also by their information content. To study how the auditory system could differentially treat signals that have different social meanings, we investigated in zebra finches (*Taeniopygia guttata*) the perception of vocalizations that are used in clearly distinct contexts. We first generated a vocalization library containing the entire repertoire of female and male zebra finches. We then investigated the neural representations of these social calls in primary and secondary auditory areas. Using both simple measures of spike rate and optimal decoding methods based on entire PSTH, we indentified 24% of 1400 single units selective for the different call categories. To further understand how neurons could process sounds to generate selective responses, we compared models of the neural response based on the acoustic properties of sounds and/or on the semantic values of calls. We found neurons that were very well explained by the simple semantic model. Combining these results with the anatomical properties of cells (positions and spike shapes) gives new insight into the neural representation of meaningful stimuli in the avian auditory neural network.

9:40

3aAB6. Cortical representation of complex spectrotemporal features in songbirds. Gunsoo Kim, Helen McLendon (Physiol., UCSF, 675 Nelson Rising Ln., Rm. 521, San Francisco, CA 94143, gkim@phy.ucsf.edu), and Allison Doupe (Psychiatry, UCSF, San Francisco, CA)

Mechanistic understanding of how the auditory cortex processes complex communication signals remains a challenge. With their rich vocal communication behaviors, songbirds can offer insights into this question. We investigated the neural representation of sound features in the cortical auditory areas of zebra finches. In the primary cortical area field L, our systematic mapping of spectrotemporal receptive fields revealed a highly organized representation in which sharpness of spectral and temporal tuning of sound is mapped along two separate anatomical axes. The clustering of temporally or spectrally selective neurons suggested that initial cortical filtering for basic perceptual qualities such as tempo and pitch occurs in a spatially organized and segregated manner. Moreover, using an information theoretic based technique, we are uncovering additional sound features beyond those represented by conventional spectrotemporal receptive fields. In field L, we find that many neurons encode a second feature that typically captures rapid spectral or temporal modulations overlapping the first feature. In the secondary auditory area CM, a major target of field L, we are discovering an emergent sensitivity to frequency stacks, prevalent in zebra finch vocalizations. Together, our data show a systematic and hierarchical mapping of sound features onto songbird cortical neurons.

10:00–10:20 Break

10:20

3aAB7. Neural encoding of learned auditory feedback statistics in avian vocal-motor circuitry. Kristofer E. Bouchard (Neurosurgery, UCSF, San Francisco, CA) and Michael S. Brainard (Physiol., UCSF, 675 Nelson Rising Ln., San Francisco, CA 94158-0444, msb@phy.ucsf.edu)

Many complex behaviors are supported by neurons with both sensory and motor properties. During behavior such sensory-motor neurons experience the probabilistic associations of pre-motor activity for current actions with feedback from previous actions. Consequently, these associations might become encoded through Hebbian mechanisms. To investigate this possibility, we measured whether and how auditory-motor neurons in the vocal premotor nucleus HVC of songbirds encode the relative probabilities that different syllable sequences were produced during singing. We recorded from HVC while playing pseudo-randomly sequenced syllables from the bird's repertoire and found that auditory responses to syllables were positively modulated by the conditional probability that preceding sequences were produced during singing. Moreover, responses integrated over seven or more syllables, with the sign, gain, and temporal extent of integration depending strongly on probability. Our findings indicate that encoding of probabilistic associations between current and previous sounds may be a general principle of vocal-motor circuits.

10:40

3aAB8. Local field potentials in the big brown bat inferior colliculus track the flow of objects moving past the bat. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu), Andrea M. Simmons (Cognit., Linguistic, and Psychol. Sci., Brown Univ., Providence, RI), Michaela Warnecke, and Jonathan R. Barchi (Neurosci., Brown Univ., Providence, RI)

Echolocating big brown bats orient and guide their flight in cluttered environments by following narrow corridors through the maze of surrounding obstacles. When flying in clutter, the bat aims its head, broadcast beam, and external ears to the front, to determine whether the upcoming path forward is safe to enter. This stabilizes the surroundings relative to flight direction to make the flow-field passing by the bat on the left and right stand out from the scene. To examine the capability of the bat's auditory system to follow the clutter as it moves past the bat, we recorded local field potentials (LFPs) from the inferior colliculus of anesthetized bats. We presented sounds that mimic FM biosonar broadcasts followed by clusters of echoes representing rows of objects moving past the bat. LFPs evoked by broadcast-echo sequences tracked the movements of the scene by registering arrays of echoes at latencies corresponding to echo delays. The limit for LFPs to follow multiple clutter objects is related to neuronal recovery times of several milliseconds. These data show that neural activity in the inferior colliculus represents dynamic changes in echo flow-fields that the bat would encounter in free flight. [Work supported by ONR and NSF.]

11:00

3aAB9. Different forms of auditory-vocal feedback control in echolocating bats. Walter Metzner (IBP, UCLA, 621 Ch. E. Young Dr. S., Los Angeles, CA 90095-1606, metzner@ucla.edu)

Auditory feedback from the animal's own voice is essential during bat echolocation: to optimize signal detection, bats continuously adjust various call parameters in response to changing echo signals. Horseshoe bats exhibit a particularly well-developed form of auditory feedback. Their echolocation pulses are dominated by a constant frequency component that matches the frequency range they hear best. To maintain echoes within this "auditory fovea," horseshoe bats constantly adjust their echolocation call frequency depending on the frequency of the returning echo signal. This Doppler-shift compensation behavior represents one of the most precise forms of sensory-motor feedback known. When examining the Lombard effect in horseshoe bats, we found that noise had different effects on call amplitude and frequency rises indicating different neural circuits and/or mechanisms underlying these changes. Both, amplitude and frequency rises were extremely fast and occurred in the first call uttered after noise onset, suggesting that, in contrast to Doppler-shift compensation, the Lombard effect did not require any auditory feedback. Bats also possess a large repertoire of communication calls, which differ greatly from those emitted during echolocation. We compared the variability of echolocation pulses and one common type of communication signal and found fundamentally different feedback mechanisms for echolocation and communication.

11:20

3aAB10. Audiomotor activity in the superior colliculus of the big brown bat engaged in a natural, acoustic orientation task. Melville J. Wohlgemuth and Cynthia F. Moss (Psych. and ISR, Univ. of Maryland, College Park, MD 20742, melville@umd.edu)

To accurately select and orient to a target, noisy, and multimodal sensory information about the target's location must be integrated into a coordinated set of orienting movements. At the hub of sensorimotor integration for species-specific orientation is the superior colliculus (SC), a midbrain structure receiving multimodal sensory inputs and projecting to premotor nuclei throughout the brainstem. Our research brings together behavioral and chronic neural recording data to examine auditory and premotor activity in the SC of the echolocating big brown bat as it performs a natural, goal-directed task. We trained bats to rest on a platform and track a tethered insect moved by a computerized stepper motor system. While the bat was tracking and capturing insects, single neuron activity was recorded across superficial, intermediate, and deep layers of the SC. Neural activity across the laminae of the SC signal auditory and pre-motor events: Echoes reflected from the sonar target evoked activity in superficial and intermediate layers, while premotor activity related to pinna, head, and vocal-motor behaviors was found at deeper recording sites. Collectively, the results of this study contribute to a deeper understanding of midbrain audiomotor activity in the context of natural goal-directed tasks.

11:40

3aAB11. Neural processing of pressure and particle motion in central auditory pathways of larval bullfrogs. Andrea Simmons and Victoria Flores (Cognit., Linguistic and Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912, Andrea_Simmons@brown.edu)

The metamorphic transition from an aquatic to a terrestrial milieu considerably impacts the functioning of the anuran auditory system. Neural responses to underwater particle motion produced by z-axis vibration and to pressure waves transmitted through the air/water interface can be recorded from the tadpole's dorsal medulla and torus semicircularis (auditory midbrain). Before metamorphic climax, these responses likely reflect stimulation of the saccule. Particle motion sensitivity in the dorsal medulla is stable in frequency range and sensitivity throughout larval development. In contrast, coding of both pressure waves and particle motion in the torus semicircularis is highly variable. There is a transient loss of pressure sensitivity in a short stage range ("deaf period") prior to metamorphic climax, correlated with the development of the middle ear. Robust responses to particle motion are seen in the torus semicircularis during late larval stages and throughout the "deaf period." During climax, however, these responses are considerably degraded or lost completely. We interpret this second "deaf period" to reflect central neural, rather than peripheral mechanical, effects, likely related to rerouting of afferent pathways.

Session 3aAO

Acoustical Oceanography: Munk Award Lecture

Andone C. Lavery, Chair
Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536

Chair's Introduction—11:00

Invited Paper

11:15

3pAO1. Ten years of seismic oceanography: Accomplishments and challenges. W. Steven Holbrook (Geology and Geophys., Univ. of Wyoming, 1000 E. University Ave., Laramie, WY 82071, steveh@uwyo.edu)

"Seismic oceanography" (SO)—the use of low-frequency marine seismic reflection data to image thermohaline fine-structure in the water column—began in 2003, with the publication of a paper in Science. Over the past ten years, the nascent SO community has demonstrated that reflection seismology can image thermohaline fine structure, over large areas, from temperature contrasts in the ocean of only a few hundredths of a ˚C. The resulting images illuminate many diverse oceanic phenomena, including fronts, water mass boundaries, internal wave displacements, internal tide beams, eddies, turbulence, and lee waves. Beyond merely producing spectacular images of ocean structure, low-frequency reflections can be processed to produce quantitative estimates of sound speed (and thus ocean temperature), turbulence dissipation, and vertical mode structure over full ocean depths, as long as fine-structure reflections are present. Yet SO has failed to become a standard tool for physical oceanographers, partly due to disciplinary boundaries, and partly due to the perceived high expense of seismic data acquisition. I will present examples of the successes of SO and discuss approaches to meet the challenges to the adoption of SO as a commonly used technique to study physical oceanographic processes.

WEDNESDAY MORNING, 4 DECEMBER 2013 GOLDEN GATE 2/3, 7:55 A.M. TO 10:00 A.M.

Session 3aBAa

Biomedical Acoustics: Recent Advances in Therapeutic Ultrasound I

Lawrence Crum, Cochair
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Kullervo Hynynen, Cochair
Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada

Chair's Introduction—7:55

Invited Papers

8:00

3aBAa1. Advances in ultrasound methods for therapy. Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca), Alison Burgess, Meaghan M. O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), Ryan Alkins, Daniel Pajek, Nicholas Ellens, and Alec Hughes (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Focused ultrasound has been shown to be the only method that allows noninvasive thermal coagulation of tissues and recently this potential has been explored for noninvasive image-guided drug delivery. In this presentation, the advances in ultrasound phased array technology for well controlled energy delivery will be discussed. In addition, some of the recent preclinical results for the treatments of brain tumors, stroke, and Alzheimer's disease will be reviewed. As conclusion, the advances in the image-guided focused ultrasound for the treatment of disease has been rapid and the future potential appears very promising.

8:20

3aBAa2. Catheter-based and endoluminal ultrasound applicators for magnetic resonance image-guided thermal therapy of pancreatic cancer: Preliminary investigations. Chris Diederich, Vasant Salgaonkar, Punit Prakash, Matt Adams, Serena Scott, Peter Jones, Daniel Hensley, Henry Chen (Radiation Oncology, UCSF, 1600 Divisadero St., Ste. H1031, San Francisco, CA 94143-1708, cdiederich@radonc.ucsf.edu), Juan Plata, Andrew Holbrook, Kim Butts Pauly, and Graham Sommer (Radiology Dept., Stanford Univ., Stanford, CA)

Ultrasound devices are being investigated for endoluminal and intraductal access for targeted thermal ablation or hyperthermia of pancreas under MR guidance and temperature monitoring. Simulations using patient-specific 3D models were developed for applicator design and development of treatment delivery strategies. MR-compatible devices were constructed for endoluminal (3-5 MHz planar or lightly focused rectangular elements, 12-mm OD assembly, expandable balloon), transgastric interstitial and intraductal (6-8 MHz multi-sectored tubular elements, 2-mm catheter) deployment. Micro-coils were integrated for active MR tracking of position and alignment. The proof-of-concept devices were tested in phantoms, *ex vivo* tissues, cadaveric porcine models, and *in vivo* animal models under 3T MR temperature imaging (MRTI). Results indicate endoluminal devices could ablate 2-2.5 cm depth from gastric wall for tumors of the pancreatic head, and multi-sectored tubular intraductal and interstitial applicators could ablate 2.3-3.4 cm diameter targets with directional control. Intraductal applicators could produce effective hyperthermia (>40 C) extending 15 mm radial. Customized tracking sequences could be used to locate 3D position of the applicators. Endoluminal, interstitial, and intraductal ultrasound applicators show promise for ablation or hyperthermia of pancreatic tumors. MR guidance can be employed for positioning these devices with active tracking coils and real time temperature monitoring. (NIH-P01CA159992.)

8:40

3aBAa3. Targeted drug delivery to the brain and brain tumors using focused ultrasound and microbubbles. Nathan McDannold (Radiology, Brigham and Women's Hospital, 75 Francis St., Boston, MA, njm@bwh.harvard.edu)

The physiology of the vasculature in the central nervous system (CNS), which includes the blood-brain barrier (BBB) and other factors, severely limits the delivery of most drugs to the brain and to brain tumors. Focused ultrasound (FUS), when combined with circulating microbubbles, is a noninvasive method to locally and transiently disrupt the BBB at discrete targets and enhance delivery across the "blood-tumor barrier." This talk aims to provide insight on the current status of this unique drug delivery technique, experience with it in preclinical models, and its potential for clinical translation. In particular, methods to monitor the procedure using acoustic receivers and the feasibility of controlling and predicting drug deposition will be reviewed. If this method, which offers a flexible means to target therapeutics to desired points or volumes in the brain, can be translated to the use in humans, it can enable the use of the whole arsenal of drugs in the CNS that are currently prevented by the BBB.

9:00

3aBAa4. Mechanism, monitoring, and drug delivery of the ultrasound-induced blood-brain barrier opening. Elisa Konofagou (Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Worldwide, neurodegenerative diseases account for more than 20 million patients. Aging greatly increases the risk of neurodegenerative disease while the average age of Americans is steadily increasing. Numerous small- and large-molecule drugs have been developed for treatment of neurodegenerative diseases but with mixed success. This is mainly because, when administered systemically *in vivo*, the blood-brain barrier (BBB) inhibits their delivery to the regions affected. Safe and localized opening of the BBB has been proven to present a significant challenge. Focused ultrasound (FUS) in conjunction with microbubbles remains the sole technique that can induce localized BBB opening noninvasively, selectively, and transiently. Over the past few years, our group has been able to unveil several aspects of the technology in order to (a) unveil the physical mechanism of opening, (b) maintain safety, (c) establish a non-MRI type of monitoring technique, (d) control the volume and permeability of opening through the microbubble used, (e) demonstrate large animal feasibility, and (f) determine the range of molecular sizes delivered. We have also shown that neurotrophic agents are capable of triggering downstream effects into the neuronal nucleus through the induced opening. All the aforementioned aspects including initial drug efficacy findings in large animals will be discussed.

9:20

3aBAa5. High-intensity focused ultrasound treatment of prostate cancer. Narendra T. Sanghvi (R & D, SonaCare Medical, 4000 Pendleton Way, Indianapolis, IN 46226, narensanghvi@sonacaremedical.com)

In the last decade, over 40,000 prostate cancer patients have been treated by HIFU systems in over 30 countries. These treatments have been conducted using two ultrasound image guided hifu devices—Ablatherm (EDAP, Lyon, France) and Sonablate® 500 (Focus Surgery, Inc., Indianapolis, IN). In addition, there is a shift in the management of prostate cancer from whole gland radical prostatectomy and radiation to focal treatment of prostate cancer. The focal treatment is guided by meticulous pretreatment imaging with multiparametric MRI to accurately localize the index lesion. The MRI images are used to render 3D deformable model of the prostate gland and provide fusion of US and MRI to guide HIFU treatment resulting in reduced complications of rectal fistula, erectile dysfunction, and urinary incontinence. The results of the clinical studies indicate that patients with recurrent cancer post radiation can benefit from HIFU treatment. Both these devices are marketed in many countries and recently have submitted PMA applications to the FDA to receive clearance to market in the United States. Long term clinical results and status of HIFU devices will be presented.

9:40

3aBAa6. Ultrasound-based neurostimulation in the mouse model. Kim Butts Pauly (Radiology, Stanford Univ., Lucas Ctr., 1201 Welch Rd., Stanford, CA 94305, kbpauly@stanford.edu), Randy King, Patrick Ye (BioEng., Stanford Univ., Stanford, CA), and Julian Brown (Neurobiology, Stanford Univ., Stanford, CA)

Ultrasound-based neurostimulation would be a useful tool prior to MR-guided focused ultrasound treatments in the brain. In this work, we report on our studies on ultrasound-based neurostimulation in the mouse model. We define the success rate as the ratio of the number of positive EMG responses to the number of sonications. A single element ultrasound transducer with a center frequency of 500 kHz was applied to the mouse head via a coupling column and coupling gel on the mouse head. EMG electrodes were placed in the mouse neck and tail muscles to measure contraction of the relevant muscles as the ultrasound transducer is moved across the mouse head. The success rate increases with ultrasound intensity or with ultrasound duration, following a sigmoidal curve. As the ultrasound frequency is increased, the ultrasound intensity must be increased for the same success rate. Movement of the ultrasound transducer across the brain changes the response in the relevant EMG systems such that the neck EMG response is stronger when the transducer is more rostrally placed, while the tail EMG response is stronger when the transducer is more caudally placed. Our findings present evidence for selective targeting in the mouse model of ultrasound-based neurostimulation.

WEDNESDAY MORNING, 4 DECEMBER 2013 GOLDEN GATE 2/3, 10:25 A.M. TO 11:30 A.M.

Session 3aBAb

Biomedical Acoustics: Distinguished Lecture: The Use of Magnetic Resonance-Guided High Intensity Focused Ultrasound to Treat Essential Tremor (ET)

Lawrence Crum, Cochair
Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Kullervo Hynynen, Cochair
Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada

Chair's Introduction—10:25

Invited Paper

10:30

3aBAb1. The use of magnetic resonance-guided high intensity focused ultrasound to treat essential tremor. William J. Elias, Diane Huss (Neurosurgery, Univ. of Virginia, Box 800212, UVA HSC, Charlottesville, VA 22908, wje4r@virginia.edu), Tiffini Voss (Neurology, Univ. of Virginia, Charlottesville, VA), Johanna Loomba, Mohamad Khaled, Robert Frysinger (Neurosurgery, Univ. of Virginia, Charlottesville, VA), Scott Sperling, Scott Wylie (Neurology, Univ. of Virginia, Charlottesville, VA), Stephen Monteith (Neurosurgery, Univ. of Virginia, Charlottesville, VA), Jason Druzgal (Neuroradiology, Univ. of Virginia, Charlottesville, VA), Binit Shah, Madaline Harrison (Neurology, Univ. of Virginia, Charlottesville, VA), and Max Wintermark (Neuroradiology, Univ. of Virginia, Charlottesville, VA)

Advances in ultrasound transducer technology have enabled for transcranial sonication with energy levels adequate to achieve tissue ablation. With MR-guidance and monitoring, precise lesioning is now possible of deep brain targets such as the thalamus and basal ganglia so that stereotactic lesioning is being reconsidered for the treatment of movement disorders. In this phase 1 clinical trial, we investigate the feasibility and safety of MRgFUS for performing a unilateral thalamotomy for medication-refractory essential tremor (ET). According to an FDA-approved protocol, 15 patients with medication-resistant ET underwent unilateral MRgFUS lesioning of the thalamus for dominant limb tremor. Intraprocedural monitoring was conducted with each incremental sonication using MR thermometry and clinical examination. Neurological assessments, validated tremor ratings, MRI, and quality of life data were recorded preoperatively and during a year post treatment. Adverse events were recorded throughout the study duration. Accurate thalamic lesioning was achieved in all cases. Dominant limb tremor subscores improved by nearly 75% while ipsilateral limb tremor was unchanged. Functional activities and quality of life measures improved significantly. Refining of the thalamic target was possible in five cases with subthreshold sonications. Serial MR imaging defined the evolution of the lesioning process.

Session 3aEA

Engineering Acoustics: Non-Traditional Electro-Acoustic Transducer Design I

John B. Blottman, Chair
Div. Newport, Naval Undersea Warfare Ctr., 1176 Howell St., Code 1535 B1170/108, Newport, RI 02840

Chair's Introduction—8:30

Invited Papers

8:35

3aEA1. Thermophone projectors using nanostructure materials. Benjamin Dzikowicz, Jeffrey W. Baldwin, and James F. Tressler (Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil)

Thermophone transducers fabricated from new nanoscale materials hold the promise of a new transducer technology for the Navy with no moving parts that operate over a broad frequency range and can be designed to be lighter and thinner than competing technologies. This potentially makes them ideal for use as high performance conformal projectors on autonomous underwater vehicles, submarines, and other small craft. Although thermophone devices have been understood for nearly a century, [Phys. Rev. **10**, 22 (1917)], new nanostructure materials with extremely low heat capacities per surface area have recently become available which have the potential of greatly increasing their efficiency. Thermodynamic models show that certain surfaces, gasses, and enclosures will increase the acoustic efficiency. However each of these modifications of the base design has drawbacks as well. These will be discussed from a theoretical standpoint and results from laboratory testing will help to verify these hypotheses. [Work supported by NRL.]

8:55

3aEA2. Thermoacoustic sound projectors using carbon nanotubes and other nanostructures. Ali E. Aliev and Ray H. Baughman (Alan G. MacDiarmid NanoTech Inst., Univ. of Texas at Dallas, P.O. Box 830688, BE 26, Richardson, TX 75083, Ali.Aliev@utdallas.edu)

The application of solid-state fabricated carbon nanotube sheets as thermoacoustic (TA) projectors is extended from air to underwater applications. Due to non-resonant sound generation, the emission spectrum of nanotube sheets in air or underwater varies smoothly over a wide frequency range, 1-10^5 Hz. Encapsulating the nanotube sheet projectors using inert gases with low heat capacity provided attractive performance at needed low frequencies, as well as a realized energy conversion efficiency in air of 0.2% and 1.5% underwater, which can be enhanced by further increasing the modulation temperature. We suggest enhancement of sound generation efficiency of encapsulated device by using high quality resonant acoustical windows and modulation of high frequency carrier current with a low frequency resonant envelope. Applications of TA projectors for high power sonar arrays and transparent flexible loudspeakers will be discussed. Finally, the alternative nanostructures for excitation of thermoacoustic sound waves will be surveyed. [We gratefully acknowledge support by Office of Naval Research grant N00014-13-1-0180.]

9:15

3aEA3. First look: Acoustic calibration of carbon nanotube transducers. Dehua Huang and Thomas R. Howarth (NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

Material researchers at the University of Texas at Dallas (UT-D) have recently been reporting on the development of underwater acoustic carbon nanotube (CNT) yarn sheets capable of high acoustic output at low frequencies with broad bandwidth. The principle transduction mechanism for their approach is through thermal acoustic means as opposed to conventional underwater transducers that utilize electromechanical vibrations. This presentation will begin with an overview of the CNTs including the design of a first generation packaging technique that was incorporated for the fabrication of three prototypes. The prototypes were acoustically calibrated in April 2013 at the US Navy acoustic calibration facility in Okahumpka, Florida. The presentation will include measured unbiased and biased transmitting voltage responses (TVRs) and directivity patterns over a two and a half decade band. Final discussions will include on-going research directions for further development. [Work supported by NAVSEA Division Newport.]

9:35

3aEA4. Carbon nanotube thermoacoustic projectors for undersea vehicles. Michael R. Zarnetske and John B. Blottman (Sensor and Sonar Syst., Naval Undersea Warfare Ctr., 1176 Howell St., B1170/R109, Newport, RI 02841, michael.zarnetske@navy.mil)

Renewed interest in the thermophone has developed with the recently demonstrated capability to manufacture carbon nanotube thin films and capacity to emit sound through the thermoacoustic effect. High fidelity broadband sound generation is attributed to the ultra-low heat capacity and low thermal inertia of these films. Motivated by the need for low-frequency, broad-bandwidth, compact sonar projectors to be embedded in the hull of unmanned sea vehicles or in the outer coating of a surface combatant or submarine as a conformal

array, a team of researchers from Virginia Tech, The University of Texas at Dallas, and the Naval Undersea Warfare Center are evaluating these novel materials and devices both theoretically and experimentally. Analytical and numerical simulations support mechanical, thermal and acoustic experimentation. Correlated results will be presented. [Work supported by Office of Naval Research, code 321MS.]

9:55

3aEA5. Alternative tonpilz and bender transducer designs. John Butler (Image Acoust., Inc, 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

Tonpilz transducer designs with half-wavelength water-sized radiating-pistons are commonly used in SONAR arrays. And here the tonpilz piston normally radiates in the broadside direction with reduced output as the array is steered to the end-fire direction. Bender transducer designs, used in low frequency applications, can take the form of a dipole transducer which, as a result of partial self-cancellation, can lead to a low source level output. We present two alternative transducer designs, which solve these problems. In the case of the tonpilz array, a cylindrical-shaped leveraged-transducer design with one-half water wavelength size and modal performance is proposed. This transducer is shown to operate in the first three modes allowing the formation of an element beam that may be steered in the general directions the array is steered, with full output in a single end-fire direction. In the case of the dipole transducer, advantage is taken of the strong near-field dipole acoustic pressure which is used to energize a nearby compliant parasitic resonator yielding a dominant monopole source of greater output. [Work supported in part by ONR.]

10:15–10:30 Break

Contributed Papers

10:30

3aEA6. Multi-degree-of-freedom model of 32(1)-mode cylindrical transducer with inactive elements. Nicholas Joseph and Michael R. Haberman (Mech. Eng., Appl. Res. Labs., The Univ. of Texas at Austin, 901 East 40th St., Apt. 301, Austin, TX 78751, nickjjoseph@gmail.com)

Piezoelectric transducers with cylindrical geometry are often designed to operate in a radial "breathing" mode. In order to tune their performance in a cost effective way, cylinders can be constructed of alternating active (piezoelectric) and inactive (non-piezoelectric) staves. Existing lumped parameter models for such a ring are based on effective piezoelectric properties of the composite ring which reduce the system to a single degree of freedom corresponding to the breathing motion [Butler, J. Acoust. Soc. Am., **59**(2), 480-482, (1976)]. Unfortunately, if the length of the staves is a sufficiently large percentage of the circumference, the transducer may demonstrate a detrimental higher frequency resonance within the desired bandwidth of operation. This parasitic resonance results from bending motion of the staves and can significantly decrease the radiated acoustic pressure and generate distortion. This work presents a multiple-degree-of-freedom lumped parameter model that captures both the breathing and bending resonances of the transducer and provides a more accurate prediction of its effective coupling coefficient. Results are compared with a one-degree-of-freedom model, finite element models, and experimental data. Modifications to account for internal volumes and nonlinear effects are also presented and discussed.

10:45

3aEA7. Design optimization of a piezoelectric microphone with in-plane directivity. Michael L. Kuntzman, Nishshanka N. Hewa-Kasakarage, Donghwan Kim (Elec. and Comput. Eng., The Univ. of Texas at Austin, 2501 Speedway, Stop C0803, Austin, TX 78712, mlkuntzman@gmail.com), Alex Rocha (Microelectronics Res. Ctr., The Univ. of Texas at Austin, Austin, TX), and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

A piezoelectric micromachined microphone with in-plane directivity has been recently introduced [Appl. Phys. Lett. **102**, 054109 (2013)]. The work is inspired by a design introduced by Miles *et al.* [J. Acoust. Soc. Am. **125**(4), 2009], which was, in-turn, inspired by the hearing mechanism of a particular type of parasitoid fly. A rocking structure pivots about a rotational hinge in response to in-plane pressure gradients, and the rocking motion is read by springs attached to the end of the rocking 'teeter-totter' structure, with the springs themselves employing thin piezoelectric films, which operate in a 3-1 mode. Prototypes have been fabricated that employ rocking structures 1 mm x 2 mm in size and functionality has been verified via directivity measurements performed in an anechoic chamber. This presentation will focus on exploring the design space of this sensor, which is accomplished with a hybrid model based on FEA and network models. Designs which maximize SNR are presented, and anticipated microfabrication challenges of these designs are highlighted.

11:00

3aEA8. Tuning a combustive sound source to meet experimental needs. Andrew R. McNeese, Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The Combustive Sound Source (CSS) is a versatile underwater sound source used in underwater acoustics experiments. The source is comprised of a submersible combustion chamber, which is filled with a combustive gas mixture that is spark ignited. Upon ignition, the combustive mixture is converted into high temperature combustion byproducts, which expand and ultimately collapse to smaller volume than before ignition. Acoustic pulses are radiated by the bubble activity. The CSS can be used as a source for array calibration, propagation measurements, bottom characterizations, and sea floor seismic testing. Current environmental regulations and varying experimental needs require a tunable source that allows users to easily alter the source level, bandwidth, center frequency, and signal duration. Present efforts have focused on designing and testing a variety of devices that alter the resultant bubble activity to tune the radiated signals to meet various experimental needs. A new combustion chamber and gas exit ports have been constructed and tested in tank experiments. The results show that the resultant acoustic pulses can indeed be varied, and that tone bursts can be created. Discussions show how the device was tuned to meet specific needs for a particular application. [Work supported by ARL:UT Austin.]

11:15

3aEA9. An ultrasonic actuator working under cryogenic and vacuum circumstance. Zhuzi Chen, Yu Chen, Tieying Zhou, and Deyong Fu (Dept. of Phys., Tsinghua Univ., Beijing 100084, China, chenyu@tsinghua.edu.cn)

In this work, we present an ultrasonic actuator that can work under cryogenic vacuum environment. It can be used for adjusting distance between capacitor electrodes in high temperature superconductor filter (HTSF) to tune its pass-band. The actuator is an single crystal chips driven nut-type ultrasonic motor, which can work under cryogenic vacuum conditions. The stator of a nut-type ultrasonic motor is a nut-shaped octagon with internal thread, which matches with the rotor external thread and a bottom at one end as fixing base. Piezoelectric chips are glued to the sides of the octagon to generate a traveling wave along the circumference. Vibration of the stator

drives the rotor to rotate via friction between thread interfaces and the thread drives the rotor to move along the axis direction. The actuator that we developed can easily acquire micrometer positioning accuracy, which enables it to tune the pass-band of a HTSF effectively. The motor was optimized with FEM harmonic response analysis. The mechanical characteristics and stepping precision of the prototype ultrasonic motor have been measured and discussed. [Work supported by NSFC.]

11:30

3aEA10. Underwater low frequency acoustic projector based on a musical instrument design. Andrew A. Acquaviva and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ.,University Park, PA, aaa261@psu.edu)

An electroacoustic projector that is small compared to the radiated wavelength presents a significant design challenge because its radiation resistance is small and the radiation impedance is highly reactive. Practical designs are often limited by high dynamic strain and high reactive drive currents. Wind musical instruments are also small compared to their lowest playing frequencies. However, wind musical instruments are not limited by the low radiation impedance. They use the low radiation impedance as a part of the regeneration mechanism that sustains the oscillation. This paper examines the hypothesis that an underwater projector designed to operate in water in the manner that a wind instrument operates in air may provide performance that is competitive with conventional electroacoustic low frequency acoustic projectors.

WEDNESDAY MORNING, 4 DECEMBER 2013 EAST LOUNGE, 10:00 A.M. TO 12:00 NOON

Session 3aEDa

Education in Acoustics: Undergraduate Research Exposition Poster Session

Mardi C. Hastings, Cochair
Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30332-0405

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Contributed Papers

All posters will be on display and all authors will be at their posters from 10:00 a.m. to 12.00 noon.

3aEDa1. Experimental and numerical analysis of the effects of depth, diameter, and tension in musical drumhead coupling. Benjamin Boe and Randy Worland (Phys. Dept., Univ. of Puget Sound, 1500 N Waner, Tacoma, WA 98416, bboe@pugetsound.edu)

The low frequency vibrations of two-headed musical drums are known to couple. However, little is known regarding the factors that determine the degree of coupling at higher frequencies. In this study, the effects that depth, diameter, and head tension have on the tendency of the drumheads to couple are investigated. Commercial finite element software was used to model a wide range of drums and to identify trends of coupling according to these factors. The numerical results were used to guide which parameters should be tested in the lab. Experimentally, two oppositely facing Electronic Speckle-Pattern Interferometry systems were used to optically view the simultaneous vibrations of both heads of a drum. Several snare and tom tom drums with different diameters, depths, and tensions were observed. To closely analyze the effect of drum depth a tom tom was modified so a range of depths from 1.5'' to 40'' could be tested while keeping the diameter and tension constant. The optical and numerical data are used to illustrate trends in the coupling of musical drumheads.

3aEDa2. The real world transmission loss chamber: A work in progress. Jay Bliefnick, Andrew M. Hulva, Dominique J. Chéenne (Audio Arts & Acoust., Columbia College Chicago, 5001, Apt. 1S, Schiller Park, IL 60176, jay.bliefnick@loop.colum.edu)

A "real world" transmission loss (RWTL) chamber was recently added to the undergraduate acoustics program at Columbia College Chicago. It aims to demonstrate concepts of transmission loss to students and to provide a "less-than-ideal" environment for construction evaluation prior to certified tests with full-size samples. Each side of the chamber can be used as either "send" or "receive," and both speaker and microphone placements are infinitely variable. The noise floor can also be adjusted to illustrate potential issues in field tests. Measurements are taken simultaneously from multiple positions, then averaged to yield TL values and an overall isolation rating. Given its construction, the chamber is not expected to produce results as per existing standards as its small size results in modal effects and non-diffuse conditions on both sides of the tested partition. This study's first goal was to better understand the performance limitations of the RWTL chamber by conducting a thorough evaluation of its maximum TL and modal properties. The second goal was to optimize the chamber's testing methodology in order to more closely reflect certified laboratory results; this was done by utilizing recently obtained certified data from a door manufacturer who tested reduced-sized samples in the RWTL chamber.

3aEDa3. Just noticeable differential estimation of source-receptor dislocations in the auralization process. Bernardo Murta (Undergrad. Program Acoust. Eng., DECC-CT-UFSM, Federal Univ. of Santa Maria, Rua da Passagem, 111, Belo Horizonte, Minas Gerais 30220-390, Brazil, be.murta@gmail.com), Jessica J. Lins, and Stephan Paul (Undergrad. Program Acoust. Eng., DECC-CT-UFSM, Federal Univ. of Santa Maria, Santa Maria, Brazil)

The precision of source-receptor transfer functions is of importance to provide reliable and ecologically valid results in auralization. A test using paired comparison was developed to estimate the overall just noticeable differential (jnd) in signals obtained from the convolution of music with slightly varying source-receptor transfer functions. Subjects from 19 to 28 years, both male and female, were selected to participate comparing 24 sets of two sounds in each battery and answering if within each pair differences are noticeable. The signals were obtained by convolution of popular rock music with 12 s of duration with impulsive responses of different source-receptor positions in a scaled room and were divided in three batteries with different dislocation directions. To check subjects reliability pairs containing the same signals were also evaluated. The other 20 pairs were made by combinations of signals obtained by the FRF of the reference position and the FRFs of positions dislocated up to 10 cm from the reference point in a random manner. By analyzing the psychometric function, it was found that the jnd was reached when the receiver position was varied about 3 to 4 cm from the reference position. Further tests will be done to clarify details.

3aEDa4. Comparing the time variance of orchestral instrument directivities in Mozart's symphony in G-minor: First movement. Kristin Hanna and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 107 Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0816, khanna@unomaha.edu)

Multiple channel anechoic recordings of the musical instrument parts in Mozart's Symphony in G-Minor (First Movement) are analyzed to study how the directivity patterns of each instrument varies with time. Static directivity patterns are well-documented for many musical instruments, but studies on how their directional patterns vary with time are not as common. Changing directional patterns in time, however, have been found to impact the realism of room acoustic computer modeling simulations. Previous work at the University of Nebraska has suggested a method for studying the time variance of musical instruments across a number of simultaneously recorded channels in an anechoic chamber. The method involves time-windowing each channel and analyzing how the overall directivity index changes across time and frequency. Comparisons of results from some of the fourteen instruments included in this Mozart symphony are presented. [Work supported by a UNL Undergraduate Creative Activities and Research Experience Grant.]

3aEDa5. An analysis of firefighter personal safety alarm effectiveness on the fire ground. Kyle Ford, Mudeer Habeeb, Joelle Suits, Mustafa Abbasi (Dept. of Mech. Eng., Univ. of Texas at Austin, 5907 Trabadora Cove, Austin, TX 78759, kyleford@utexas.edu), Preston S. Wilson (Appl. Res. Lab. and Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX), and Ofodike Ezekoye (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

For firefighters in the line of duty, the last line of defense and chance for rescue oftentimes relies on the effectiveness of their Personal Alert Safety System (PASS) devices. When activated, a PASS device emits an alarm signal to notify others that a firefighter is in distress. However, there have been notable instances where PASS devices have confused rescue personnel or created a more hazardous situation, for instance, when noise interference originating from other objects is involved. This research compiles data from various sources, for example, firefighter near miss reports and National Institute for Occupational Safety and Health (NIOSH) fatality reports, regarding PASS device effectiveness. The research will investigate the causes of confusion and danger as well as take a look at the situations where the device achieved its goal and was able to save a life. The implications of discovering how interfering noises can render PASS devices ineffective could save several lives in the future and ultimately lead to increased firefighter safety.

3aEDa6. Correlation analysis of military aircraft jet noise. Zachary Anderson, Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., 345 E 600 N F1, Provo, UT 84606, zachary-anderson@hotmail.com), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Correlation analysis is useful in extracting spatiotemporal relationships between signals and can be used to examine features of near-field jet noise for source properties. Characteristic correlation envelopes determined by Harker *et al.* [J. Acoust. Soc. Am. **133**, EL458 (2013)] can be used to relate correlation lengths to fine and large-scale turbulent structures. As an extension, cross-correlation shows spatial variation in jet noise and further reveals the transition between short (fine-scale) and long (large-scale) correlation lengths. These analyses are applied to a military jet dataset of a ground based linear microphone array positioned 11.6 m from the jet axis. Correlation analyses over multiple engine conditions and observation directions are reported. In particular, a maximum correlation coefficient greater than 0.5 exists over a range spanning multiple wavelengths in the region of greatest overall sound pressure level at military power. [Work supported by ONR.]

3aEDa7. Autocorrelation analysis of lab-scale jets. Kelly R. Martin, Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, kellymartin013@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Ashville, NC)

Autocorrelation (AC) analysis is useful in examining temporal relationships in a waveform and can be used to provide insight into properties of jet noise. Using techniques developed by Harker *et al.* [J. Acoust. Soc. Am. **133**, EL458 (2013)] for full-scale jet data, AC analysis has been applied to unheated, laboratory-scale jet noise data. To more consistently compare the AC at various locations around the jet, it is important to account for the spatial variation in the spectrum by scaling with the peak frequency. In addition to this frequency scaling, the spatiotemporal variations in the autocorrelation are more plainly seen when an envelope function is applied. Calculated AC envelope functions from measured data are compared with theoretical curves for fine and large-scale jet noise radiation. Results are compared against those from a full-scale, military jet aircraft. [Work supported by ONR.]

3aEDa8. A rapid computational method to investigate the directivities of quasi-omnidirectional sources of sound. Jeshua H. Mortensen and Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., 765 N 400 E, Provo, UT 84606, meako490@gmail.com)

While dodecahedron loudspeakers are widely used in acoustical measurements as quasi-omnidirectional sources of sound, other multiple-driver configurations may also be used for this purpose. Previous experimental work has shown that loudspeakers with higher-order Platonic solid geometries tend to produce higher omnidirectional cutoff frequencies than their lower-order counterparts. However, as their radiated fields transition from omnidirectional to multidirectional at higher frequencies, their directivities may or may not be closer to the omnidirectional ideal. Additional testing has been required to better understand the effects, but it has been cumbersome because of the difficulty of constructing and measuring many modified loudspeakers. This poster presents a practical method to estimate the directional characteristics of multiple-driver sources based on spherical enclosure geometries and the use of common mathematical software such as MATLAB. It enables one to easily and rapidly predict the directivity patterns of these sources and the effects of altered driver diameters, positions, numbers, vibrational patterns, and enclosure volumes. The method is shown to produce several interesting results that are validated by the boundary element method and experimental measurements.

3aEDa9. Evaluation of a small variable-acoustics chamber for speech accommodation research. Matthew F. Calton, Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., 266 N. 300 E. #26, Provo, UT 84606, mattcalton@gmail.com), and Eric J. Hunter (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Many studies have been conducted over the years to explore speech in rooms and its intelligibility to listeners. Speech accommodation by talkers is another developing field in speech and architectural acoustics. In some occupations, a talker's voice is used nearly continuously throughout the workday. Acoustical conditions in the workplace can significantly affect vocal effort and the health and longevity of the vocal folds. Experimental resources are needed to better understand these conditions and how they may be optimized for the well-being of talkers. The present study investigates the range of acoustical conditions that may be established in a small variable-acoustics chamber for this type of research. The chamber is characterized using many pertinent room-acoustics parameters. Volunteers read passages in the chamber with no visual cues to impact their perception of its

changing acoustical treatments. Various measurements were made to establish relationships between the room conditions and vocal efforts.

3aEDa10. Assessing the effectiveness of geometrically modified pyramidal diffusers: Scattering coefficient measurements. Ariana F. Sharma and David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604, arsharma@vassar.edu)

A diffuser is a surface with a non-planar geometry used in acoustically sensitive spaces to help mitigate unwanted effects from strong reflections such as echoes and focusing. Although a variety of diffuser surface geometries exist, new designs are constantly being generated for use in specific real-world projects in an effort to expand the aesthetic options available to the architect and acoustical consultant. The effectiveness of these new designs must be determined as part of the responsible acoustical design process. In the current project, the acoustic behavior of surfaces with a pyramidal base pattern has been measured according to standard ISO 17497-1. In particular, the standard outlines the measurement of the scattering coefficient, a quantifier of how much energy has been reflected away from the specular direction. This coefficient gives a general indication of the diffusing effectiveness of the surface and is useful in computational acoustics room modeling. The surface pyramidal base pattern has been varied to create an array of surfaces in an effort to find an optimal combination of geometric input parameters. Certain measurement conditions prescribed in the ISO standard were also varied to determine their effect on measurement accuracy. Results and analysis will be presented.

3aEDa11. Simulated acoustical environments for the evaluation of vocal effort. Jennifer Whiting (Brigham Young Univ., 2011 South 1175 East, Bountiful, UT 84010, lundjenny@comcast.net), Timothy Leishman (Brigham Young Univ., Provo, UT), and Eric Hunter (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Realistic simulations of acoustical environments allow researchers to quickly manipulate the auditory experiences of human subjects. For studies investigating human speakers' perceptions of their own voices, a mixed-reality environment is most suitable. In this work, we have developed a system to create low-latency real-time convolutions of speakers' voices with simulated room impulse responses at their ears. The latter are based in part on measured voice directivity patterns. The simulated rooms included classrooms, lecture halls, and auditoria. They were all simulated for human speakers within an anechoic chamber. We also added realistic noise to the simulations, including chatter and other ambient effects, and measured the subjects' vocal efforts. The ultimate aim of the simulation and measurement system is to assess teachers' vocal efforts in classrooms and other settings with easily controlled acoustical conditions. However, the setup may also be easily adapted to other studies for speech or music.

3aEDa12. Investigating tonal spaces using an extension of VoiceSauce voice analysis software. Kate Silverstein and Kristine M. Yu (Linguist, Univ. of Massachusetts, 181 Presidents Dr, Amherst, MA 01003, ksilvers@student.umass.edu)

We extended VOICESAUCE (Shue, Keating, and Vicenik, 2009), a MATLAB application which provides automated voice measurements over time from audio recordings, to include utilities for command line processing and testing. The command line utilities allow users to access core VOICESAUCE functionality, including batch processing of wave (*.wav) files and parameter manipulation, independently of a graphical user interface. The testing framework provides an automated process for tracking and measuring the effects of manipulating parameter settings across runs. In addition, we modified VOICESAUCE to be compatible with Octave and ported it to Python in order to facilitate use and development from a wider community. We use this software to compare the inclusion of phonation measures in the set of voice source parameters against f0 alone across White Hmong and Cantonese. Phonation, specifically breathy voice, plays a perceptual role in tone identification in both languages; however, in White Hmong, breathy voice is a necessary cue for accurate tonal identification (Garellek *et al.*, 2012) whereas in Cantonese, phonation may facilitate perception but is not critical (Yu, 2011).

WEDNESDAY MORNING, 4 DECEMBER 2013 — CONTINENTAL 5, 10:00 A.M. TO 2:00 P.M.

Session 3aEDb

Education in Acoustics: Hands-On Acoustics Demonstrations for Middle- and High-School Students

David T. Bradley, Cochair
Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604

Andrew C. Morrison, Cochair
Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomenon. In this session "Hands-On" demonstrations will be set-up for a group of middle school students from the San Francisco area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should email David T. Bradley (dabradley@vassar.edu) or Andrew C. H. Morrison (amorrison@jjc.edu).

Session 3aNS

Noise, Animal Bioacoustics, and ASA Committee on Standards: Wind Turbine Noise

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118

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

Invited Papers

8:30

3aNS1. A statistical analysis of wind turbine A-weighted sound levels. Paul D. Schomer (Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com), George Hessler, and David Hessler (Hessler Associates Inc., Haymarket, VA)

Hessler and Hessler collected a two week database of wind turbine 10-min L-90 A-weighted levels in three orthogonal directions from an essentially solitary wind turbine (unit located at end of north-south row). These data have been analyzed statistically and this analysis shows that the wind turbine emissions during the day and at night form a clearly normal distribution with a mean level of 32 dB and standard deviation of 2.4 dB for daytime, 7 Am to 10 PM and a mean level of 36 dB and standard deviation of 2 dB, for nighttime, 11 PM to 5 AM. The nighttime hours were selected as the louder hours of the night when, presumably, an inversion was present. The statistical plots clearly show the data collected for wind turbine non-operation, the transition region between non-operation and full operation, the region of full turbine power, and the data that represent discreet noisier events. This result from a comprehensive, single survey suggests that it is necessary to analyze the noisier nighttime hours separately from daytime or the entire 24-h day, if one is to correctly predict or measure the noise during these critical nighttime hours.

8:50

3aNS2. A critical analysis of the "Wind Turbine Health Impact Study: Report of Independent Expert Panel". Paul D. Schomer and Pranav K. Pamidighantam (Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

The "Wind Turbine Health Impact Study: Report of Independent Expert Panel" study, herein the "Massachusetts study," says: "The Massachusetts Department of Environmental Protection (MassDEP) in collaboration with the Massachusetts Department of Public Health (MDPH) convened a panel of independent experts to identify any documented or potential health impacts of risks that may be associated with exposure to wind turbines, and, specifically, to facilitate discussion of wind turbines and public health based on scientific findings." It continues to say: "The scope of the Panel's effort was focused on health impacts of wind turbines *per se*." The Massachusetts study treats health affects broadly in accordance with WHO and includes direct health effects, annoyance, and sleep disruption. In many ways, the Massachusetts study is a critique of the literature relating to wind farm acoustic emissions and health effects. This paper is a critique of the critics. In particular, this critique examines some of the physical acoustic findings and some of the social survey findings. The Massachusetts study employed very strict standards to what they deemed to be quality, acceptable studies, and it is only fair that they be judged by their own criteria. It is the judgment of this reviewer that they failed.

9:10

3aNS3. Measuring wind turbine infrasound in the presence of wind. Richard Carman and Michael Amato (Wilson, Ihrig & Associates, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, rcarman@wiai.com)

A windscreen enables noise measurements to be made as accurately as possible under typical field conditions. Measuring wind turbine noise presents a greater level of difficulty than normal. For there to be wind turbine noise, there needs to be wind for the turbines to operate. Greater wind turbine noise is generally associated with higher wind speeds. Multiple windscreens have been found to significantly reduce artificial wind noise in the range of hearing of 20 to 20 kHz, but cannot reduce the very low frequency pressure fluctuations associated with the movement of air during gusts of wind or in the case of interior measurements fluctuations due to wind pressurizing the building. Wind turbines have been demonstrated by others to produce infrasound in the range of 0.5 to 10 Hz. Measuring infrasound in the presence of local wind is a challenge, since finding a less windy time is not an option. Area-wide measurements of wind turbine noise were conducted at two wind turbine facilities. In analyzing the recorded data, a cross-spectral method was used to reduce the transient effects of local wind in the infrasound range. The technique for doing this is presented and its effect on the data is discussed.

9:30

3aNS4. Area-wide infrasound measurements for two wind turbine facilities. Richard Carman and Michael Amato (Wilson, Ihrig & Associates, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, rcarman@wiai.com)

Area-wide measurements of low frequency wind turbine noise were conducted in residential areas adjacent to two different wind turbine facilities in Southern California. The residential measurement location distances ranged from 615 m to 9 km from wind turbines. Additional measurements were also conducted at distances as close as 125 m from the wind turbines. To obtain the residential measurement data, simultaneous digital recordings were made inside and outside residences using microphones designed to achieve a linear response down to 0.07 Hz. The outdoor measurements were conducted with a ground board and two windscreens. The recorded data at residences were analyzed using a cross-spectral technique to minimize the effects of wind acting on the microphone. The data clearly show the presence of infrasound at the blade passage frequency of the wind turbines as well as at the associated harmonics. The primary range of interest is frequencies between 0 and 10 Hz. The residential data in some instances indicate higher levels of infrasound indoors compared to outdoors, indicating a potential amplification of very low frequency sound energy by the residential structure. Representative infrasound data for both facilities are presented and discussed.

9:50–10:00 Break

Contributed Papers

10:00

3aNS5. Acoustic interaction as a primary cause of infrasonic spinning mode generation and propagation from wind turbines. Kevin A. Dooley (Kevin Allan Dooley, Inc., 55-1817 Harbour Square, Toronto, ON M5J 2L1, Canada, kadooleyinc@rogers.com) and Andy Metelka (Sound and Vibrations Solutions, Inc., Acton, ON, Canada)

Relatively balanced load and velocity related pressure waves from the rearward facing surface of each rotor blade, are at a frequency of 1 cycle per revolution of the turbine and are phase shifted by 120 degrees from each other. The superposition of these infrasonic waves destructively interfere. This action results in a non-propagating rotor locked mode; however, the shielding (reflecting) effect of the tower as each blade passes, interrupts the balanced destructive interference for a small portion of rotor angle three times per revolution. The momentary un-balance between the destructive interfering waves results in the generation of Tyler-Sofrin spinning mode series, which propagate into the far field. The spinning mode radiation angles, coupled with the low decay rate of infrasound, result in higher far field sound pressure levels than would be predicted for a point source. An analysis approach partially derived from Tyler-Sofrin (1962) is presented. Field microphone data including phase measurements identifying spinning modes are also presented.

10:15

3aNS6. Significant infrasound levels a previously unrecognized contaminant in landmark motion sickness studies. Kevin A. Dooley (Kevin Allen Dooley, Inc., 55-1817 Harbour Square, Toronto, ON M5J 2L1, Canada, kadooleyinc@rogers.com)

Airborne Infrasound at any given point can be accurately described as fluctuations or cyclic changes in the local barometric pressure. Variations in a motion sickness test subject's elevation result in fluctuations in the surrounding barometric pressure by similar degrees to that experienced on a ship in high seas. Cyclic variation in the lateral or linear velocity of a subject in a vehicle or platform in atmospheric air may also be subject to infrasonic pressure fluctuations due to the Bernoulli principle and possibly vortex shedding effects. Calculations presented demonstrate that in at least one landmark study (McCauley *et al*., 1976) test subjects were exposed to infrasonic sound pressure levels in excess of 105 db at discrete frequencies between 0.063 and 0.7 Hz. The infrasonic sound pressure level necessarily present in cyclic motion in free atmospheric air does not appear to have been accounted for as a nausea influencing factor in the McCauley *et al*. (1976) motion sickness studies.

10:30

3aNS7. Narrowband low frequency pressure and vibration inside homes in the proximity to wind farms. Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Narrowband fast Fourier transform measurements made outside and inside homes indicate that unlike audible tones, low frequency pressure waves penetrate homes virtually un-attenuated. Simultaneous multi-channel linear weighted pressure measurements were made in various locations as well as seismic floor velocities using various dynamic signal analyzers and sensors. Several areas of concern would include harmonics of blade pass frequencies and also modulation at 20, 30, and 40 Hz (audible). Traditional acoustic models do not predict or measure low frequencies as raw unweighted pressure. Measurements to be presented indicate pressure levels much higher than the audible tone pressure levels. Similar signatures were measured as seismic ground vibration in the basements of homes at relatively low levels. In order to minimize the effects in the homes and to locate wind turbines properly, it may be important to establish measurement standards for low frequencies before locating wind turbine developments.

Session 3aPAa

Physical Acoustics, Noise, Structural Acoustics and Vibration, and Engineering Acoustics: Jet and Other Aeroacoustic Noise Source Characterization I

Kent L. Gee, Cochair
Brigham Young Univ., N243 ESC, Provo, UT 84602

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

Invited Papers

8:30

3aPAa1. On the crest factor of noise from supersonic jets. Kent L. Gee, Tracianne B. Neilsen (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

An important consideration in characterizing noise from heated, supersonic jets is the crest factor (CF). The large CF in high-speed jet noise is the result of a positively skewed probability density function for the waveform, which translates into infrequently occurring, large-amplitude positive peak pressures. Sufficient system headroom is required in the data acquisition system to provide an accurate representation of these peak pressures and thus avoid clipping or microphone saturation/distortion. But the question remains as to the importance of capturing the single largest pressure out of potentially millions of waveform samples or if a percentile-based CF is adequate. Measurements near a static tactical aircraft reveal CF increases with engine power, with the maximum CF directed upstream of the overall sound pressure level, and a maximum CF of 20 dB at full afterburner. Second, clipping of measured waveforms at different thresholds reveals that a CF definition based on the 99.99 percentile is sufficient to represent overall and band pressure levels to within 0.1 dB and waveform and time-derivative skewnesses to within ~1%. If an estimate of the time-derivative kurtosis is needed within 1% accuracy, then the 99.999 percentile CF is required for headroom estimates.

8:50

3aPAa2. Tactical aircraft noise reduction using fluidic nozzle inserts. Philip Morris, Dennis McLaughlin, Michael Lurie, and Alex Karns (Aerosp. Eng., Penn State Univ., 233C Hammond Bldg., University Park, PA 16802, pjm@psu.edu)

The noise levels generated by tactical aircraft pose health hazards to personnel working in the vicinity of the aircraft (such as on an aircraft carrier deck) and are annoying to communities close to airbases. The engine exhausts are hot and supersonic and generally operate in an off-design condition, where the nozzle exit and ambient pressures are unequal. This results in shock cells in the jet plume. The interaction between the jet turbulence and the shock cells generates broadband shock-associated noise. The dominant noise radiation is in the downstream direction and is associated with the supersonic convection of turbulence in the jet. This paper describes the development of a technology to reduce both noise sources and involves the controlled injection of air into the diverging section of the nozzle to generate flow corrugations. This enables the jet to operate closer to its design condition and also breaks up the large scale turbulent structures that are responsible for the dominant noise radiation. Both flow and acoustic measurements are described. In addition, steady RANS computations provide information on the flow upstream of the nozzle exit and the effect of injector operating conditions on the flow field. Estimates of nozzle performance are also described.

9:10

3aPAa3. Aeroacoustics of volcanic jets: An overview. Robin S. Matoza (Scripps Inst. of Oceanogr., Univ. of California, San Diego, IGPP 0225, La Jolla, CA 92093-0225, rmatoza@ucsd.edu), David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, Fairbanks, AK), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Darcy E. Ogden (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Explosive volcanic eruptions can inject large volumes of ash into heavily traveled air corridors; they pose a significant societal and economic hazard. They also generate large amplitude atmospheric infrasound waves (~0.01-20 Hz), which can be recorded at thousands of kilometers from the eruption and can provide detailed information on the timing, duration, and relative vigor of the volcanic explosions. In order to provide more detail about the eruption process based on acoustic signals, a quantitative model for the acoustic source process within the volcanic eruption column is needed. Volcanic eruption columns are modeled by a momentum-driven jet flow, transitioning with altitude into a thermally buoyant plume. Infrasound recordings from such activity resemble the large-scale turbulence similarity spectrum, indicating that large-scale volcanic jet flows generate an infrasonic form of jet noise. However, volcanic jet noise deviates from pure-air laboratory jet noise because of complexities such as multiphase flow (especially loading with ash particles); nozzle/crater geometry and roughness; buoyancy effects; and high temperature and density effects. We propose a new framework for understanding acoustic sources at volcanoes based on aeroacoustics research, which is being developed through multi-disciplinary integration of field, numerical, and laboratory studies.

9:30

3aPAa4. High Skewness Infrasound from the eruption of Nabro Volcano, Eritrea: Comparison with supersonic jet and rocket engine data. David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, 903 Koyukuk Dr, Fairbanks, AK 99775, dfee@gi.alaska.edu), Robin S. Matoza (Inst. of Geophys. and Planetary Phys., Scripps Inst. of Oceanogr., La Jolla, CA), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Darcy E. Ogden (Inst. of Geophys. and Planetary Phys., Scripps Inst. of Oceanogr., La Jolla, CA)

An understanding of volcanic jets is critical to determining volcanic eruption column dynamics and mitigating volcanic hazards. However, volcanic jets are inherently difficult to observe directly due to their violence, opacity, and complex multi-phase and multi-component flow features. Recent work has shown similarities between the sound produced from explosive volcanic jets and man-made jet engines and rockets. We show that infrasound generated by the 2011 eruption of Nabro Volcano, Eritrea has high waveform skewness and similar waveform statistics to sound produced by supersonic jet engines and rockets. The infrasound from Nabro reported here strongly indicates that infrasound from some volcanic eruptions is produced in similar ways to man-made jet noise from heated, supersonic jet engines and rockets. Noise sources and flow dynamics of jet engines and rockets are better characterized and understood than volcanic jets, suggesting volcanologists could utilize the modeling and physical understandings of man-made jets.

9:50

3aPAa5. Effective Gol'dberg number for diverging waves. Mark F. Hamilton (Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Interest in characterizing nonlinearity in jet noise has motivated consideration of an effective Gol'dberg number for diverging waves [Baars and Tinney, Bull. Am. Phys. Soc. **57**, 17 (2012)]. Fenlon [J. Acoust. Soc. Am. **50**, 1299 (1971)] developed expressions for the minimum value of Γ, the Gol'dberg number as defined for plane waves, for which shock formation occurs in diverging spherical and cylindrical waves. The conditions were deduced from a generalized Khokhlov solution and depend on the ratio x_{sh}/r_0, where r_0 is source radius, and x_{sh} the plane-wave shock formation distance for $\Gamma=\infty$. Alternatively, by taking the ratio of the nonlinear and thermoviscous terms in Fenlon's Eq. (2), it is proposed here that effective Gol'dberg numbers may be identified for spherical and cylindrical waves: $\Lambda=\Gamma\exp(-\pi x_{sh}/2r_0)$ and $\Lambda=\Gamma/(1+\pi x_{sh}/4r_0)$, respectively. For a given value of Λ, the diverging waves achieve approximately the same degree of nonlinear distortion as a plane wave for which the value of Γ is the same. Conversely, to achieve the same degree of nonlinear distortion as a plane wave with a given value of Γ, the value of Γ for, e.g., a spherical wave must be larger by a factor of $\exp(\pi x_{sh}/2r_0)$. Extensions to other spreading laws are presented.

10:10–10:30 Break

Contributed Papers

10:30

3aPAa6. Nonlinear sound propagation associated with a high mass flow cold jet. Andrew Marshall and Neal Evans (Southwest Res. Inst., 6220 Culebra Rd., San Antonio, TX 78238-5166, andrew.marshall@swri.org)

It is well-known that aircraft and rocket engines produce high amplitude broadband noise, but such noise can also be generated by high-pressure gas venting from piping systems. Compared to rocket and jet engines, however, the gas exiting these systems can be very low in temperature. During a recent full-scale blow-down test at Southwest Research Institute, noise measurements of a cold jet were obtained. High pressure gas was forced through a valve, pipe, and nozzle system to simulate a natural gas blow-down event in order to measure stresses at welded connections. Nitrogen gas flowed vertically through a 50 cm nozzle with an average mass flow rate of 27.7 kg/s. Noise measurements were made perpendicular to the jet direction at two ranges (18.3 and 157 m). Peak amplitudes of 155 and 138 dB were obtained at the near and far range, respectively. A comparison between this data and rocket engine measurements from the literature will be discussed, including indicators of nonlinear propagation.

10:45

3aPAa7. Preliminary phased-array characterization of near-field military jet aircraft noise. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., 657 E 420 N, Provo, UT 84606, blaineharker@byu.net), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Ashville, NC)

Major developments over the past decade in aeroacoustic beamforming techniques provide more accurate estimates of jet noise source phenomena. In a recent experiment, near and mid-field measurements of an F-22A using linear and planar microphone arrays were taken at various engine conditions about the jet plume. To locate and provide accurate amplitude levels of jet noise sources, conventional beamforming techniques are used with various array shading methods. Equivalent source reconstructions are shown for different engine conditions, observation angles, and frequencies to explore the source region. In addition, different datasets from spatially separated arrays are combined for improved source reconstructions and to account for spatially dependent spectral content. These results are preliminary to further techniques—such as deconvolution methods—to better understand noise source mechanisms within the jet plume. [Work supported by ONR.]

11:00

3aPAa8. Aeroacoustic source measurement methods for characterizing the sound generated by ducted flow devices with higher-order modes. Timothy J. Newman, Anurag Agarwal, Ann P. Dowling (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom, tjn25@cam.ac.uk), and Ludovic Desvard (AeroAcoust. Res. Team, Dyson Ltd., Malmesbury, United Kingdom)

The International Organization for Standardization (ISO) method 5136 is widely used in industry and academia to determine the sound power radiated into a duct by fans and other flow devices. The method involves placing the device at the center of a long cylindrical duct with anechoic terminations at each end to eliminate reflections. A single off-axis microphone is used on the inlet and outlet sides that can theoretically capture the plane-wave mode amplitudes but this does not provide enough information to fully account for higher-order modes. In this study, the "two-port" source model is formulated to include higher-order modes and applied for the first three modes. This requires six independent surface pressure measurements on each side or "port." The resulting experimental set-up is much shorter than the ISO rig and does not require anechoic terminations. An array of six external loudspeaker sources is used to characterize the passive part of the two-port model and the set-up provides a framework to account for transmission of higher-order modes through a fan. The relative importance of the higher-order modes has been considered and their effect on inaccuracies when using the ISO method to find source sound power has been analyzed.

11:15

3aPAa9. Ballistic shock wave localization estimation of shooter position and velocity using difference of time of arrival (DTOA) algorithm in orthogonally arranged discrete acoustic arrays: Part II. Murray S. Korman (Dept. of Phys., United Stated Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu) and Antal A. Sarkady (Dept. of Elec. and Comput. Eng., U.S. Naval Acad., Annapolis, MD)

A mathematical algorithm was developed to estimate the parameter space involving both the displacement of the shooter with respect to the array and the velocity of the projectile using the difference in time of arrival, DTOA, of the ballistic shock wave cone at each position of an N-element array. [J. Acoust. Soc. Am. **133**, p. 3506, May 2013.] The array geometry involves orthogonally arranged discrete point-like line arrays using wide-band microphone or piezo-electric elements. The algorithm utilizes a nonlinear least squares parameter fit by summing the squares of [DTOA (experimental)—DTOA (theoretical parameters)] values where the DTOA (theoretical) equation involves a lengthy Taylor series expansion of the exact "difference in time of arrival" theoretical equation. Earlier results, in the absence of noise ($N=7$) showed that the model has good versatility in estimating displacement (location) and velocity in simulated computer trials. Here, near-field noise is simulated by a rotor blade resulting in uncertainty in the arrival time of the shockwave at each sensor, leading to uncertainty in the estimation of the fit parameters. For example, simulated scatter plots of the azimuthal angle vs the elevation angle parameters for different projectile miss distances become useful in computing parameter uncertainty for different signal-to-noise ratios.

11:30

3aPAa10. Nonlinear acoustics of combustion instability in solid-propellant rocket motors. Hunki Lee, Taeyoung Park, Won-Suk Ohm (Yonsei Univ., Seoul, South Korea), and Dohyung Lee (Agency for Defense Development, Daejeon, South Korea)

Combustion instability, a large oscillation of pressure in a combustion chamber, is known to be a major source of rocket failure. A common approach to analyzing combustion instability is to regard it as an acoustical phenomenon in an enclosure, driven by the combustion process occurring in a thin region near the grain (solid propellant). Because of the large pressure excursion associated with combustion instability, it exhibits many salient features of nonlinear wave process such as waveform distortion, shock formation, and even chaotic behaviors. In this paper, a comprehensive analytic model for combustion instability of a solid rocket motor is presented. Our focus is on the way in which nonlinearity manifests itself under complex grain geometry, where the acoustic modes can be either harmonically or anharmonically related. Predictions from the model are compared with the static test data for a few representative rockets.

WEDNESDAY MORNING, 4 DECEMBER 2013 — CONTINENTAL 2/3, 8:00 A.M. TO 9:55 A.M.

Session 3aPAb

Physical Acoustics: Phonons and Lattice Dynamics

Veerle M. Keppens, Cochair
Univ. of Tennessee

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

Chair's Introduction—8:00

Invited Papers

8:05

3aPAb1. Glass-like phonon scattering from spontaneous nanostructures in silver-antimony-tellurium. Jie Ma (Quantum Condensed Matter Div., Oak Ridge National Lab., PO BOX 2008 M.S. 6430, Oak Ridge, TN 37831, jea@ornl.gov), Olivier Delaire, Andrew May (Mater. Sci. and Technol. Div., Oak Ridge National Lab., Oak Ridge, TN), Chris Carlton (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Michael McGuire (Mater. Sci. and Technol. Div., Oak Ridge National Lab., Oak Ridge, TN), Lindsay VanBebber (Dept. of Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN), Douglas Abernathy, Georg Ehlers, Tao Hong (Quantum Condensed Matter Div., Oak Ridge National Lab., Oak Ridge, TN), Ashfia Huq (Chemical and Eng. Mater. Div., Oak Ridge National Lab., Oak Ridge, TN), Wei Tian (Quantum Condensed Matter Div., Oak Ridge National Lab., Oak Ridge, TN), Veerle Keppens (Dept. of Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN), Shao-Horn Yang (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Brian Sales (Mater. Sci. and Technol. Div., Oak Ridge National Lab., Oak Ridge, TN)

Materials with very low thermal conductivity are of great interest for both thermoelectric and optical phase-change applications. Synthetic nanostructuring is most promising for suppressing thermal conductivity arising from scattering phonons, but challenges remain in producing bulk samples. In crystalline AgSbTe2, we show that a spontaneously forming nanostructure leads to a suppression of thermal conductivity to a glass-like level. Our mapping of the phonon mean-free-paths provides a novel bottom-up microscopic account of thermal conductivity and also reveals intrinsic anisotropies associated with the nanostructure. Ground-state degeneracy in AgSbTe2 leads to the natural formation of nanoscale domains with different orderings on the cation sublattice, and correlated atomic displacements, which efficiently scatter phonons. This mechanism is general and suggests a new avenue for the nanoscale engineering of materials to achieve low thermal conductivities for efficient thermoelectric converters and phase-change memory devices.

8:25

3aPAb2. Inelastic scattering and resonance ultrasound spectroscopy for functional materials studies. Raphael P. Hermann (Juelich Ctr. for Neutron Sci. JCNS and Peter Gruenberg Institut PGI, JARA-FIT, Forschungszentrum Juelich GmbH, Leo Brand Str. 1, Juelich 52425, Germany, r.hermann@fz-juelich.de)

The combined use of inelastic scattering, resonant ultrasound spectroscopy, and other macroscopic thermodynamics characterization techniques will be presented. In particular, inelastic neutron scattering and nuclear inelastic scattering (NIS) by Mössbauer resonant nuclei are two techniques that probe acoustic phonons and provide a microscopic counterpoint to direct speed of sound measurements. Results from new developments for the Sb and Te element specific NIS with sub-meV resolution will be presented for materials with thermoelectric or phase change properties. Specifically the lattice softening in the YbFe4Sb12 skutterudite [Moechel *et al*., Phys. Rev. B **84**, 184306 (2011)] and the systematic softening observed in nanostructured thermoelectric materials [Claudio *et al*., J. Mater. Sci. **48**, 2836 (2013)] with respect to their bulk counterpart will be discussed, as well as lattice softening in magnetocaloric MnFe4Si3. [The European Synchrotron Radiation Facility is acknowledged for provision of the synchrotron radiation facility at beamlines ID18 and ID22N; the Institut Laue Langevin for beamtime at IN6; the Deutsche Forschungsgemeinschaft for funding SPP-1386 ”Nanostructured Thermoelectrics” and SFB-907 ”Nanoswitches"; the BMBF for NanoKoCh 03X3540; and the Helmholtz Gemeinschaft Deutscher Forschungszentren for VH-NG-407 and HRJRG-402.]

8:45

3aPAb3. A long-sought phase transition in superconducting cuprates observed via resonant ultrasound spectroscopy. Albert Migliori, Arkady Shekhter, Brad J. Ramshaw, and Ross D. McDonald (NSEC-NHMFL, Los Alamos National Lab., M.S. E536, Los Alamos, NM 87545, migliori@lanl.gov)

Among the biggest mysteries of high-temperature superconductors is the so-called pseudogap—somewhat similar to the gap in the electronic density of states found in the superconducting phase, but occurring at a different temperature. The pseudogap may represent either the gradual onset of a precursor to superconductivity or an entirely new phase, characterized by the gain or loss of some hidden order. Several experiments in recent years have favored the latter, but the smoking gun, the thermodynamic signature of a pseudogap phase transition, had not been observed. Using resonant ultrasound spectroscopy, we measured the temperature-dependent elastic stiffness of two cuprate superconducting crystals, one underdoped and one overdoped and found a break in slope at a doping-dependent temperature T*. For the underdoped cuprate, T* coincides with the onset of the pseudogap and with earlier neutron-scattering measurements of the appearance of magnetic order (blue squares). Crucially, for the overdoped cuprate, $T^* < T_c$ so that extrapolating to higher doping where $T^* = 0$ will yield a quantum critical point, which may be key to understanding the mechanism of high-temperature superconductivity (Nature **498**, 75 (2013).

9:05

3aPAb4. Hypersound in simple one-dimensional device structures. G. Todd Andrews (Phys. and Physical Oceanogr., Memorial Univ., Prince Philip Dr., St. John’s, NF A1B 3X7, Canada, tandrews@mun.ca)

Brillouin spectroscopy, an inelastic laser light scattering technique capable of probing long wavelength acoustic phonons in a variety of material systems, was used to study hypersound in simple one-dimensional mesoporous silicon-based device structures formed using electrochemical etching methods. Brillouin spectra of porous silicon superlattices with binary periodicity on the order of the hypersound wavelength reveal zone folding, band gaps, and localized modes, indicating that these structures behave as hypersonic phononic crystals. Superlattices with smaller modulation wavelengths act as effective elastic media. New results on the behavior of hypersound in stacked superlattices and those with deliberately introduced defects will also be presented. Collectively, these studies have led to an improved fundamental understanding of classical wave behavior and interaction in low-dimensional systems and open up exciting opportunities for phonon engineering in a silicon-based platform.

Contributed Papers

9:25

3aPAb5. High temperature elastic constants of rare-earth doped $Sr_{0.9}X_{0.1}TiO_{3-\delta}$ (X = Pr, Y). Josh R. Gladden, Sumudu P. Tennakoon (Phys. & NCPA, Univ. of MS, NCPA, 1 Coliseum Dr., University, MS 38677, sptennak@go.olemiss.edu), Rasheed Adebisi (SOAIR, LLC, University, MS), Qin Zhang (Phys. & NCPA, Univ. of MS, University, MS), A. M. Dehkordi (Mater. Sci. and Eng., Clemson Univ., Clemson, SC), S. Bhattacharya, T. M. Tritt (Phys. and Astronomy, Clemson Univ., Clemson, SC), and H. N. Alshareef (Mater. Sci. and Eng., King Abdullah Univ. of Sci. and Technol., Thuwal, Saudi Arabia)

Temperature dependence of the elastic constants of polycrystalline rare-earth doped strontium titanate (STO) [$Sr_{0.9}X_{0.1}TiO_{3-\delta}$ (X = Pr, Y)] was investigated in the temperature range of 300 K—750 K using resonant ultrasound spectroscopy. Elastic constants of undoped STO decrease linearly indicating typical softening with increased temperature. Yttrium (Y) doped STO also exhibits a monotonic softening, however, with a pronounced curvature in this high temperature regime. Trends of elastic constants of the praseodymium (Pr) doped STO show a non-monotonic stiffing from room temperature to 475 K, followed by a gradual softening. Changes in attenuation were quantified by the inverse quality factor (1/Q) averaged over measured resonances. Undoped STO showed a monotonic gradual increase of attenuation with increasing temperature while yttrium doped STO showed little variation. In contrast, attenuation of Pr doped STO exhibited a peak around 425 K. These results will be compared to thermal conductivity measurements in the same temperature range and phonon scattering mechanisms will be discussed.

9:40

3aPAb6. Capacitive micromachined ultrasonic transducers as tunable phononic crystals. Shane Lani (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 1454 Catherine St., Decatur, GA 30030, swl5059@gmail.com), M. Wasequr Rashid (School of Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), Karim G. Sabra, and F. Levent Degertekin (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Capacitive micromachined ultrasonic transducer (CMUT) arrays are made up of microscale (10-100μm wide) membranes with embedded electrodes for electrostatic excitation and detection of acoustic waves. While typically used for far-field imaging, CMUT arrays also support dispersive evanescent surface waves. These surface waves derive their dispersive properties not only from the periodic structure of the array, but also from the membrane resonance. One advantage of CMUTs as a metamaterial is that the dispersive qualities of the array can be tuned by changing the applied bias voltage to the membranes, which in effect changes the membrane stiffness. A second advantage is that the CMUT array elements can be used as receivers to record the acoustic waves with high spatial resolution, which make laser displacement measurement based characterization unnecessary. These properties allow the possibility of CMUTs to exploit these slowly propagating evanescent waves as a means for creating subwavelength resolution fields for high-resolution ultrasound imaging and sensing in the near field by appropriately tuning the physical characteristics of individual membranes. The dispersive behavior of these evanescent surface waves propagating along a CMUT array was quantified using a computationally efficient, boundary element method based model and validated with both finite element analysis and experimental data obtained from a 1 x 16 CMUT array with a membrane resonance tunable between 5 and 6.5 MHz.

WEDNESDAY MORNING, 4 DECEMBER 2013 — POWELL, 8:00 A.M. TO 9:40 A.M.

Session 3aSAa

Structural Acoustics and Vibration, Noise, and ASA Committee on Standards: Groundborne/Structureborne Noise and Vibration from Transportation

James T. Nelson, Chair
Wilson, Ihrig & Associates, 6001 Shellmound, Emeryville, CA 94608

Chair's Introduction—8:00

Invited Papers

8:05

3aSAa1. Noise reduction performance of wheel vibration absorbers. James T. Nelson (Wilson, Ihrig & Associates, 6001 Shellmound, Emeryville, CA 94608, jnelson@wiai.com)

The noise reduction effectiveness of wheel vibration absorbers was evaluated at the Bay Area Rapid Transit System (BART). The vibration absorbers were fitted to the steel tires of aluminum-centered wheels. Wayside noise was recorded at tangent and curved track with resilient direct fixation fasteners, and under-car noise data were recorded throughout the system. Data were collected with and without wheel vibration absorbers installed. The wheel vibration absorbers had little effect on wayside and under-car noise at audio frequencies, though a minor reduction of noise at about 500 Hz was observed, suggesting a change in wheel dynamics. Third octave band noise reductions are summarized for on-board and wayside measurements.

8:25

3aSAa2. Noise reduction performance of wheel and rail vibration absorbers. Thom Bergen (Wilson, Ihrig & Associates, Inc., 15719 165th Pl. NE, Woodinville, WA 98072, tbergen@wiai.com) and James T. Nelson (Wilson, Ihrig & Associates, Inc., Emeryville, CA)

The rolling noise reduction effectiveness of wheel and rail vibration absorbers was evaluated at Trimet in Portland, Oregon. Tests included tangent ballast and concrete tie track and curved track slab track with bi-block concrete ties. Under-car noise data were recorded throughout the system with and without wheel vibration absorbers. Results of the rolling noise tests are summarized for various combinations of treated and untreated rails and wheels. The wheel and especially the rail vibration absorbers significantly reduced rail vibration at audio frequencies, but had little effect on wayside rolling noise. Wayside noise at tangent track was slightly higher with rail vibration absorbers than without. However, the "singing rail" noise was eliminated entirely. The "singing rail" vertical vibration transmission spectrum had a pass-band characteristic as expected for periodic supports. The vibration data indicate that above 250 Hz the wheel was the dominant source of noise relative to the rail. A reduction of stick-slip noise was observed over most curves, though results were inconsistent.

8:45

3aSAa3. Groundborne noise produced by rail transit tunnel construction. Thom Bergen (Wilson, Ihrig & Associates, Inc., 15719 165th Pl. NE, Woodinville, WA 98072, tbergen@wiai.com), James T. Nelson, and Derek L. Watry (Wilson, Ihrig & Associates, Inc., Emeryville, CA)

Groundborne noise and vibration produced by rail transit tunnel construction activities was studied recently in Seattle, Washington. Recent and ongoing tunneling is accomplished with tunnel boring machines (TBMs) and associated supply trains. Vibration produced by construction activities in the tunnels 30 to 40 m below the surface propagated efficiently to through the soils in the Seattle area. The local geology consists largely of "overconsolidated glacial till" that is very stiff. The study revealed that the supply trains traveling over rail joints was the dominant source of ground vibration, and the re-radiated groundborne noise in numerous homes above the tunnels was audible. The noise was effectively mitigated by patching and smoothing out the rail joints, and supporting the ties on natural rubber pads.

9:05

3aSAa4. Line source response from geotechnical data. Gary Glickman (Wilson, Ihrig & Associates, 65 Broadway Ste. 401, New York, NY 10006, gglickman@wiai.com)

Predicting groundborne noise and vibration environmental impacts associated with rail transit projects involves determination of the line source response (LSR) to characterize ground vibration propagation characteristics. The presentation discusses the use of geotechnical data to develop model LSRs. GIS is then used to apply model LSR data on a larger scale and perform a detailed analysis using Federal Transit Administration (FTA) methodology, which can be refined with field testing at a later stage. Conceptual track vibration mitigation measures are discussed for controlling groundborne noise and vibration to nearby receptors.

Contributed Paper

9:25

3aSAa5. Measurement of dynamic viscoelastic properties of flexible polyurethane foam under compression for application to seat vibration analysis. Deokman Kim, Won-Sok Yoon (School of Mech. Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133-791, Korea, Seoul ASI | KR | KS013, South Korea, deokman@hanyang.ac.kr), Hyun-kyu Park (Res. & Development Div., HYUNDAI·KIA MOTORS, Seoul, South Korea), Hak-Sung Kim, and Junhong Park (School of Mech. Eng., Hanyang Univ., Seoul, South Korea)

Supporting stiffness of the seat is one of the important components affecting dynamic characteristics recognized by a passenger. To analyze dynamic characteristic of seat for vehicles operating on various road conditions, the seat should be understood together with the oscillation due to road irregularity. In this study, the viscoelastic properties of flexible polyurethane foam under compression was measured and used in estimating the dynamic characteristic of seat analyzed as a simplified geometry. The beam transfer function method is used to obtain the dynamic properties of the foam under compression. The viscoelastic properties were obtained to the maximum compression level of 70%. The simple seat model was composed rigid base, edge blocks, elastic supports, and flexible polyurethane foam, and the method is used in the same way to obtain the dynamic support properties. The equivalent support stiffness was estimated on various locations on the seat, and the effects of each component such as the compression level, the foam type, and the stretch of elastic support were analyzed.

Session 3aSAb

Structural Acoustics and Vibration, ASA Committee on Standards, and Engineering Acoustics: Structural Health Monitoring I

Tribikram Kundu, Cochair
Civil Eng. & Eng. Mech., Univ. of Arizona, 1209 E. 2nd St., Bldg. # 72, Tucson, AZ 85721

Wolfgang Grill, Cochair
ASI Analog Speed Instruments GmbH, Burgweg 8, Koenigstein im Taunus 61462, Germany

Chair's Introduction—10:00

Invited Papers

10:05

3aSAb1. Analysis of acoustic harmonic generation in a solid with multiple nonlinear interfaces. Shiro Biwa and Yosuke Ishii (Dept. of Aeronautics and Astronautics, Kyoto Univ., C-Cluster III, Katsura, Nishikyo-ku, Kyoto 615-8540, Japan, biwa@kuaero.kyoto-u.ac.jp)

Weak bonds and delaminations are typical examples of imperfect interfaces in multilayered structures. Nonlinear acoustic/ultrasonic methods are expected to offer a promising means to monitor these imperfect interfaces, as such interfaces behave nonlinearly when subjected to high-amplitude waves and result in the occurrence of nonlinear frequency components such as higher harmonics. Harmonic generation at a single nonlinear interface has been studied by many investigators from both theoretical and experimental points of view. In this presentation, a theoretical analysis of harmonic generation at multiple nonlinear interfaces is presented within a framework of one-dimensional elastic wave propagation in the frequency domain. The analysis is based on the perturbation expansion of the wave field by assuming the weak nonlinearity. Specifically, the second-harmonic generation is analyzed by first solving the linear transmission of the incident fundamental component, and then the propagation of the second-harmonic components generated at nonlinear interfaces. Some numerical results are demonstrated and compared to the results of time-domain analysis using the finite element method. The present analysis shows that harmonic generation in multilayered solids is remarkably frequency-dependent, as both the fundamental and the harmonic components interact with the layered structure in a complex manner.

10:25

3aSAb2. Monitoring material nonlinearity and attenuation variations in mortar subjected to freezing-thawing cycles. Jesus N. Eiras (Instituto de Ciencia y Tecnología del Hormigón (ICITECH), Universitat Politècnica de València, Valencia, Spain), Tribikram Kundu (Civil Eng. & Eng. Mech., Univ. of Arizona, 1209 E. 2nd St., Bldg. # 72, Tucson, AZ 85721, tkundu@email.arizona.edu), John S. Popovics (Civil and Environ. Eng., Univ. of Illinois, Urbana, IL), J. Monzó, M. V. Borrachero, and J. Payá (Instituto de Ciencia y Tecnología del Hormigón (ICITECH), Universitat Politècnica de València, Valencia, Spain)

Standard vibration resonance frequency tests have been widely used for prediction of material modulus of elasticity and for monitoring damage in cement-based materials. More recently, dynamic non-classical nonlinear analyses show promise for damage diagnosis through a variety of test methods that are generally called nonlinear elastic wave spectroscopy (NEWS) techniques. In this study, we monitor the nonlinear dynamic behavior and mechanical wave attenuation of mortar subjected to varying numbers of freezing-thawing cycles. The nonlinear analysis is deployed using a new signal processing technique applied to standard resonance frequency test generated data. The proposed technique is demonstrated on damaged and un-damaged mortar bar samples.

10:45

3aSAb3. Nonlinear ultrasonic waves for monitoring thermal stresses in solids. Claudio Nucera and Francesco Lanza di Scalea (Structural Eng., Univ. of California San Diego, 9500 Gilman Dr., MC 0085, La Jolla, CA 92093, flanza@ucsd.edu)

It is known that nonlinear ultrasonic waves in solids are sensitive to quasi-static stresses. The stress sensitivity of elastic waves is typically associated to finite strains (e.g., theory of acoustoelasticity). In the case of waveguides, classical nonlinear theories for guided waves are still based on the assumption of finite strains. In the case of constrained solids subjected to thermal excursions, however, there are theoretically no finite strains (for perfectly constrained solids) associated with thermal stresses. A new model is therefore needed to justify the existence of wave nonlinearities in this case of stress without strain. This problem is solved on the basis of the interatomic potential of the solid that indicates a "residual" strain energy, due to the prevented thermal expansion, which is at least cubic as a function of strain. Consequently, a nonlinear wave equation can be derived. The solution to this equation leads to a new nonlinear parameter for double harmonic generation that is directly related to the thermal stresses in the structure. This study finds applications in the monitoring of thermal stresses in buckling-prone structures, such as continuously welded railroad tracks and pipelines. Experimental tests conducted on railroad tracks with realistic support will be also presented.

11:05

3aSAb4. Noncontact fatigue crack detection using nonlinear wave modulation spectroscopy. Peipei Liu, Hoon Sohn (Dept. of Civil and Environ. Eng., KAIST, Deajeon, South Korea,, hoonsohn@kaist.ac.kr), and Tribikram Kundu (Dept. of Civil Eng. and Eng. Mech., Univ. of Arizona, Tucson, AZ)

Nonlinear wave modulation spectroscopy (NWMS) has been used to evaluate nonlinear acoustic signature of fatigue cracks in materials and thus to get an idea about the degree of material nonlinearity. It is done by generating ultrasonic waves at two different frequencies and measuring their modulation. The choice of two distinct frequencies plays a significant role in NWMS for different structures. In this paper, instead of using signals at two distinct frequencies, only one broadband pulse signal is used as the driving signal, which can be generated by a laser beam. This driving signal generates multi-frequency responses as different resonance frequency modes and/or Lamb wave modes, generated in a plate-like structure. Nonlinear wave modulation occurs among these frequencies when material nonlinearity exists. It increases the sideband energy and the number of peaks in the spectral plots. These two features, namely sideband energy ratio (SER) and sideband peak number (SPN), are extracted from the spectral plots to measure the material nonlinearity caused by fatigue cracks. The noncontact laser system has been built for NWMS measurement by integrating and synchronizing a Q-switched Nd:YAG laser for ultrasonic wave generation and a laser Doppler vibrometer for ultrasonic wave detection. The proposed modified NWMS technique with the noncontact laser system has been successfully used for the identification of metallic plates with fatigue cracks.

11:25

3aSAb5. "Incubation of damage" state quantification in laminated composites and metallic alloys. Sourav Banerjee (Dept. of Mech. Eng., Univ. of South Carolina, 300 Main St., Rm. A117, Columbia, SC 29208, banerjes@cec.sc.edu)

In this presentation, a comparatively simple but efficient novel approach is proposed to quantify the "incubation of damage" state using scanning acoustic microscope (SAM). The proposed approach exploits the nonlocal micromorphic field theory to quantify intrinsic (multi-scale) damage state. Defying the conventional route of 'bottom-up' multi-scale modeling methods, a hybrid 'top-down' approach is presented, which is then correlated to ultrasonic signature obtained from composite and metallic alloy specimens. A parameter to quantify the incubation of damage at meso-scale has been identified in this paper. The intrinsic length scale dependent 'parameter called 'damage entropy' closely resembles the material state due to fatigue, extreme environments, operational hazards or spatio-temporal variability, etc. The proposed quantification process involves fusion between micromorphic physics and high frequency ultrasonic. The proposed approach is validated through an experimental study conducted on sequentially fatigued glass-fiber reinforced polymer composites and Aluminum 5xxx aluminum alloy specimens. Specimens were characterized under scanning acoustic microscope (50 and 100 MHz). The imaging data and the sensor signals are characterized to quantify the incubation of damage state by a new parameter called 'damage entropy.'

WEDNESDAY MORNING, 4 DECEMBER 2013 PLAZA B, 8:30 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication: Acquiring Speech: Children and Adults

Catherine L. Rogers, Chair

Communication Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620

Contributed Papers

8:30

3aSC1. Younger versus older infants' use of prosody-like boundaries to locate musical phrases. Kara E. Hawthorne (Linguist, Univ. of AB, 2-40 Assiniboia Hall, Univ. of Alberta, Edmonton, AB T6G 2E7, Canada, kara.hawthorne@gmail.com) and LouAnn Gerken (Linguist, Univ. of Arizona, Tucson, AZ)

In spoken language, particularly of the infant-directed variety, clauses are marked with prosodic cues, such as final lengthening and pitch resets at boundaries (e.g., Soderstrom *et al*., 2008). Though prosody is specific to language, similar acoustic cues mark phrase boundaries in music, and infants are sensitive to the correlation of these cues at musical boundaries (Jusczyk and Krumhansl, 1993). In the present study, we ask whether younger (4-month-old) and older (16-month-old) infants can use prosody-like boundary cues to facilitate recognition of musical phrases. In a musical extension of Soderstrom *et al*. (2005), infants were familiarized with one of two brief melodies, then were tested using the head-turn preference procedure on their ability to recognize a phrase versus phrase-straddling excerpt from the familiarization melody when it is embedded in a new musical passage. Preliminary results suggest that younger infants show a familiarity preference for the test melody containing a phrase from the familiarized melody, while older infants do not discriminate between the test item types. This suggests that young infants use prosody-like cues to group acoustic stimuli in a domain-general way, whereas older infants may not.

8:45

3aSC2. Assessing language acquisition from parent-child interaction: An event-related potential study on perception of audio-visual cues in infancy. Eeva Klintfors, Lisa Gustavsson, Iris-Corinna Schwarz, Tove Gerholm, and Ulrika Marklund (Linguistics/Phonet., Stockholm Univ., Universitetsv 10 C, Stockholm SE – 106 91, Sweden, eevak@ling.su.se)

This paper promotes a theory-driven model development of parent-child interaction. In our project, we identify, test, and simulate some of the fundamental components of speech, gestures, and social-emotional behaviors and the consequences they might have on child language development. Our theoretical position is part of the connectionist tradition; language acquisition is described to be an emergent consequence of the interplay between the infant and the ambient linguistic environment, including sensory information of all modalities. It is well known that speech comprehension and production are significantly influenced by the presence of co-speech gestures. These gestures may be articulatory in nature or hand/beat co-gestures that keep the rhythm of speech. However, since the extent of this integrated relationship is difficult to determine from behavioral research solely, studies addressing neural mechanisms that underlie cognitive processes and behaviors are of importance. This paper reports an electroencephalography/event-related potential (EEG/ERP) pilot study on children's early perception of congruent versus incongruent audio-visual pairings (e.g., acoustic information matching vs not matching the articulation shown). Ultimately, it is our hope that understanding the integrated speech-gesture relationship may provide insights into how children allocate resources while speaking and help clinicians/teachers to better identify and treat children with developmental disorders.

9:00

3aSC3. Assessing language acquisition from parent-child interaction: An event-related potential study on perception of intonation contours in infancy. Lisa Gustavsson, Eeva Klintfors, Iris-Corinna Schwarz, Tove Gerholm, and Ulrika Marklund (Linguistics/Phonet., Stockholm Univ., Universitetsv 10 C, Stockholm SE-106 91, Sweden, lisag@ling.su.se)

The aim of this paper is to present our multidisciplinary project to study parent-child interaction. The goal of the project is to identify, test, and simulate components of child and adult speech and gestures and the consequences they might have on child language acquisition. Since typical parent-child interaction is built upon both interlocutors' intention-reading, responsiveness to joint-attention, and imitation of speech/gestures, we make video recordings along with recordings of speech data to grasp the integration of semantic and pragmatic aspects of language acquisition. The understanding of parent-child interaction benefits further from information on brain activation involved in speech processing. As a first step to achieve the project goals, an electroencephalography/event-related potential (EEG/ERP) study exploring children's early perception of intonation contours involved in human interactions was performed. This paper discusses the characteristics of integration of multimodal social-emotional (speech, prosody, faces, posture) signals as part of the dynamics of communication in typically developing children. Possible application fields are social signal processing (SSP; an emerging research domain that aims to provide computers ability to understand human social signals), and improvement of diagnosis of late or atypical language development in pathologies that affect the dynamics of social interaction (such as autism spectrum disorders).

9:15

3aSC4. Stop production in bilingual and second language-learning children. Sue Ann Lee (Texas Tech Univ Health Sci. Ctr., 3601 4th St., Lubbock, TX 79430, sueann.lee@ttuhsc.edu), Gregory Iverson (Univ. of Wisconsin, Milwaukee, WI), and Jahyung Lee (Ewha Womans Univ., Seoul, South Korea)

This study examined stops produced by 7 year-old Korean-English bilingual (KEB) children and age-equivalent Korean children who had learned English as a second language (L2) in order to investigate how duration of exposure affects the PHONETIC systems of their two languages. A total of 60 children participated (15 per group; monolingual English, monolingual Korean, KEB and L2 children). Word-initial VOT and f0 values in the following vowel were measured in both languages. Comparison of English and Korean stops produced by monolingual children showed that the two English (voiced and voiceless) and three Korean (fortis, lenis, & aspirated) stop types were fully distinguished. Like the monolinguals, KEB children produced English and Korean stops distinctively, indicating that they possess two separate stop systems. But while L2-learning children distinguished English voiced from Korean fortis, and English voiceless from Korean lenis, they produced English voiceless and Korean aspirated stops similarly. Compared to adult Korean L2 learners who did not distinguish English voiced from Korean fortis (Kang and Guion, 2006), the results here suggest that young L2 children express more sophisticated phonetic categories than do adult L2 learners. [Funded by NICHD (RHD061527A).]

9:30

3aSC5. The role of orthographic information in the learning of allophonic variation. Chung-Lin Yang (Linguist, Indiana Univ., 2100 E Lingelbach LN Apt. 503, Bloomington, IN 47408, cy1@indiana.edu) and Isabelle Darcy (Second Lang. Studies, Indiana Univ., Bloomington, IN)

Exposure to L2 orthography may facilitate learning a novel vocalic (e.g., Escudero *et al.*, 2008) or tonal (Showalter and Hayes-Harb, 2013) L2 contrast. Yet it is unclear whether the benefit of orthographic information applies to the learning of L2 words involving allophonic variants. We investigated whether exposure to L2 orthography can help L2 learners establish a single lexical representation for words containing allophones. We used an invented language, with word-pairs of free variants (test condition) involving the vowel alternation [ɔ]-[u], both of which can be spelled as <o>. In the control condition, vowel alternation [e]-[a] contrasted word meanings. In a word learning experiment, Mandarin and American English speakers were presented with words paired with pictures. In addition, one subgroup of participants saw the spellings when they heard the words, while another did not. Then, in a picture-auditory word matching task, participants who learned that the variants were allophonic were expected to link the two variants to one single picture in the test condition only, not in the control. A facilitative effect of orthography on the learning of free variation was observed for Mandarin speakers. This shows that orthography may help L2 learners establish a single lexical representation for allophonic variants.

9:45–10:00 General Discussion

10:00–10:30 Break

10:30

3aSC6. Difficulty in the acquisition of Mandarin high level and high falling tones by Cantonese learners. Xianghua Wu (Dept. of East Asian Lang. and Cultures, Univ. of California,, 3110 Dwinelle,Berkeley, CA 94720-2230, xianghwa.wu@gmail.com) and Kazuya Saito (School of Commerce, Waseda Univ., Shinjuku, Japan)

Native speakers of tone languages commonly have difficulty discriminating tones with the same phonological function (Huang, 2001), such as Cantonese high level and high falling tones; however, acquisition of such tones with distinctive phonological status in a second language (L2) remains unclear. This study tested 34 Cantonese learners before and after training on Mandarin high level, mid-rising, and high falling tones. Perception was evaluated using a forced-choice identification task, and nine native speakers of Mandarin judged productions from repetition and narrative tasks. Despite improvement in post-tests, perception of high level and high falling tones was found to be more difficult than mid-rising tone in both pre- and post-tests. Misidentification patterns also showed more confusion between high level and high falling tones than other tone pairings, but no effect of training was observed. Compared to the perception results, high falling tone was produced more frequently as high level tone, particularly in the narrative task before and after the training. The results suggest that L2 tone acquisition is complicated by the complex phonological correspondence between L1 and L2 tones. [Research supported by Language Learning Research Grant.]

10:45

3aSC7. Development of vowel spaces from age 21 to age 49 in a group of 8 talkers. Auburn Lutzross, William Schuerman, Ronald Sprouse, and Susanne Gahl (Linguist, Univ. of California Berkeley, 2435 Grant St., Apt. 2, Berkeley, CA 94703, alutzross@berkeley.edu)

We describe age-related change in speech during young to middle age adulthood using a new resource for phonetic and sociolinguistic analysis. This resource is based on the "Up" series of documentary films, showing a set of 11 individuals filmed at seven year intervals over a period of 42 years. We analyzed 67 sample utterances (minimum duration = 10 s), containing 4493 vowels produced by eight talkers, with the aim of understanding how vowel spaces change in young and middle-age adulthood, prior to physiological changes often observed in elderly talkers. We measured the first and second formants of each vowel token and analyzed several measures of vowel distribution in F1/F2 space: Euclidean distance from the talker's average F1/F2 ("dispersion"), within-category variability (intra-vowel dispersion), and vowel space area. Area of the vowel space was measured using averages of point vowels (/a/, /ae/, /i/, /u/), as well using the convex hull of all vowels. For some individuals, vowel spaces from age 21 to age 49 come to be more compact, in a manner that has previously been observed in elderly speakers. However, we also find considerable individual variability, with no clear age-related trend across speakers.

11:00

3aSC8. Non-native vowel production accuracy and variability in relation to overall intelligibility. Svetlin Dimov and Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208-4090, svetlindimov2011@u.northwestern.edu)

Previous research suggests that accuracy (i.e., distance to the average location of native productions) has less effect on adaption to non-native speech than category variability [e.g., Wade *et al*., Phonetica **64**, 122-144 (2007)]. Here we investigate the relationship between overall intelligibility of Mandarin-accented English for native English listeners and (a) vowel production accuracy, and (b) vowel production consistency. Intelligibility estimates were based on sentence-in-noise recognition accuracy scores. Vowel accuracy and consistency estimates were based on formant measurements of point vowels (/i/, /u/, /ae/, and /a/) extracted from words in the sentence materials that were presented to listeners for intelligibility testing (8-20 samples/vowel/talker). If listeners have expectations about a vowel category location based on accumulated exemplar storage, then greater accuracy (smaller Euclidean distance to native category mean) should be positively related to intelligibility. If listeners are sensitive to vowel category distributions, then greater consistency (smaller standard deviation of f1 or f2 within categories) should be beneficial to intelligibility. A mixed effects linear model revealed that only consistency was a significant predictor of intelligibility. Accuracy was not a significant predictor. This result suggests that intra-speaker variability is detrimental to L2 intelligibility, regardless of distance to native categories.

11:15

3aSC9. Individual differences in learning to perceive novel phonetic contrasts: How stable are they across time and paradigms? Mirjam Broersma (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, P.O. Box 9103, Nijmegen 6500 HD, Netherlands, m.broersma@let.ru.nl), Dan Dediu, and Jiyoun Choi (Max Planck Inst. for PsychoLinguist, Nijmegen, Netherlands)

Previous research has shown that learners differ widely in the success with which they learn to perceive novel phonetic contrasts. Little is known, however, about the stability of such differences over time and over paradigms. Are individuals who are good at learning to perceive novel speech sounds consistently good at it, or does the success of learning fluctuate over time, or with the use of different paradigms? First, we investigate the stability of individual differences over time by assessing performance during five (pre- and post-training) test moments on three separate days with one-week intervals. Second, we investigate the stability over paradigms by comparing the two most commonly used tests of speech sound perception, namely discrimination and identification. 70 native speakers of Dutch participated in a series of training and test sessions, during which they were trained to perceive the Korean three-way lenis-fortis-aspirated contrasts /p-p*-ph/, /t-t*-th/, and /k-k*-kh/, which are difficult for them to distinguish. Results showed, first, that individual differences were very stable over time. Second, the correlation between individuals' discrimination and identification scores was only moderate. Thus, individual differences in learning to perceive novel phonetic contrasts seems to be a stable individual trait over time, but not over paradigms.

11:30

3aSC10. The effect of sleep on learned sensitivity to a non-native phonetic contrast. Sayako Earle and Emily Myers (Univ. of Connecticut, 123 Davis Rd., Storrs, CT 06268, frances.earle@uconn.edu)

Consolidation during sleep is thought to play a role in integrating newly learned words into the preexisting lexicon (e.g., Dumay and Gaskell, 2007), while the effect is in stabilizing degraded information against decay when learning to map synthesized speech onto native phonology (Fenn, Nusbaum, Margoliash, 2003). In the current study, we investigated the effects of overnight consolidation on discrimination between new (nonnative) phonetic categories. Fifty-four monolingual English speakers were trained to categorize tokens from a non-native dental-retroflex contrast. Half of the participants were trained in the evening, and the other half were trained in the morning. Discrimination ability was tested 8-14 h post-training and 22-26 h post-training in order that the effects of intervening sleep (evening group) or daytime activity (morning group) could be assessed. Discrimination in the trained vowel context improved after the overnight between-session interval in the night group, but declined slightly in the morning group. Both training groups improved in discrimination ability for an untrained vowel context immediately following the overnight between-session interval, but not before. Results suggest that memory consolidation during sleep, proactive interference during native language exposure, or both play a role in non-native phonetic learning.

11:45–12:00 General Discussion

Session 3aSP

Signal Processing in Acoustics: Acoustical Communications and Sound Source Localization

James C. Preisig, Chair
WHOI, MS #11, Woods Hole, MA 02540

Contributed Papers

8:30

3aSP1. Energy efficient transmission policies in non-stationary underwater acoustic channels. Beatrice Tomasi and James C. Preisig (AOP&E, Woods Hole Oceanogr. Inst., 266 Woods Hole Rd, Woods Hole, MA 02543, btomasi@whoi.edu)

This work focuses on making underwater acoustic communications energy-efficient, by reducing the amount of unsuccessful transmissions. The approach is enabled by *a priori* information regarding the second order statistics of the channel quality. However, the non-stationarity of the physical processes that primarily influence the acoustic propagation makes both channel representation and identification challenging. Therefore, we first evaluate the different types of second order statistics of the underwater acoustic channel measured during two experiments, SPACE08 and KAM11, during which the same source and receiver hardware was employed in different environmental conditions. Then, we classify the different observed second order statistics estimated over a few minutes time intervals and propose suitably trained Markov models to represent the evolution of these different second order statistics. This channel representation is used to derive an optimal transmission scheduling that minimizes the number of transmissions required to deliver a given amount of information B by a given deadline T. In particular, we provide insights on how the structure of the optimal policy changes with the different observed behaviors of the second order statistics of the channel.

8:45

3aSP2. Adaptive orthogonal frequency division multiplexing underwater acoustic communications with limited feedback. Xiaopeng Huang (Stevens Inst. of Technol., Castle Point on Hudson, Hoboken, NJ 07030, xhuang3@stevens.edu), Aijun Song (Univ. of Delaware, Newark, DE), Walid Ahmed (Stevens Inst. of Technol., Hoboken, NJ), Moshen Badiey (Univ. of Delaware, Newark, DE), and Victor Lawrence (Stevens Inst. of Technol., Hoboken, NJ)

In orthogonal frequency division multiplexing (OFDM) underwater acoustic (UWA) communications, some subcarriers may be subject to deep fading. If the channel state information (CSI) is available at the transmitter side, adaptive transmission techniques (e.g., power allocation) can be applied to mitigate the selective fading effect and increase the overall performance. Therefore, it is more valuable to analyze the UWA channel with limited CSI feedback. In this paper, we adopt two time-varying shallow water acoustic channels (slow-varying environment and fast-varying environment) as examples. Lloyd algorithm is employed to quantize the CSI at the receiver and construct the codebook, which is also known to the transmitter. Simulation results compare the performance between two different channels, and the performance between a few bits of feedback, perfect feedback, and non-feedback, respectively.

9:00

3aSP3. Smart sonic detection and ranging for blind sources localization and separation. Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, sean_wu@wayne.edu) and Na Zhu (Dept. of Eng. Technol., Austin Peay State Univ., Clarksville, TN)

A new methodology for blind sources localization and separation in arbitrary three-dimensional space is presented. The underlying principle this methodology is the discrete short-time sonic detection and ranging (SODAR) [J. Acoust. Soc. Am. **133**(6), 4054—4064 (2013)] that searches sound sources over discrete frequency-time regions in a systematic and automatic manner. For simplicity yet without loss of generality, only the dominant sound source within each discrete frequency-time region is considered. Once the search is completed, the frequency contents and time instances that correspond to the sound sources identified at the same coordinates are strung together and played back one by one. In this way, one can use N number of sensors to locate and separate S sources, where S can be much larger than N. Numerical simulations and experimental validations of using this methodology for locating and separating arbitrarily time-dependent sound sources in arbitrary three-dimensional space are demonstrated. Advantages and limitations of this methodology are discussed as well.

9:15

3aSP4. Information-theoretic quantification of underwater acoustic source localization performance. Thomas J. Hayward (Naval Res. Lab., 4555 Overlook Ave SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Metrics that historically have been applied to quantify the performance of signal processing for source localization are algorithm-dependent. For example, performance of conventional beamforming or matched-field processing is usually quantified by main-peak width and secondary-peak levels of the beam response or spatial ambiguity function, while performance of Bayesian localization may be quantified by measures of the statistical dispersion of the *a posteriori* pdf of source location. While algorithm-dependent performance metrics permit comparisons within a given class of signal processing algorithms, they do not provide comparability across algorithm classes. The present work identifies fundamental information-theoretic quantities that can be used as metrics to quantify the source localization performance of diverse signal processing algorithms and thus provide for performance comparisons across signal-processor classes. These quantities include conditional entropy of source location given processor output, mutual information of source location and processor output, and cross-entropy of actual and posterior source-location probability distributions. Applications of these information-theoretic metrics are illustrated in examples of Bayesian localization, conventional beamforming, and matched-field processing of a time-harmonic source in a range-independent shallow-water acoustic waveguide. The results are interpreted in the light of the data processing inequality of information theory. [Work supported by ONR.]

9:30

3aSP5. Application of information-theoretic performance measures to optimization of array spatial configurations for underwater acoustic source localization. Thomas J. Hayward (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Information-theoretic measures of acoustic source localization performance provide for performance comparisons that are valid across classes of signal processing algorithms and can be used as performance criteria for the optimization of receiver-array spatial configurations for source localization. This work investigates the use of fundamental information-theoretic quantities, including mutual information of source location and processor output and conditional entropy of source location given processor output, as performance criteria for the optimization of array configurations. Applications of these criteria to the optimization of horizontal and vertical arrays are illustrated in examples of Bayesian localization, conventional beamforming, and matched-field processing of the acoustic field of a time-harmonic source in a range-independent shallow-water waveguide. The optimized array spatial configurations are compared with results obtained using traditional (energy-based) performance measures. [Work supported by ONR.]

9:45

3aSP6. A relative-entropy approach to distributed passive detection. Peter C. Mignerey (Acoust. Div., Naval Res. Lab., Peter Mignerey Code 7160, Washington, DC 20375-5350, peter.mignerey@nrl.navy.mil)

There is currently much interest within the ocean acoustics community on using distributed sensor networks to monitor ocean properties. One such task is the application of distributed sensors to passive detection of weak acoustic sources. The joint likelihood detection ratio for a set of distributed sensors leads naturally to the comparison of relative entropy with a detection threshold. As a nondimensional additive measure of information, relative entropy enables data fusion among disparate kinds of sensors, e.g., acoustic and electromagnetic, and mitigates calibration issues. Furthermore, because relative entropy is an integral over probability densities of sensor outputs, it is insensitive to false alarms from transients. In this talk, the theory of relative-entropy detection will be presented, and the method illustrated using acoustic intensity data from hydrophones deployed by the Transverse Acoustic Variability Experiment (TAVEX). For the method to work, accurate estimates of the probability densities for noise and signal intensities must be obtained. For the TAVEX data, superior receiver operating characteristic curves are obtained when the noise and signal distributions are represented by log-normal distributions in comparison with gamma and nonparametric distributions. [Work supported by the Office of Naval Research.]

10:00

3aSP7. Position estimation of rotating sound source using Kalman filtering based on time difference of arrival measurements. Jaehyung Lee, Young-Ju Go, and Jong-Soo Choi (Aerosp. Eng., Chungnam National Univ., Yusunggu Gungdong 220, Daejeon 305-764, South Korea, aerojhl@cnu.ac.kr)

In this work, we are interested in tracking a rotating sound source using a Kalman filtering technique based on a set of non-linear time difference of arrival (TDOA) measurements. Array of microphones measure acoustic signal emitted from a rotating source and the TDOA estimates are calculated followed by a solution for hyperbolic position fix. The position estimation of sound source based on TDOA is a popular technique in source localization. The method involves calculation of a set of nonlinear equations and poor accuracy of TDOA estimates often results in inaccuracy in location. In this work, the range difference is expressed by a model movement on which a recursive extended Kalman Filter has been developed. The TDOA measurements optimize the estimated values which are reduced as observation in extended Kalman filtering algorithm. Location estimation is updated from TDOA measurements along with the time history data. The Cramer-Rao Lower Bound (CRLB) is derived and simulations are compared. [Work supported by National Research Foundation of Korea (NRF) grant funded by the Korea government (MEST) (No. 2010-0014978).]

WEDNESDAY MORNING, 4 DECEMBER 2013 PLAZA A, 8:30 A.M. TO 12:00 NOON

Session 3aUW

Underwater Acoustics: Contributed Papers in Underwater Acoustics (Poster Session)

Megan S. Ballard, Chair
Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758

Contributed Papers

All posters will be on display from 8:30 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 a.m. to 10:15 a.m. and contributors of even-numbered papers will be at their posters from 10:15 a.m. to 12:00 noon.

3aUW1. Acoustic radiation force. Part I: Finite element modeling for elastic objects. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com) and Ivars Kirsteins (NUWC, Newport, Newport, RI)

The study of the acoustic radiation force produced by acoustic beams has been the topic of active research in the last few years mainly due to its ability for particle trapping and non-contact manipulations. However, efforts in modeling the radiation force have mainly focused on analytic solutions and thus have been limited to spherical objects, where the acoustic radiation force has been calculated for a plane wave as well as various types of beams. But up until recently, even those efforts were limited to using on-axis beams, where the incident beam is along an axis that goes through the center of the sphere and thus reduces the problem to an axially symmetric one. In this work we use the finite element technique to compute the acoustic radiation force for an arbitrary elastic object and for an arbitrary incident beam. One of the main objectives of this work is to understand and interpret the data we collected at a recent experiment we conducted at the Washington State University test tank facility, where a 2.5-in. PMMA sphere was ensonified by an intense amplitude-modulated ultrasound beam focused at its surface.

3aUW2. Acoustic radiation force. Part II: Eigenmode excitation of a scaled target. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com) and Ivars Kirsteins (NUWC, Newport, Newport, RI)

We describe an experiment that we recently performed at the Washington State University test tank where dynamic acoustic radiation forces were used to excite an eigenmode of a 2.5 in. PMMA sphere in water. The dynamic acoustic radiation force was generated by an intense, amplitude modulated ultrasound signal focused on the surface of the sphere whose amplitude modulation frequency was set to generate acoustic radiation forces modulated at a rate matched to an eigenmode of the sphere. A conventional hydrophone was used to listen to the sphere's acoustic emissions from that particular eigenmode. Unlike conventional acoustic insonification, the dynamic acoustic pressure force is generated by momentum transfer, which creates a mechanical excitation on the object's surface at a rate equal to twice the modulation frequency, analogous to a hammer striking it. In the experiment, the modulated ultrasound beam was scanned horizontally across the sphere using a computer-controlled actuator with the sphere's acoustic emissions measured at each position. To confirm and understand the experimental results, the measured sphere's acoustic emissions were compared to finite element model predictions as a function of horizontal position.

3aUW3. Test of a towed line array suitable for geoacoustic inversion. Joel Abdullah, Neil Woodson, Jason D. Sagers, David P. Knobles, Steven A. Stotts, and Thomas Muir (Appl. Res. Labs., Univ. of Texas, ARL:UT PO BOX 8029, Austin, TX 78713-8029, joela@arlut.utexas.edu)

Small line arrays towed by ships may give a unique, cost effective approach for acoustic reconnaissance of seabed characteristics in littoral seas. A 16 element, 1 m spaced, towed line array was designed and developed at ARL:UT for the purpose of ship-towed statistical inference of the seabed. The towed array was tested at the ARL:UT Lake Travis Test Station under a variety of conditions to study array performance. Additional non-acoustic sensors provided information about array shape, stability, and drag as a function of speed, depth, and tail termination. Acoustic data were collected during the test and processed with an adaptive beamformer yielding high SNR, well-localized signals from a fixed 500 Hz source as well as the received signals from small boats of opportunity. The acoustic data are analyzed with a statistical inference approach to estimate the geoacoustic properties of the lake bottom. [Work supported by the ARL:UT internal research and development program.]

3aUW4. Three-dimensional propagation: Comparison of finite element and coupled-mode solutions. Megan S. Ballard, Benjamin M. Goldsberry, and Marcia J. Isakson (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

Three-dimensional propagation over an infinitely long cosine-shaped hill is studied using finite element and coupled-mode models. The finite element approach is based on a longitudinally invariant solution technique. The solution is formulated in a Cartesian coordinate system and a cosine transform is applied to eliminate the range-independent dimension. The resulting equation is two dimensional and the solution is calculated for a sufficient range of values of the transform variable. Then the spatial solution is obtained using an inverse cosine transform. The coupled-mode model is formulated in a cylindrical coordinate system, and the solution is obtained using a separation of variables. Modal amplitudes are calculated from the horizontally separated part of the Helmholtz equation using a hybrid technique such that a parabolic solution provides the description of horizontal refraction in the azimuthal direction and a stepwise coupled-mode technique accounts for mode-coupling in the radial direction. The finite element model provides a highly accurate result, limited only by the discretization of the environment and sampling of the cosine transform integral. The coupled-mode solution is approximate, but an examination of the model amplitudes is used to gain insight into the effects of environmental inhomogeneities on the acoustic field. [Work supported by ONR.]

3aUW5. Ultrasonic measurements of suspended sediment concentrations at Harris Bayou, Mississippi. Wayne O. Carpenter (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38677, wocarpen@olemiss.edu), Thomas A. Kajdan (Civil Eng., Univ. of MS, Oxford, MS), Bradley T. Goodwiller (Mech. Eng., Univ. of MS, University, MS), Cristiane Q. Surbeck (Civil Eng., Univ. of MS, University, MS), James P. Chambers (Mech. Eng., Univ. of MS, University, MS), Daniel G. Wren, and Roger A. Kuhnle (National Sedimentation Lab., USDA-Agricultural Res. Service, Oxford, MS)

The use of ultrasonic acoustic technology to measure the concentration of fine suspended sediments has the potential to greatly increase the temporal and spatial resolution of sediment measurements while reducing the need for personnel to be present at gauging stations during storm events. In collaboration with the USGS, a customized field deployable system was installed to monitor fine sediment particles, less than 100 micron in diameter, in suspension at Harris Bayou near Alligator, MS. Calibration measurements show good agreement between laboratory grade equipment and the new prototype system. The field unit consists of two immersion ultrasonic transducers measuring attenuation of 20 MHz acoustic signals propagated through suspended particles. The results of field prototype will be presented here.

3aUW6. Numerical study of the spatial and temporal variability of ambient noise in the coastal region east of Taiwan Strait. Andrea Y. Chang, Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, yychang@ntu.edu.tw), Sheng-Fong Lin (Green Energy and Environment Res. Lab., Industrial Technol. Res. Inst., HsinChu, Taiwan), and Ruey-Chang Wei (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

Noise soundscape represents the characteristics and spatial distributions of the ambient noise level under various noise source mechanisms. This is a significant index while describing an underwater acoustic environment, especially in the subjects related to marine mammal protection. Under the effects of topography, sediment, and oceanographic features, underwater soundscape varies with time and space. This paper focuses on estimating mean soundscape of wind driven noise and shipping noise and their spatial and temporal variability in the coastal region east of Taiwan Strait, which is the main habitat of the Sousa Chinensis. The ambient noise is studied numerically and local wind field and shipping density observed by Automatic identification Systems (AIS) are applied to generate noise source field. As for the ocean environment, the time varying/spatial dependent temperature profiles generated by the Taiwan Coastal Ocean Nowcast/Forecast System (TCONFS), which formulated on the basis of the Princeton Ocean Model, is used for water column variability, and both topography database and geo-acoustic database are used to describe the bottom. The modeling results demonstrate the temporal/spatial variability induced by ocean environment, manifested by measured data. [This work was supported by National Science Council of Taiwan and Bureau of Energy (Grant No.102-D0105).]

3aUW7. Prototype development of underwater noise impact alert region prediction system. Yu-Chen Cheng, Andrea Yuan-Ying Chang, Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No.1, Sec 4, Roosevelt Rd., Taipei 10617, Taiwan, r01525053@ntu.edu.tw), and Sheng-Fong Lin (Inst. of Green Energy and Environment Technologies, Industrial Technol. Res. Inst., Hsinchu, Taiwan)

Offshore wind farms are the main project in western Taiwan. Since the underwater noise generated by piling poses a threat to the marine mammals, the issue of detrimental impact of noise on *Sousa Chinensis* is proposed. To avoid causing behavioral disturbance and injury from pile driving noise, Underwater Noise Impact Alert Region Prediction System (UNIARPS) was established to estimate the acoustic field at any depth and distance from piling source. The system can be illustrated as four components, environment databases, acoustic propagation model, source modeling, and noise level prediction. The ocean numerical model (TCONFS) generates time spatial dependent temperature profiles for water column variability, and geo-acoustic and bathymetry databases are imported as environmental inputs.

Adiabatic mode theory is used to simulate the piling noise propagating in shallow water and the impulsive noise emanating from source is evaluated via finite element method. While the auditory threshold of cetacean set as criterion level, the system can demonstrate the modeling outputs and predict noise impact region, and these results are useful to prior planning on how to station the guarding boats in preventing dolphins entering the noise impact region. [The financial support provided by Bureau of Energy (Grant No.102-D0105) is gratefully acknowledged.]

3aUW8. Acoustic mode coupling due to subaqueous sand dunes in the South China Sea: Extension of the adiabatic criterion to waveguides with bedforms. Linus Chiu (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., No. 70, Lienhai Rd., Kaohsiung 80424, Taiwan, linus@mail.nsysu.edu.tw) and Davis B. Reeder (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

The large subaqueous sand dunes on the upper continental slope of the South China Sea (SCS) create a range-dependent ocean acoustic waveguide within which acoustic energy is expected to couple between propagating normal modes. Here, the criterion of adiabatic invariance is extended to the case of a waveguide possessing bedforms. The morphological features of the bedforms modeled in this theoretical and numerical investigation are based on echosounder observations of the SCS sand dune field during a research cruise in the spring of 2012 on the Taiwanese R/V Ocean Researcher 2 (OR2). Using the extended criterion for adiabatic invariance to examine mode propagation over these bedforms, results demonstrate that bedforms increase mode coupling strength such that the criterion for adiabatic propagation is exceeded for waveguides with small bedform amplitude to water depth ratios; increasing bedform amplitude enhances mode coupling. Physically, initially bottom-trapped mode 1 energy abruptly couples to higher adjacent modes, with most of the energy preferentially settling into a few select modes. The scattered energy fills the water column to near-surface depths downrange of the bedforms. Numerical simulations confirm the extended criterion parameterization. [This work was supported by National Science Council of Taiwan.]

3aUW9. Properties of the Umov vector in shallow water and its dependence on sea surface conditions. David R. Dall'Osto and Peter H. Dahl (Mech. Eng., UW-Seattle, 914 N 38th St., Seattle, WA 98103, dallosto@u.washington.edu)

In this work, the effects of a rough sea surface on shallow water acoustic propagation are examined using experimental data collected during the ONR sponsored Target and Reverberation Experiment (TREX) off of the coast of Panama City, Florida, in May 2013. The acoustic data were collected from a bottom deployed recording tower that coherently recorded data on a horizontal line array (HLA), a vertical line array (VLA), and on an accelerometer-based vector sensor which was combined with a co-located hydrophone to formulate the Umov vector, or instantaneous intensity vector. The source was lowered from the stern of a research vessel to a depth one-third and two-thirds of the water depth (18 m), and transmitted a multi-frequency pulse from 1 to 4 kHz. These measurements were repeated at positions approximately 10, 20, and 40 water depths away from the tower, along a bearing perpendicular and parallel to the surface wave-crests. During the experiment, the sea surface directional-wave spectrum was measured by a Datawell Waverider buoy moored at the experimental site. Properties of the Umov vector are shown to relate to roughness and directional characteristics of the sea surface. The Umov vector is also studied in relation to the HLA and VLA measurements.

3aUW10. Depth-tracking of a near-surface target from the deep ocean. Sheida Danesh and Henrik Schmidt (Massachusetts Inst. of Technol., 143 Albany St., Cambridge, MA 02139, sdanesh@mit.edu)

Determining the depth of an acoustic source in a deep ocean environment can be approached using a variety of existing methods, each with inherent limitations. A method for determining the depth of a moving near-surface acoustic target from a fixed vertical array below the deep ocean critical depth is presented using the characteristics of the Lloyd mirror pattern of a near-surface acoustic signal and a library of calculated patterns. Depth estimates of the moving target are made in real-time and incorporated into a confidence metric for tracking the target motion. Results indicate that this method is robust, performing well in conditions involving environmental mismatch and a moderate amount of surface noise.

3aUW11. Evaluation of complex broadband biomimetic waveforms for active sonar. Peter Dobbins (Future Systems, Ultra Electronics Sonar Systems, Leanne House, Avon Close, Weymouth, Dorset DT4 9UX, United Kingdom, peter.dobbins@ultra-sonar.com)

There is an expanding requirement to reduce the impact of man-made sound, including active sonar transmissions, on marine mammals in the defence, offshore, and other sectors. One way this might be achieved in sonar applications is to use signals derived from natural sounds such as the vocalizations of the animals themselves. It might be expected that such sounds would appear more familiar, thus reducing possible abnormal behavioral impacts. This paper reviews the use of such waveforms and presents the results from a trial designed to compare the detection capabilities of a variety of broadband signals, both conventional and 'novel', with a medium frequency active sonar. Two biomimetic signals were tested, one based on sperm whale echolocation clicks and the other on pilot whale whistles. Preliminary analysis suggests the detection performance of these signals using conventional matched filters is comparable with linear FM chirps with a similar bandwidth, but may be improved with detection techniques commonly used for marine mammal vocalizations, such as spectrogram correlation. The paper will conclude with an assessment of the potential impacts of such signals on marine life.

3aUW12. Underwater navigation using an acoustic spiral wave front beacon. Benjamin Dzikowicz (Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil) and Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A spiral wave front beacon consists of an array of transducers which produce a signal whose phase depends on the azimuthal angle at which it is received and a reference signal with constant phase [J. Acoust. Soc. Am. **131**, 3748 (2012)]. A vehicle can determine aspect to the beacon by comparing the phase of the two signals. Progress in the development of this navigation technique will be discussed including results from experiments at Dodge Pond in Connecticut where an unmanned surface vehicle determined its aspect to within 10° by receiving signals from the beacon. Also, tests of a new spiral beacon design [J. Acoust. Soc. Am. **130**, 2506 (2011)] in laboratory and underwater environments will be presented. Underwater experiments are performed at the Navy's Seneca Lake facility in upstate New York using an UUV to record signals. Overall, these results demonstrate that intrinsic phase shifts in the beacon can be handled by signal processing at the receiving vehicle and that the spiral navigation technique is robust in reverberant environments. [Work supported by the Office of Naval Research.]

3aUW13. Information capacity of an acoustic field in a Pekeris waveguide with an absorbing bottom and spatially correlated surface noise field. Steven I. Finette and Earl Williams (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

It is well known that the Shannon theory of information sets an asymptotic bound on the maximal rate of transmission of information through a channel with negligible probability of error. This rate, known as the information capacity, can be computed analytically or numerically and numerous results for the capacity in terrestrial communications have been derived. Most analytic results, however, involve free space propagation while in underwater acoustics, the propagation region is spatially bounded. In this presentation, we consider the information capacity in an ocean waveguide comprised of a uniform sound speed in the water column, a penetrable, absorbing bottom and a correlated surface noise field described by the Kuperman-Ingenito source sheet model. An expression for the information capacity of the acoustic field in the water column exterior to a distributed source region is determined by first applying singular value decomposition to the Green's function matrix to obtain independent communication

channels and then using Lagrange multipliers to solve a multiply constrained optimization problem involving the mutual information between source and receive regions. [Work supported by the Office of Naval Research.]

3aUW14. Supervised machine learning for estimation of rough bottom anisotropy direction using bistatic acoustic scattered fields. Erin M. Fischell (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu) and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

An issue when trying to use scattered acoustic fields to classify underwater targets is the strong directional scattering due to anisotropic rough bottom structure that occurs in the 1-5 kHz frequency range. Autonomous Underwater Vehicles (AUVs) are uniquely suited to exploit this three dimensional field with the goal of estimating the anisotropy direction of the bottom roughness. Estimation of the angle of bottom roughness ridges relative to an acoustic source is carried out using a combination of supervised machine learning and AUV behaviors. Anisotropic Goff-Jordan power spectrum bottom scattered fields in 15 degree angle increments are generated using SCATT and OASES acoustic packages. The amplitudes of these fields are sampled into sets of 5-20 AUV waypoints. Support Vector Machine (SVM) regression is used to train a model, and an independent test data set is used to evaluate the validity of the model. A confidence model is constructed and critical waypoints identified using the test set results. The confidence and SVM models can then be used to determine bottom roughness angle in real time, as demonstrated using the LAMSS MOOS-IvP simulation environment.

3aUW15. Generation and propagation of oceanic T-waves using elastic parabolic equation solutions. Scott D. Frank (Mathematics, Marist College, 3399 North Ave.,Poughkeepsie, NY 12601, scott.frank@marist.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and Robert I. Odom (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Oceanic T-waves are essential for location and identification of seismic sources since they travel long distances in the ocean and are typically the largest signals received at hydrophone arrays or coastal monitoring stations. T-waves either link directly into the SOFAR channel by conversion of elastic wave energy at a downward sloping interface between the elastic and fluid media, or are generated when elastic energy couples into low order acoustic modes due to bathymetric inhomogeneities. Elastic parabolic equation solutions will demonstrate generation and long range propagation of oceanic T-waves in the water column when the source is located in an elastic ocean bottom. Elastic parabolic equation solutions will be used to describe effects of ocean bottom parameters on transmission characteristics of a sloping boundary. The impact of small-scale bathymetric changes, for example due to a rough ocean bottom, will be characterized by averaging acoustic wavenumber spectra resulting from multiple bottom realizations. Favorable characteristics for T-wave generation will be determined. The impact of large scale bathymetry changes, such as a seamount or underwater ridge, will also be discussed. [Work supported by ONR.]

3aUW16. Information transfer of broadband sonar echoes. Charles F. Gaumond (Acoust. Div., US Naval Res. Lab., CODE 7162, 4555 Overlook Ave. SW, Washington, DC 20375, charles.gaumond@nrl.navy.mil)

The ability to compute information transfer of a broadband signal corrupted with additive noise is straightforward, but the case of signal corrupted by convolution with a stochastic, propagation impulse-response is more complicated. The known, integral solution is presented along with numerical methods for estimating differential entropy and mutual information using kernel density estimators. Numerical examples using simulations of echo and propagation responses are shown. Results are shown using echo data from Clutter09, an ocean experiment performed with an echo repeater. These echoes were generated with a known, modeled target response. The mutual information between the modeled target response and the echoes, which propagated through ocean environments, is also shown. These results are discussed with respect to developing a method of obtaining an information sonar equation that could be used to estimate the sonar parameters required to perform an information based task, such as signal classification. [Research funded by the Office of Naval Research.]

3aUW17. Clutter statistics of long-range wideband echoes from fish aggregations off the Oregon coast. Roger C. Gauss, Joseph M. Fialkowski (Acoust. Div., Naval Res. Lab., Code 7164, 4555 Overlook Ave., S.W., Washington, DC 20375-5350, roger.gauss@nrl.navy.mil), and Richard H. Love (BayouAcoust., Abita Springs, LA)

Echoes from fish can be the dominant source of reverberation over a range of important sonar frequencies and grazing angles. Moreover, fish echoes from broadband signals often retain coherent structure (generate clutter) after undergoing normalized match-filter processing. Coupled with their inherent spatiotemporal variability, fish can thus be a significant clutter problem for active sonars. Using a towed source and horizontal line-array receiver, measurements of mid-frequency (1.5-11 kHz) backscattering from aggregations of fish were made from the R/V New Horizon in five shallow-water and shelf-break areas off the coast of Oregon (Astoria Canyon to Heceta Bank) during July and August 2012. The experiment and the frequency-dependent echo statistics in relation to the observed distribution and behavior of the two primary resident fish species (Pacific hake and Pacific sardines) are discussed. For example, the short-time echo variability and spatial patchiness were characteristic of the mid-water (hake) and near-surface (sardine) fish observed concurrently on echosounder displays. Furthermore, the clutter's probability density functions were found to be non-Rayleigh but well modeled by NRL's Poisson-Rayleigh clutter model that provides a physical context for relating data distributions to scatterer attributes. [Work supported by the Office of Naval Research.]

3aUW18. Interpulse noise assessment during shallow water seismic survey using airgun excitations. Shane Guan (National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Joseph F. Vignola, and Teresa J. Ryan (Dept. of Mech. Eng., Catholic Univ. of America, Washington, DC)

Offshore energy exploration and geophysical research activities using seismic airgun arrays are known to generate intense underwater sound. Such seismic imaging has the potential to impact marine mammals through hearing impairment and behavioral modification. Few studies have investigated multipath propagation and reverberation from these airgun impulses. These phenomena could increase the duty cycle within the sound field sufficiently to impact long distance communication or result in detrimental acoustic masking for marine mammals. We report initial findings on the elevation of the sound level between airgun impulses during a shallow, open-water seismic survey. This work uses continuous recordings collected from three bottom-mounted hydrophones deployed in the Beaufort Sea in summer 2012. A quantitative method is used to examine the root-mean-squared noise levels between seismic impulses and the noise level dependence on source range. Preliminary results show that ambient noise increases above non-impulsive harassment levels defined by the National Marine Fisheries Service (120 dB re: 1 microPa) for a portion of the time between seismic impulses at intermediate ranges from the source. In addition, the duration of reverberation is related to the source range, with significantly longer decay times measured on hydrophones at greater distances from the source.

3aUW19. Dual-channel orthogonal modulation differential pattern time delay shift coding underwater acoustic communication method. Xiao Han, Jingwei Yin, Xiao Zhang, and Chi Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Bldg., Nantong St., Nangang District, Harbin, Harbin 150001, China, hanxiao1322@hrbeu.edu.cn)

Information is carried by the time delay between adjacent code elements in differential pattern time delay shift coding system. It has an ability of anti inter-symbol interference and the Doppler effects. In order to obtain higher communication rate and reduce the interference between channels, this paper proposes a dual-channel orthogonal modulation differential pattern time delay shift coding underwater acoustic communication method and selects the balance Gold sequence as Patterns. At the transmitter end, divide input bits into two channels to differential pattern time delay shift encode and modulate

orthogonally. Then add the encoded signal of two channels together to transmit. At the receiver end, demodulate the received signal orthogonally, search the correlation peak position of patterns. And estimate time delay to restore the original input information. Dual parallel channel transmission mode effectively improves communication rate of differential Pattern time delay shift coding and the orthogonal modulation method greatly reduces the interference between channels. Simulation research is carried on for the method, at transmission data $2x10^{4bit}$ and SNR -5dB, information is recovered at a communication rate 205 bit/s with very low bit error rate.

3aUW20. Propagation effects of surface waves in two unperturbed mode models. Frank S. Henyey and Eric I. Thorsos (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, frank@apl.uw.edu)

Two unperturbed mode simulation models for acoustic propagation with a wavy surface have been developed. The first, LIN, is the first order in the surface elevation for a fixed number of modes. It is the model that provides the starting point for deriving transport equations, as the mode coupling spectra needed for transport theory are practical to evaluate. The second model, DAE, is more accurate for a larger number of modes, but the mode coupling spectra are very difficult to evaluate. The results are compared for an example propagation environment, and consequences to the accuracy of the transport theory are discussed.

3aUW21. Three-dimensional primary acoustic field characterization for a seismic airgun array. Arslan M. Tashmukhambetov, George E. Ioup, Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA), Natalia A. Sidorovskaia (Dept. of Phys., Univ of Louisiana at Lafayette, Lafayette, LA), Joal J. Newcomb (Naval Oceanogr. Office, Stennis Space Ctr., MS), James M. Stephens, Grayson H. Rayborn (Phys. and Astronomy, Univ. of Southern Mississippi, Hattiesburg, MS), and Phil Summerfield (Geodetics & Cartography, ExxonMobil Corp., UIT, Houston, TX)

The Littoral Acoustic Demonstration Center conducted the Source Characterization Study in 2007 (SCS07) to measure the 3-D acoustic field of a seismic airgun array in the Gulf of Mexico. Three moorings with sensitive and desensitized hydrophones at different depths were deployed as well as hydrophones suspended from a ship, while a seismic source vessel shot specified lines. Hydrophone positions were measured. Peak pressures, RMS sound pressure levels (SPL), sound exposure levels, total shot energy spectra, one-third octave band analyses, and source directivity studies are used to characterize the field. Summary results are first calculated for each hydrophone. These are then combined to give isopleths for azimuthal cuts at 0, 45, 90 degrees, etc., for the spatial domain measures. Plots for each solid angle bin give these and frequency measures analyzed versus range. Zero-to-peak pressures directly under the array go from 210 dB for depths less than 200 m down to 195 dB at 1200 m. At 2000 m horizontal range the pressures go from 160 dB near the surface to 175 to 180 dB at 1200 m. RMS SPL is about 5 dB smaller. [Research supported by the Joint Industry Programme through the International Association of Oil and Gas Producers.]

3aUW22. Multi-static scattering characteristics of submerged objects with experimental investigation. Yoon Hee Ji, Gi Hoon Byun, Jea Soo Kim (Ocean Eng., Korea Maritime Univ., Dong-Sam dong, YoungDo Gu, Busan 606-791, South Korea, 1002wine@hanmail.net), Ho Seuk Bae, and Woo Shik Kim (Agency for Defense Development, Changwon, South Korea)

The scattering characteristics of target echoes are essential for detecting and classifying the submerged objects. The target strength, which is widely used in mono-static sonar system, is also important in multi-static sonar system to identify the submerged target. In this presentation, a series of experiments in the acoustic water tank was conducted to measure the target echoes from submerged cylinder-shaped target with multi-static measurement system, which consists of a single transmitter and 16 receivers. The target strengths are presented in 2-dimensional plane as a function of receiver position according to target aspect angle. The numerical simulation results based on Kirchhoff approximation are presented to explain some characteristics of the measured multi-static target echoes. [Work supported by Agency for Defense Development, Republic of Korea.]

3aUW23. Fluctuations of arriving narrowband signal's direction in horizontal plane in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru) and Valery Grigorev (Phys., Voronezh Univ., Voronezh, Russian Federation)

Directions of amplitude (envelope) and phase fronts in horizontal plane of signals with some width of spectrum coming to receiving array in shallow water are studied. As an example experiment Shallow Water 2006 is analyzed, where LFM signals of the frequency 300 ± 30 Hz were used for acoustical sounding on the distance ~20 km. Visible length of horizontal part of L-shaped array was ~200 m. It was shown that fluctuations of direction of phase and amplitude fronts took place with angle between them about $2.5^\circ\pm1.5^\circ$ were registered. Results are interpreted as manifestation of frequency dependence of horizontal refraction initiating variation of horizontal angle of coming rays from pulse to pulse during time of observation. [Work was supported by RFBR and BSF.]

3aUW24. Estimation of level-crossing rate of signals reflected by ocean surface based on rough surface scattering theory. Joonsuk Kim (Dept. of Elec. and Electron. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Rm. 381B Yonsei Eng. Res. Park, Seoul 120-749, South Korea, kimjs1st@yonsei.ac.kr), Il-Suek Koh (Dept. of Electron. Eng., Inha Univ., Incheon, South Korea), and Yongshik Lee (Dept. of Elec. and Electron. Eng., Yonsei Univ., Seoul, South Korea)

A method is proposed for theoretical estimation of the level-crossing rate of the underwater acoustic communication signals that are reflected from the ocean surface. The variation in the reflection coefficient of the moving ocean surface causes the intensities of the received signals to fluctuate. In this work, the reflection coefficient is obtained by modeling the time-varying characteristics of the ocean surface based on rough surface scattering theory. The surface of the ocean is modeled by Gaussian processes that are characterized by ocean spectra such as the Pierson-Moskowitz and Durden-Vesecky spectra. Furthermore, Gaussian random functions with a particular correlation time are incorporated to model the surface that continuously varies over time. Then the standard Periodogram analysis is applied to estimate the autocorrelation function. Finally, the level-crossing rate is calculated with the negative curvature of the autocorrelation function. For verification, comparison of the simulated results with the measured data is provided.

3aUW25. Effects of wideband source types on accuracy of wideband path-loss prediction based on finite-difference time-domain scheme. Yongjune Kim (Elec. and Electron. Eng., Yonsei Univ., C133, Eng. Bldg. C, 134, Shinchon-dong, Sudaemoon-ku, Seoul 120-749, South Korea, sum57@yonsei.ac.kr), Il-Suek Koh (Electron. Eng., Inha Univ., Incheon, South Korea), and Yongshik Lee (Elec. and Electron. Eng., Yonsei Univ., Seoul, South Korea)

The finite-difference time-domain (FDTD) scheme is a well-known numerical algorithm that can solve a wideband response of a wave equation in time domain. When the FDTD scheme is applied to the problem of the underwater path-loss, however, the wideband source excitation may generate various problems at low frequency band. For instance, when a source is implemented with a large magnitude near the zero frequency, a spurious response can be generated in the FDTD simulation due to the DC component that cannot propagate. On the other hand, when a source is implemented with very small magnitude at low frequencies, the numerical accuracy cannot be guaranteed. In this work, implementation of a new wideband source for FDTD scheme is proposed that is suitable for response over a wide bandwidth, including very low frequencies. The Tukey window is applied to a wideband Gaussian pulse in the frequency domain, which eliminates the DC component effectively. By utilizing the steep slope of a Tukey window, higher accuracy is achieved in the low frequencies. To verify the proposed wideband source, the path-loss results based on the FDTD scheme are compared with the famous normal mode solution, KRAKEN, as well as the ray solution, BELLHOP.

3aUW26. A new type flextensional transducer. Yu Lan, Wei Lu, Yongjie Sang, and Kuan Li (Harbin Eng. Univ., Nantong St. No.145, Hei Longjiang Province, Harbin, Harbin 0086, China, lanyu_2013@126.com)

In a field of an oceanographic survey, low frequency sound wave is often used because of low attenuation and good propagation characteristic in water. It is well known that flextensional transducer is a typical low frequency underwater source for oceanographic research, utilizing flexural vibration to realize low frequency radiation with small size. However the size of traditional flextensional transducer becomes relative large when the frequency decreases to a few hundred Hertz, and it is still a difficult problem to solve. A new type of class IV flextensional transducer was proposed, and original piezoelectric ceramic stacks were replaced by three groups of small class IV flextensional transducers, resulting in frequency decrease and volume displacement expansion. Make use of ANSYS finite element software, the new class IV transducer was modeled and analyzed. The experimental data showed the frequency of this new transducer design obviously decreased compared with traditional IV flextensional transducer at the same size. Key words: Flextensional transducer; low frequency; small size; Finite Element Method

3aUW27. Sensitivity of the underwater sound field in submarine canyons to water column variability. Ying-Tsong Lin, Weifeng Gordon Zhang, and Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Bigelow 213, M.S.#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

The ocean dynamics in the geologically and morphologically complex submarine canyons can have strong spatial and temporal variability due to the presence of internal tides/waves and upwelling currents. Our fundamental research question is what the acoustic effects of these oceanographic processes are in such environments where the sound propagation is also strongly influenced by the seafloor complexity. A simple example showing the joint effects is the bottom reflection of sound that has complicated patterns depending on the shape of canyon seafloor, seabed properties, and acoustic incident angles. Among these factors, the incident angle is the link connecting the acoustic effects of marine geology and physical oceanography. Specifically, the ocean dynamics changes the water column stratification and thus the incident angle of sound onto the seafloor, which explicitly determines the reflection of sound from the complex canyon seafloor. More involved examples using integrated regional ocean and full-field sound propagation models will be shown in the talk, and sensitivity analysis of underwater sound propagation along and across canyons will be performed. [Work supported by the Office of Naval Research.]

3aUW28. Computational modeling of acoustic wavefronts propagating in an underwater environment with uncertain parameters. Sheri Martinelli (202 Rochambeau Ave., 1176 Howell St., Newport, RI 02841, sheri_martinelli@alumni.brown.edu)

High frequency simulation of underwater sound propagation is a vital part of modeling and simulation of acoustic systems for evaluation and performance prediction. Existing simulations use deterministic ray tracing to propagate the acoustic field and simulate uncertainty by varying results according to basic distributions (e.g., adding "jitter" to ray arrival angle). Rather than incorporate randomness as a form of post-processing, this work seeks to model uncertainty where it exists in the underwater environment where it is easier to specify, and then propagate the relevant random quantities through the system applying stochastic collocation to an existing deterministic model. Further, this work addresses the drawbacks of ray tracing by taking the deterministic method to be a model that computes propagation of entire wavefronts rather than rays, thus maintaining error control over the physical domain. To this end, generalized polynomial chaos expansions are applied to a level-sets based wavefront propagation method to model the effects of uncertain parameters in an underwater environment. This approach allows for not only simple extraction of the process moments, but also yields an expression for the wavefronts in terms of random variables which can readily be simulated. [Work supported by ONR.]

3aUW29. Modeling the generation and propagation of hydrodynamic hull noise near the ocean surface. Rob Doyle (Explosion and Fluid Dynam., Martec Ltd., 5189 South St., Apt. 4, Halifax, NS B3J 1A2, Canada, rob.doyle@dal.ca), Mae Seto (Manned and Unmanned Systems for Mine Defence, DRDC Atlantic, Halifax, NS, Canada), and Julio Militzer (Mech. Eng., Dalhousie Univ., Halifax, NS, Canada)

Hydrodynamic hull noise is an important consideration for determining the detection envelope of SONAR domes mounted to surface vessels. In order to model the generation of this noise by a moving ship, a hybrid computational hydro-acoustic modeling methodology has been developed by combining the Lighthill-Curle acoustic analogy with the Numerical Wind Tunnel computational fluid dynamics code. This model has been shown in previous work to significantly over-predict the experimentally observed far field sound of Canadian Forces Auxiliary Vehicle Quest in at-sea acoustic trials. This deficiency is shown to be due in part to neglecting the Lloyd's Mirror interference effect of the sea surface in the Lighthill-Curle equations. By utilizing a method of images solution to the acoustic analogy to simulate the Lloyd's Mirror interference, an average sound pressure level improvement of 25 dB was obtained. This solution is compared and contrasted to a model utilizing the Lloyd's Mirror interference of a simple source, and a normal mode sound propagation model, and is shown to be superior for transmission ranges up to 2 km.

3aUW30. Near- and far-field simulations of coherent backscattering from scatterers in finite sized aggregations. Adaleena Mookerjee and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., 2010 Autolab, Ann Arbor, MI 48109, adaleena@umich.edu)

Classification of scatterers is a difficult but important step in active sonar applications. Active sonar signals in an ocean environment are scattered by surface roughness, and volume inhomogeneities in the bottom and water column. Fish schools may be important water-column clutter sources and, under some circumstances, may preferentially backscatter sound because of acoustic coherent backscattering enhancement (CBE). Here, the addition of in-phase scattered waves from propagation path pairs can readily explain a scattered intensity enhancement of a factor of two in the direction opposite to that of the incident wave. This presentation describes CBE simulations for finite sized aggregations of point scatterers using the Foldy (1945) equations that show much larger enhancements are possible in the far-field of the scattering aggregation. The simulations are validated in the near field with the theory from Akkermans *et al*. (1986), and with existing CBE optics and acoustics experiments from Wolf and Maret (1985) and Aubry *et al*. (2007). The dependence of the width of the CBE backscattered peak in the far-field is reported. Extension of these results to sonar pulse scattering from schools of fish is also very briefly discussed. [Work supported by the Office of Naval Research.]

3aUW31. Modeling of underwater noise from pile driving using coupled finite element and parabolic equation model with improved parabolic equation starting field. Jungyong Park, Woojae Seong, and Keunhwa Lee (Dept. Ocean Eng., Seoul National Univ., Seoul, South Korea, ioflizard@snu.ac.kr)

An offshore wind farm will be constructed in the Yellow Sea, west of Korean Peninsula, where there are extensive fishing activity and numerous fishery farms. To study the effect of underwater piling noise on fishing and marine lives, we model the pile driving noise propagation using coupled FE and PE model. The near-field noise is computed by FE model, considering detailed specifications of the pile driving system. We apply 2D axis-symmetric geometry and utilize acoustic structure interaction analysis in the frequency domain. The FE results are used to compose the starting field for PE model, where appropriate range selection is an important factor to cover most of the contributing ray paths. Extrapolation technique to compensate the lack of FE data and the numerical filtering method to smooth the FE result are discussed. In the far-field, the noise propagation is modeled by the split step Pade PE algorithm. The improved PE starting field seems to give refined result than previous coupled model.

3aUW32. Directionality of ambient noise measurements in Barrow Strait of the Canadian Arctic. Nicos Pelavas, Sean Pecknold, Carmen E. Lucas, and Garry J. Heard (DRDC Atlantic, 9 Grove St., Dartmouth, NS B3A 3C5, Canada, nicos.pelavas@drdc-rddc.gc.ca)

In August 2012, a field trial was carried out in Barrow Strait south of Gascoyne Inlet in the vicinity of 74.630 N 91.340 W. Underwater acoustic data was collected using a JASCO Autonomous Multichannel Acoustic Recorder (AMAR) and in-house designed sensor systems called Starfish Cubes. The Starfish Cubes were deployed twice, at different locations, each for one week duration and at depths of approximately 110 m. The Cubes consist of seven hydrophones with 1 m spacing and geometrically configured as three cross-dipoles with a central hydrophone, and have an operational frequency range of 5—750 Hz. During the trial 400 and 500 Hz tones were transmitted from discrete locations at various ranges. By using a beamforming method the tones were used to determine the orientation of the Starfish Cubes during their data collection periods. This enables investigation of the horizontal and vertical directionality of ambient noise. Unique localized sources contributing to the ambient noise are discussed such as a nearby grounded iceberg and a low frequency wandering tonal.

3aUW33. Scattering objects imaging using an autonomous underwater vehicle towing a source and an horizontal array. Samuel Pinson and Charles W. Holland (Penn State Univ., Appl. Sci. Bldg., Rm. 202a, State College, PA 16802, samuelpinson@yahoo.fr)

Recently, seabed sound-speed profile measurement by the image source method had been performed using an Autonomous Underwater Vehicle (AUV) towing a broadband source (frequency band from 1600 to 3500 Hz) and a linear array of hydrophones. This method provides an automatic process by the use of the semblance function. In that communication, the semblance ability to detect coherent reflections is used to image scattering objects such as mud volcanoes by integrating the results of successive measurement along the AUV track. Due to the horizontal array configuration, there is an ambiguity on the scatterers localization that could be solved by the use of hydrophone triplets.

3aUW34. Backscattering spectrum of a solid cylinder next to a horizontal surface when the cylinder's axis is not horizontal. Daniel Plotnick, Phillip L. Marston (Phys., Washington State Univ., 1510 NW Turner DR, Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com), Aubrey Espana, and Kevin L. Williams (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

When a solid cylinder lies proud on horizontal sand sediment there has been progress in understanding the backscattering spectrum as a function of grazing angle and the viewing angle relative to the cylinder's axis [Williams *et al*., J. Acoust Soc. Am. **127**, 3356-3371 (2010)]. The resulting evolution of the target strength spectrum is sometimes referred to as the "acoustic color" or the "acoustic template." For cylinders having identical ends and a transducer at a fixed grazing angle relative to the cylinder's center, viewing the cylinder over a 90 degree range is sufficient for characterizing the template. If the cylinder's axis has a vertical tilt such that one end is partially buried in the sand, then the symmetry of the template is altered and a 180 degree range is required. Some of the changes in the template can be approximately modeled using a combination of geometrical and physical acoustics. The resulting analysis gives a simple approximation relating certain changes in the template with the vertical tilt of the cylinder. A similar approximation also applies to a metallic cylinder adjacent to a flat free surface and was confirmed in tank experiments. [Work supported by ONR.]

3aUW35. Some results from the very shallow water TREX13 reverberation experiments using the Five Octave Research Array triplet module. John R. Preston (ARL, Pennsylvania State Univ., P. O. Box 30, M.S. 3510,State College, PA 16804, jrp7@arl.psu.edu)

A large experimental effort called TREX13 was conducted in April-May 2013 off Panama City, Florida. As part of this effort, reverberation and clutter measurements were taken in a fixed-fixed configuration in very shallow water (~20 m) over a 22 day period. Results are presented characterizing reverberation, clutter, and noise in the 1800-5000 Hz band. The received data are taken from the triplet sub-aperture of the Five Octave Research Array (FORA). The array was fixed 2 m off the sea floor and data were passed to a nearby moored ship (the R/V Sharp). An ITC 2015 source transducer was fixed 1.1 m off the seafloor nearby. Pulses comprised of gated CWs and LFMs were used in this study. Matched filtered polar plots of the reverberation and clutter are presented using the FORA triplet beamformer. There are clear indications of biologic scattering. Some of the nearby shipwrecks are clearly visible in the clutter, as are reflections from a DRDC air-filled hose. The noise data show a surprising amount of time-dependent anisotropy. Some model-data comparisons are made using the author's normal mode based reverberation model. Help from the Applied Physics Laboratory at the University of Washington was crucial to this effort. [Work supported by ONR code 322OA.]

3aUW36. Eigenspace dynamics of sample covariance matrices. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., A405, Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

Estimation of the sample covariance matrix is a challenge in array processing, particularly with large-aperture arrays operating in dynamic environments affected by fast maneuvering interferers and background noise. Minimizing the impact of time-dependent variations in the underlying signal statistics requires short observation intervals, thereby reducing the number of snapshots available for covariance estimation. Distinguishing between true variations in the received signal statistics and artifacts introduced by insufficient samples is still an open field of research. Recent developments in random matrix theory (RMT) have provided mathematical foundations to understand the behavior of sample eigenvalues and eigenvectors, and how they deviate from their population counterparts due to the lack of snapshot support. Similarly, expressions have been obtained to describe the "distance" between an initial eigenspace (spanned by p eigenvectors), relative to a subsequent eigenspace (spanned by q eigenvectors). In this paper, simulations corresponding to a horizontal array operating in a realistic environment are used to investigate RMT-based metrics that quantify time-dependent eigenspace stability. This research develops mathematically justifiable methods for proper data segmentation into intervals that exhibit local stationarity, providing data-driven higher bounds for the number of snapshots available for the computation of sample covariance matrices.

3aUW37. Passive ranging in strongly range-dependent environments: Effects of mode coupling on the waveguide invariant. Alexander W. Sell (Acoust., Penn State Univ., 830 Cricklewood Dr., Apt. 207, State College, PA 16803, aws164@psu.edu) and R. Lee Culver (Acoust., Penn State Univ., University Park, PA)

Prior work has shown that the value of the shallow water waveguide invariant changes in range-dependent environments due to non-uniform phase and group speed along the propagation path caused by either a range-dependent bathymetry or sound speed profile. Much of the work on these scenarios has dealt with weak range-dependence and the effects of mode coupling were neglected. In certain situations when mode coupling occurs, energy from higher order, surface-reflected-bottom-reflected modes may be lost to lower order, surface-refracted-bottom-reflected modes. These lower order modes, which do not interact with the surface, are associated with waveguide invariant values that differ greatly from the standard shallow water approximation where the waveguide invariant equals one. This talk will examine a case from the 2007 CALOPS experiment where the adiabatic approximation for modal propagation is no longer valid and mode coupling appears to be important. Acoustical data and analysis will be presented to demonstrate the effect that mode coupling has on the waveguide invariant in strongly range-dependent environments, and methods for incorporating coupled modes into waveguide invariant estimates will be discussed. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

3aUW38. Multistatic sound speed profile estimation. Hisashi Shiba (Radio Application Div., NEC Corp., 1-10, Nisshin-Cho, Fuchu, Tokyo 183-8501, Japan, h-shiba@aj.jp.nec.com)

Sonar is an indispensable component of harbor security systems. Sound propagation is one of the big problems for coverage estimations, since sound speed profiles are complicated under the complex environment like harbors. A new approach for sound speed profile estimation had been proposed using surface scattering by single sonar for frequent measurements which are required in the operation planning phase of high accuracy coverage evaluations. Although the configuration is simple, it consumes much time for higher accuracy estimations, because multiple angle transmissions need multiple waiting time and averaging considering surface fluctuations also need lots of time. This requirement is not favorable for quick sonar operations. One of the ideas reducing estimation time is using multiple sonars under multi-static configurations. This new concept uses multiple sonars, and it is suitable for harbor protections, since all sonars on the bottom do not become obstacles for ship navigations unlike tethered arrays used in acoustical tomography. And it can be said that multiple sonars should be deployed for covering wide and complex harbor areas with networks. Under this situation, a multi-static operation is a natural conclusion for harbor security. This new approach reduces the total estimation time. The rest problem is accelerating computing time of solving nonlinear simultaneous equations. A new acoustical structure model is under evaluation. The trial results are reported, if it is found to work well.

3aUW39. Measurements of the peak pressure and sound exposure level from underwater explosions. Alexander G. Soloway (Dept. of Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, soloway@u.washington.edu) and Peter H. Dahl (Dept. of Mech. Eng. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

There is an interest by the Navy to determine the sound field produced by underwater explosions to minimize the impact on marine life during training exercises. This work presents measurements of underwater explosions collected 7 km off the coast of Virginia in shallow water (depth 14 m) with sound speed conditions considered approximately iso-speed. Explosive charges with TNT equivalent weight 0.1 to 6.0 kg (W) were deployed at approximately mid-water and bottom depths. Acoustic data were recorded using a 9 element vertical line array at range 430 m and single-element autonomous systems at ranges 170, 430, and 950 m. The peak pressures and sound exposure levels (SEL) are calculated from the data; at 430 m peak pressures as high as 220 dB re 1 μPa and SEL as high as 190 dB re 1 μPa^2 s were measured. The peak pressures are compared to semi-empirical equations that are functions of range and W to the one-third power, such as Arons [J. Acoust. Soc. Am. **26**, 343-346 (1954)], and both the peak pressures and SEL are compared to simulations obtained using the parabolic wave equation. [Research supported by Naval Facilities Engineering Command.]

3aUW40. Surface wave shape inversion from forward scattered ocean acoustic data. Sean Walstead and Grant Deane (ECE/SIO, UCSD, 9500 Gilman Dr., 0407, La Jolla, CA 92093-0407, swalstead@ucsd.edu)

Prior work has shown that surface wave shape can be determined by analyzing underwater surface reflected acoustic signals in a wave tank. In this talk, forward scattered data from the Surface Processes and Communications Experiment (SPACE08) is analyzed with regard to surface wave shape inversion. Multipath arrivals representing surface, bottom-surface, and surface-bottom paths are distinguishable, implying that knowledge of the surface is known approximately 1/3, 1/2, and 2/3 the distance between source and receiver. Surface scattering losses including out of plane scattering and small scale roughness are numerically simulated and compared to actual ocean data. Methods are proposed for including these loss factors in a forward model of surface scattering that can be correlated with environmental conditions observed at the Martha's Vineyard Coastal Observatory.

3aUW41. Analysis of ambient noise in the habitat of Indo-Pacific humpback dolphin (*Sousa chinensis*) in the West Coast of Taiwan. Ruey-Chang Wei, Lian-Han Kuo (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), Jeff Chih-Hao Wu, and Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei, Taiwan, d98525001@ntu.edu.tw)

The west coast of Taiwan is one of the major habitats of Indo-Pacific humpback dolphin (*Sousa chinensis*). Ambient noise, changes with natural environment and human activities, is possible to affect the behaviors of marine mammals. Thus, it is necessary to conduct a long-term and systematic investigation of ambient noise in this area. This study deployed two underwater acoustic recorders (SM2M) in New Huwei River of the Yun-Lin coastal area (site YL) and Waisanding sandbar (site WS). 68- and 45-day acoustic data were collected in site YL and site WS. Results show that the low-frequency noise in site WS is lower than site YL due to the contributions of shipping or mechanical noises. In site YL, ambient noise of 1 to 2 kHz contains periodic changes because of the behaviors of croakers. Croakers usually appears before midnight in site YL, but no similar phenomenon is found in site WS. The frequency overlap between hearing range of marine mammal and high-level ambient noise is possible to cause the masking effect, even hearing loss. [Sponsored by the Forestry Bureau, Council of Agriculture, Taiwan under project "Population Ecology of Chinese White Dolphins and Ambient Noise Monitoring in its Habitat" No. 101-08-SB-14).]

3aUW42. The effects of source motion and bottom bathymetry on temporal coherence in shallow water propagation. Jennifer Wylie and Harry DeFerrari (Appl. Marine Phys., Univ. of Miami, RSMAS, 4600 Rickenbacker Cswy., Ops 11, Key Biscayne, FL 33432, jennie.wylie@gmail.com)

Previous studies on coherence have been focused on the effects of water column fluctuations on temporal coherence with a stationary source/receiver setup. However, with focus being turned to moving source setups there has been documented a significant drop in temporal coherence. With a moving platform, the propagation path will change based on relative source/ receiver position, and hence the bathymetry along the path will vary. Here, we will examine the effects that this bathymetric variation and related ship speed contribute to coherence loss. A range dependent parabolic equation model will be used to predict the temporal coherence for individual mode arrivals. A slowly varying random bottom will be introduced to the model and the coherence calculated for different ship speeds and for both radial and tangential tracks. Results will be compared with stationary source/receiver setups in order to determine at what ship speed/ bottom bathymetry does source motion become the driving factor in loss of coherence versus water column fluctuations from a stationary setup. Preliminary results indicate that at a speed of 2 knots, there is remarked loss of coherence at all modes except the first with even small variations in bottom bathymetry, which is in agreement with experimental results.

3aUW43. Research on phase generated carrier demodulation algorithm phase drift of fiber-optic hydrophone. Ge Yu and Jinshan Fu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Bldg. 145, Nantong St., Harbin, Harbin 150001, China, liz.221@163.com)

The paper outlines phase generated carrier (PGC) modulation and demodulation principles of interferometric fiber-optic hydrophone. For demodulation error caused by phase drift, the paper proposes a simple and practical method, using the cross-correlation between the measured signal and the two reference signals to lock the small signal of desired frequency and measure it. Two reference signals with a constant phase difference can directly output the phase of the measured signal. The paper realizes a real-time PGC demodulation system based on LABVIEW with sampling frequency 200 kHz and demodulates 400 Hz underwater acoustic signals. The experiment results show that this system can guarantee the interferometer to operate sensitively. The effectiveness and robustness of the proposed method is demonstrated via this experiment.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

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

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.

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.

WEDNESDAY MORNING, 4 DECEMBER 2013 UNION SQUARE 9, 10:45 A.M. TO 12:00 A.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
National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92016

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 9:00 a.m. on Tuesday, 3 December 2013.

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.

Session 3pAA

Architectural Acoustics: American Institute of Architects Continuing Education Units Course Presenter Qualification

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

Bennett M. Brooks, Cochair
Brooks Acoustics Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066

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

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. Architectural acoustics short course presentation material. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects, called "Architectural Acoustics." An architect can earn one continuing education unit (CEU) by attending this short course, if it is presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, finish treatments, and implementation of quality acoustical spaces. This paper will cover the course material in order to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this special session and membership in TCAA are required.

2:05

3pAA2. Architectural acoustics continuing education course—Presenter registration and reporting requirements. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects called "Architectural Acoustics," for which attendees can earn one continuing education unit (CEU). This paper will cover the administrative requirements of the AIA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork so that AIA members may receive credit for the course. The manner in which the course is given is also dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course. Of course, anyone is free to register with the AIA to provide their own CEU program. However, the advantages of participating in this program are that the TCAA short course is already prepared, it is pre-approved by the AIA, and the registration fees are paid by the Acoustical Society of America.

Session 3pAB

Animal Bioacoustics, Signal Processing in Acoustics, Psychological and Physiological Acoustics, and Speech Communication: Neural Mechanisms of Complex Sound Discrimination II

Andrea Simmons, Cochair
Brown Univ., Box 1821, Providence, RI 02912

Hiroshi Riquimaroux, Cochair
Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan

Contributed Papers

1:00

3pAB1. Timing patterns of strobe groups for echolocating big brown bats performing a target detection task. Laura N. Kloepper, James A. Simmons, Jason E. Gaudette (Dept. Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu), Ryan Himmelwright, and Dan Robitzski (Neurosci., Lafayette College, Easton, PA)

While flying in dense clutter or close to objects, bats often produce "strobe groups," pairs of pulses emitted at short pulse intervals followed by longer pulse intervals. Previous studies of free-flying bats demonstrate relatively consistent trends in strobe groups depending on the degree of clutter. To investigate strobe group production in stationary bats, three big brown bats were trained to perform a target detection task and their echolocation signals were analyzed. Variation in the number of pulses within strobe groups as well as strobe group characteristics varied substantially between individuals and experimental trials, yet when strobe groups were produced the time between pulses and strobe groups remained relatively stable. These data suggest that although bats demonstrate flexibility in the production and characteristics of strobe groups, the inherent timing of pulses within and across strobe groups is stable. These patterns may reflect precise timing adaptations in sound-producing motor circuits of individual bats.

1:15

3pAB2. The role of saccular resonance in fish audition. Mardi C. Hastings and Rachel Rozin (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Dynamic characteristics and resonance of the saccule play a fundamental role in audition in all teleosts, including those with direct connections between the swim bladder and inner ear. The saccule is an accelerometer with its rigid mass, the otolith (or saggita), coupled to the sensory epithelium through mechanical impedances of the otolithic membrane and hair-cell ciliary bundles. Relative displacement between the saggita and sensory epithelium induced by sound correlates with hearing sensitivity. Dynamic models of the peripheral auditory system in fishes from five different orders (the oscar, broad whitefish, oyster toadfish, dab, and goldfish) were developed for a comparative analysis. Species selected included one without a swim bladder and one with Weberian apparatus that transmits swim bladder motion directly to the saccule. Results for all fishes agreed with audiograms published in the literature. The lowest frequency marking the band of best sensitivity was found to be at the saccular resonance in all species; however, width of the band depended on excitation of the saccule indirectly from motion of the swim bladder and/or Weberian apparatus. Species with swim bladders and larger saggita had best sensitivities at lower frequencies, but with smaller bandwidths because the saccule could not respond to indirect stimulation.

1:30

3pAB3. A model for peripheral auditory mechanics in the oyster toadfish, *Opsanus tau*. Rachel Rozin (Georgia Inst. of Technol., Georgia Inst. of Technol., Atlanta, GA 30332-0405, rachelrozin@comcast.net), Peggy L. Edds-Walton (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Mardi C. Hastings (Georgia Inst. of Technol., Atlanta, GA)

Frequency response of the peripheral auditory system in the oyster toadfish was analyzed using a biomechanical model based on morphometric data obtained from a CT scan of a mature female, 21-cm long. Tissue properties for system equations were estimated from those found in the literature. The saccule is considered to have a single degree-of-freedom corresponding to the primary directional orientation of the hair cells. The model determines relative displacement between the sensory epithelium and otolith due to response of the saccule to motion from the sound source (direct path) and swim bladder (indirect path). Largest relative displacements correlate with highest auditory sensitivity (lowest thresholds). Results indicate a flat response at low frequencies with high sensitivity near 100 Hz, and are in good agreement with best stimulus frequencies measured physiologically in the auditory medulla and midbrain, and with the toadfish behavioral audiogram. Moreover, results confirm that the indirect path has little, if any, influence on auditory thresholds. Detection of the phase difference between direct and indirect signals, however, may contribute to the ability of oyster toadfish to localize sound sources.

1:45

3pAB4. Auditory steady-state response measurement of the temporal dynamics of hearing sensitivity in an echolocating bottlenose dolphin (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmpfoundation.org), James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, San Diego, CA), and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Studies with some echolocating odontocetes demonstrate that receiver-based automatic gain control (AGC) compensates for reductions in echo strength resulting from acoustic spreading loss. This study examined AGC in an echolocating bottlenose dolphin by measuring changes in hearing sensitivity over time courses corresponding to single click-echo pairs. The electrophysiological auditory steady-state response (ASSR) elicited by a 113-kHz sinusoidally amplitude-modulated tone was recorded while the dolphin performed a target discrimination task. Auditory electrophysiological responses were extracted from the instantaneous electroencephalogram and coherently averaged using the modulation rate of the 113-kHz tone as a reference. A Fourier transform was then performed with a 10-ms sliding window to obtain the ASSR amplitude as a function of time relative to the dolphin's outgoing click and received echo. The ASSR amplitude initially decreased at the time of click emission and then recovered over a course of 25 to 70 ms, depending on target range. This relatively long time course of

3p WED. PM

recovery appears to be consistent with forward-masking, as opposed to an AGC mechanism based on the contraction and gradual release of middle ear muscles coincident with click emission. [Work funded by SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

2:00

3pAB5. Investigating biosonar automatic gain control in a dolphin using auditory evoked potentials. James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Studies with echolocating odontocetes suggest that forms of automatic gain control mediate auditory electrophysiological responses to target echoes. This study used a phantom echo generator and auditory evoked potential measurements to examine automatic gain control in a bottlenose dolphin. Auditory evoked potentials to outgoing clicks and incoming echoes were recorded for simulated ranges from 2.5 to 80 m. When geometric spreading loss was simulated, echo-evoked potential amplitudes were essentially constant up to 14 m and progressively decreased with increasing range. When the echo levels were held constant relative to clicks, echo-evoked potential amplitudes increased with increasing range up to 80 m. These results suggest that automatic gain control maintains distance-independent echo-evoked potential amplitudes at close range, but does not fully compensate for attenuation due to spreading loss at longer ranges. The automatic gain control process appears to arise from an interaction of transmitter and receiver based processes, resulting in a short-range region of distance-independent echo-evoked potential amplitudes for relevant targets, and a longer-range region in which echo-evoked potential amplitudes are reduced. [Work funded by SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

2:15

3pAB6. Not-so-automatic gain control in the bottlenose dolphin: Source level distance compensation depends on prior knowledge of target distance. Laura N. Kloepper (Dept. Neurosci., Brown Univ., 185 Meeting St., Box GL-N,Providence, RI 02912, laura_kloepper@brown.edu), Paul E. Nachtigall, and Adam B. Smith (Zoology, Univ. of Hawaii, Honolulu, HI)

Although termed "automatic gain control," previous field and laboratory investigations into source level distance compensation in bats and odontocetes have relied on animals echolocating targets or arrays at predictable distances. To test the "automatic" nature of gain control in the bottlenose dolphin, the source level distance compensation was measured for three target distances (2, 4, and 7 m) in two types of sessions: predictable, in which the target distance was held constant within a session, and random, in which the target distance varied within a session. In the predicable sessions the dolphin demonstrated source level distance compensation at a rate of 10 log (distance), a level approximately half that predicted by past gain control experiments. In the random sessions the dolphin did not demonstrate source level distance compensation and, regardless of target distance, produced source levels that were equivalent to those produced for the predictable sessions at 4 m distance. These data suggest that gain control is not automatic, and, in the absence of prior knowledge of target distance, echolocating animals may adopt a strategy of fixing their source level to an intermediate distance of predicted target range.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 GOLDEN GATE 2/3, 1:00 P.M. TO 3:15 P.M.

Session 3pBA

Biomedical Acoustics: General Topics in Biomedical Acoustics

Robert McGough, Chair
Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824

Contributed Papers

1:00

3pBA1. A technique for measuring bone density using an ultrasonic imaging system. Catherine J. Miller, Morgan R. Smathers, Cameron R. Thurston, and Brent K. Hoffmeister (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, milcj-16@rhodes.edu)

Introduction: Osteoporosis is a degenerative bone disease that affects millions of Americans. Osteoporosis causes normally porous bone tissue, called cancellous bone, to become more porous and weak. It is possible that ultrasonic imaging systems may be used to detect changes in bone density (porosity) caused by osteoporosis. Methods: Ultrasonic images were acquired from 25 cube shaped specimens of cancellous bone in a water tank using a Terason 2000 + ultrasonic imaging system with a 5 MHz linear array transducer. Images were analyzed using an image processing program called ImageJ. Pixel brightness values were plotted as a function of depth in the images of each bone specimen. Pixel value gradient (PVG) was defined as the slope of the resulting graph. Results: PVG was negative for all specimens, and was found to decrease (become more negative) with bone density. PVG demonstrated a moderate but highly significant ($p < 0.001$) linear correlation with bone density (R = -0.79). Conclusion: Ultrasonic images of bone may be analyzed in ways that yield quantitative information about bone density.

1:15

3pBA2. Dual gate ultrasonic backscatter technique compared to x-ray microtomography parameters. Morgan Smathers, Joseph A. McPherson, Mark Sellers, and Brent K. Hoffmeister (Phys., Rhodes College, 2000 N Parkway, Memphis, TN 38112, smamr@rhodes.edu)

Over 52 million Americans suffer low bone mass and at least 10 million suffer from osteoporosis. This study seeks to develop a dual gate ultrasonic technique for predicting bone quality as well as bone quantity. Ultrasonic pulses from a 5 MHz transducer were propagated into regions of porous bone in 18 bone specimens from one bovine and four human donors. The dual gate technique considered the normalized mean of the backscatter difference (nMBD), which is the power difference between two gated regions of 2 μs each placed 1 μs apart over the returned signal. This ultrasonic parameter was compared to eight X-Ray MicroCT parameters describing bone quality and quantity. Among these are the Structural Model Index (SMI) and Relative Bone Volume (BV/TV). SMI grades the structure of a specimen based on its plate and rod characteristics, making it a bone quality characteristic. SMI produced an R value of 0.982 with nMBD. BV/TV, a bone quantity indicator, finds the ratio of bone volume in the total specimen volume, and showed an R value of 0.993 with nMBD.

1:30

3pBA3. Nonlinear propagation effects on measurement of backscatter coefficient of tissue-mimicking materials. Timothy Stiles (Phys., Monmouth College, 700 E Broadway Ave., Monmouth, IL 61462, tstiles@monmouthcollege.edu)

Measurement of the ultrasonic backscatter coefficient (BSC) holds great promise in providing quantitative diagnostic information on various diseases. Many clinical studies utilize the reference phantom method. In this method, the measured BSC is the ratio of the power spectrum of the scattered sound from the patient to the power spectrum from the reference phantom multiplied by the known BSC of the reference and an attenuation correction factor. In these studies, nonlinear propagation has been ignored. Nonlinear propagation causes changes in the power spectrum as incident energy is converted to harmonics and the acoustic signal undergoes nonlinear attenuation. This study characterized the effects of nonlinear propagation on reference phantom measurements of BSC from four tissue-mimicking samples in the frequency range from 2 to 20 MHz and incident pressure from 1 to 10 MPa (measured in water) using single element focused transducers. Samples consisted of glass microspheres suspended in a mixture of agar and concentrated milk. The resulting BSC varied by up to a factor of 30 depending on the incident pressure. Substantial changes in the overall shape of the BSC vs frequency were also observed. New methods of measurement of BSC that account for nonlinear propagation are explored.

1:45

3pBA4. 20—100 kHz, ultrasound assisted treatment of chronic wounds. Peter A. Lewin, Joshua A. Samuels (Biomed_7_701, Drexel Univ., 3145 Market St., Philadelphia, PA 19104, plewin@coe.drexel.edu), Michael S. Weingarten (Dept. of Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), Leonid A. Zubkov, Youhan Sunny, Christopher R. Bawiec (Biomed_7_701, Drexel Univ., Philadelphia, PA), and David J. Margolis (Dept. of Epidemiology, Univ. of Pennsylvania Perelman School of Medicine,, Philadelphia, PA)

We report the results of a limited (20 patients) clinical study involving treatment of chronic wounds (venous ulcers) using novel, fully wearable ultrasound array applicator operating in the range of 20—100 kHz and generating pressure amplitudes close to 55 kPa (about 100 mW/cm^2, Sptp). 20 kHz, 15 min exposure was determined to be the most effective in terms of expediting reduction in wound area ($p < 0.05$). The applicator was designed as compact, tether-free, device that can be comfortably worn by subjects at home, permitting active (combined with traditional compression) therapy away from the clinical setting. The system is safe for extended periods of application and the arrangement of the piezoelectric elements in the applicator has been adjusted allowing patient treatment customization, especially treatment of irregularly shaped ulcers. A set of systematic experiments *in vitro* aiming at the identification of potential mechanisms of the wound healing was also performed; the results verified that the exposure matrix used in clinical setting as mentioned above (20 kHz treatment for 15 min) produced the greatest increase in cellular metabolism ($p < 0.05$) and cell (fibroblasts) proliferation ($p < 0.01$) vs. a sham. The subsequent *in vitro* experiments will focus on collagen production, crucial to wound healing. [NIH 5R01EB009670.]

2:00

3pBA5. An ultrasound technique for wireless power transmission through tissue to implanted medical devices. Leon Radziemski (Piezo Energy Technologies, 5153 N Via Velazquez, Tucson, AZ 85750, ljrpet@comcast.net) and Inder R. Makin (Piezo Energy Technologies, Mesa, AZ)

An ultrasound electrical recharging system (USER^{TM}) is developed and tested, which wirelessly transmits significant amount of energy through animal tissue to charge implantable devices, batteries, or capacitors. The goal of this approach is wireless power transmission to active human implant devices. Experiments with transducers with resonant frequencies between 0.5 and 3.5 MHz led us to adopt 0.75 to 1.25 MHz as the range of optimum efficiency. *In vitro* experiments demonstrated significant charging of 4.1 V medically qualified Li-ion batteries across tissue depths of up to 5 cm. Charging currents close to 300 mA were achieved *in vitro*. Several *in vivo* tests confirmed the power delivery in a porcine model. In an *in vivo* survival test, tissue was exposed to 1 MHz ultrasound at an average intensity of 0.4 W/cm^2 for 11.5 h. Histology of the exposed tissue showed tissue changes primarily attributable only to surgical implantation of the prototype device. Many traditional and developing implanted medical devices are targets for the introduction of this method of power delivery, to reduce the number of battery replacement operations, and improve performance compared to the existing electromagnetic method of wireless power delivery. [Work supported by the NIH/NIBIB R44EB007421.]

2:15

3pBA6. Integrated transmission-reflection quantitative ultrasound non-invasive prediction of trabecular bone principal structural orientation validated with mechanical testing. Liangjun Lin, Wei Lin, and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., 100 Nicolls Rd., Rm 212, Bio-Eng. Bldg., Stony Brook, NY 11794-5281, john85726@gmail.com)

Quantitative ultrasound (QUS) measurement was shown to have the ability to predict the principal structural orientation (PSO) with a spherical bone model. It is hypothesized that with a cubic bone model, integrated transmission-reflection QUS measurement can predict the PSO and therefore improved the correlation with the mechanical and structural parameters. Twelve trabecular bone cubes were harvested from bovine distal femur. Compression testing and μCT of 30 μm resolution were performed to obtain the mechanical and structural parameters. QUS measurement was performed on the transverse plane in a range of angles, from —-30° to 30° to medial-lateral orientation at 5° increment. For each angle, reflection mode was used to measure the thickness of the sample in the specific scanning angle. Then, the sample thickness was used to normalize the transmission mode measurement in the same angle. The thickness measured by reflection mode QUS was highly correlated to the results measured by caliber (slope = 0.99, R2 = 0.85). Compared to the traditional transmission mode, the correlation coefficients (R2) between transmission-reflection mode ultrasound velocity versus mechanical and structural parameters were improved (elastic modulus, 0.62 to 0.73; SMI, 0.74 to 0.90; BV/TV, 0.75 to 0.85; Tb.N, 0.60 to 0.75; Tb.Sp, 0.66 to 0.73).

2:30

3pBA7. A comprehensive computational model of sound transmission through the porcine lung. Ying Peng (Mech. Eng., Univ. of Illinois at Chicago, 2951 S King Dr. Apt. 1805, MIE, Chicago, IL 60616, YPENG6@UIC.EDU), Zoujun Dai, Brian Henry (BioEng., Univ. of Illinois at Chicago, Chicago, IL), Hansen Mansy (Rush Univ., Chicago, IL), and Thomas Royston (BioEng., Univ. of Illinois at Chicago, Chicago, IL)

A comprehensive computational simulation model of sound transmission through the porcine airways and lung is introduced and experimentally evaluated. This "subject-specific" model utilizes parenchymal and major airway geometry derived from x-ray CT images. The lung parenchyma is modeled as a poroviscoelastic material using Biot theory. A finite element (FE) mesh of the lung that includes airway detail is created, and COMSOL FE software is used to simulate the vibroacoustic response of the lung to sound input at the trachea. The FE model is validated by comparing simulation results with experimental measurements using scanning laser Doppler vibrometry on the surface of an excised and preserved lung. The FE model is also used to calculate and visualize vibroacoustic pressure and motion inside the lung and its airways caused by the acoustic input. The effect of diffuse lung fibrosis and a tumor on the lung acoustic response is simulated and visualized using the FE simulation. In the future, this type of visualization can be compared and matched with experimentally-obtained elastographic images in order to better quantify lung material properties. [Work supported by NIH Grant EB012142.]

2:45

3pBA8. A simulation model of cyclic variation of red blood cell aggregation under Couette and Poiseuille flows. Qi Kong, Kwon-Ho Nam, and Dong-Guk Paeng (Ocean system Eng., Jeju National Univ., Rm. 5464,College of Ocean Sci. 4th Bldg., Jeju National Univ., Ara 1-Dong Jeju-Si Jeju-Do, Jeju, South Korea, kongqi2011@gmail.com)

The aggregation of red blood cells (RBCs) is a reversible phenomenon in which RBCs form a pile or network at low shear rate via the interactions of electrostatic and aggregating forces. In previous experimental results under both Couette and Poiseuille flows, cyclic variations in blood echogenicity were observed but their cyclic patterns were different. In this paper, a two-dimensional particle model capable of mimicking the main characteristics of RBC aggregation kinetics was proposed to elucidate the different effects of hemodynamics under Couette and Poiseuille flows on RBC aggregation. In simulation results, cyclic variation of RBC aggregation was observed but its magnitude of mean aggregate size (MAS) was not changed by variations of velocity and stroke rate under Couette flow. These results are in agreement with the experiment results. In contrast, the simulation results under Poiseuille flow revealed that cyclic variation of RBC aggregation and its MAS magnitude were changed by variations of velocity and stroke rate. As stroke rate increased from 20 to 120 beats/min, the phase of MAS variation compared with velocity profile was shifted. These simulation results may provide the theoretical explanation of the different experimental results of cyclic variation of RBC aggregation under Couette and Poiseuille flows. [Work supported by NIPA-2013-H0401-13-1007 and 2013R1A1A2043478.]

3:00

3pBA9. Errors in ultrasonic scatterer size estimates due to mixed scatterer populations. Anthony L. Gerig and Breanna P. Swan (Mathematics and Phys., Viterbo Univ., 900 Viterbo Dr., La Crosse, WI 54601, algerig@outlook.com)

Ultrasonic scatterer size estimation provides an accurate measure of actual scatterer size when those sizes are narrowly distributed about a single, mean value. Although often the case, there are instances in tissue where two or more scatterer types with significantly different sizes are believed to contribute to the same signal. The purpose of this work is to characterize the errors in the size estimates obtained for one scatterer type when contaminating scatterers of a second type are present. Theoretical results for the error are compared with simulation and experimental results for uniform phantoms containing a binary mixture of scatterers. These results indicate that errors can be significant for the frequency bands typically used in size estimation, especially when contaminant scatterers are larger than the scatterers of interest. Results also indicate that, however, these errors can be reduced by shifting the frequency band used to estimate size. A technique for correcting the errors is also described and applied to the phantom data. Although effective, the method requires prior knowledge of the backscatter coefficient of the contaminant scatterers, and the variability of the corrected values can limit its utility as contaminant scattering strength increases.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 MASON, 1:30 P.M. TO 2:45 P.M.

Session 3pEA

Engineering Acoustics: Non-Traditional Electro-Acoustic Transducer Design II: Contemporary Micro-Mechanical Devices

John B. Blottman, Chair
Div. Newport, Naval Undersea Warfare Ctr., 1176 Howell St., Code 1535 B1170/108, Newport, RI 02840

Contributed Papers

1:30

3pEA1. Characterization of a microelectromechanical microphone using the finite element method. Mads J. H. Jennings (Comsol AB, Kgs. Lyngby, Denmark), Wade Conklin, and Jordan Schultz (Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60143, wade.conklin@knowles.com)

Recent development and advances within numerical techniques and computers now enable the modeling, design, and optimization of many transducers using virtual prototypes. Here, we present such a virtual prototype of a Knowles SiSonicTM MEMS microphone. The virtual prototype is implemented using the finite element method with COMSOL Multiphysics and includes description of the electric, mechanic, and acoustic properties of the transducer. The acoustic description includes thermal and viscous losses explicitly solving the linearized continuity, Navier-Stokes, and energy equations, that is, thermoacoustics. The mechanics of the diaphragm are also modeled including electrostatic attraction forces and acoustic loads. A sub-model approach is used to model and lump the acoustic properties of the small perforations in the microphone backplate. The model has no free fitting parameters and results in the prediction of the frequency response, in the audible and ultrasound range, as well as other relevant characteristics. The model results show good agreement with measured data.

1:45

3pEA2. A piezoelectric micromachined ultrasonic transducer array for parametric loudspeakers. Yub Je (Mech. Eng., Pohang Univ. of Sci. and Technol., San 31, Hyojadong Namgu, Pohang, Kyungbuk, KIRO 416, Pohang, Gyungbuk, South Korea, wkmoon@postech.ac.kr), Haksue Lee (Agency for Defense Development, Jinhae, South Korea), and Wonkyu Moon (Mech. Eng., Pohang Univ. of Sci. and Technol., Pohang, South Korea)

A parametric array is a nonlinear conversion process that can generate a highly directional sound beam with a small aperture. Since parametric sound generation requires high-intensity ultrasound for nonlinear interaction, efficient sound generation is an important issue in the practical use of parametric loudspeakers. In this study, a piezoelectric micromachined ultrasonic transducer array was investigated in order to generate directional audible sound with a parametric array. A piezoelectric micromachined ultrasonic transducer, with a micron-thick radiating plate, was verified to generate ultrasound with high-efficiency in air. Two types of unit transducers with different resonance frequencies (f1 = 100 kHz, and f2 = 110 kHz) were arranged in the transducer array to extend the frequency bandwidth. The electroacoustic efficiency of the transducer was measured to 71% at its resonance frequency. The $\pm$3dB-frequency bandwidth of the transducer array was 17 kHz. The spatial distributions of the difference frequency wave were measured and compared with the

computed data in the audible frequency range. The fabricated transducer array consumed 1 W of electric power while generating a 10 kHz-difference frequency wave with sound pressure level of 80 dB.

2:00

3pEA3. The performance enhancement of a micro-machined microphone based on field-effect-transistor and electrets. Kumjae Shin, Yub Je (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technol., KIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk 790-784, Korea, Republic of, forhim13@postech.ac.kr), Haksue Lee (6th R&D Institute-1, Agency for Defence Development, Changwon, Kyoungnam, South Korea), James E. West (Dept. of Electric & Comput. Eng., Johns Hopkins Univ., Baltimore, MD), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technol., Pohang, Gyungbuk, South Korea)

Most microphones use a capacitive type transduction. However, this form of transduction faces problems with microminiaturization. The most prominent issue is a decrease in sensitivity at low frequency. Although several works suggested a microphone based on the field-effect-transistor (FET) to solve this problem, other issues, such as low signal to noise ratio and a need for high bias voltage due to metal electrode, remained. To overcome this limit, a micro-machined microphone based on the FET and electret was proposed, and its feasibility was shown in 2012. Its principle is that the electric field arising from the electret controls the channel of the FET embedded on the membrane. Although its feasibility as an acoustic-sensitive device was shown, several problems still exist in terms of stability, sensitivity, and noise. To realize stable and highly sensitive modulation, parametric analysis for the transduction was done to enhance performance. In particular, the surface potential of the electret was increased more than the previous one. It resulted in a stronger electric field applied at the gate. Therefore, it made the FET more sensitive to membrane vibration. The sensitivity evaluation setup was modified for a more accurate measurement. The evaluation results are to be presented.

2:15

3pEA4. Experimental comparison of 3-3 and 3-1 mode piezoelectric microelectromechanical systems. Donghwan Kim (Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712, donghwan.kim@utexas.edu), Nishshanka N. Hewa-Kasakarage, Michael L. Kuntzman, and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

A common architecture for piezoelectric MEMS sensors and actuators is a thin piezoelectric film patterned atop a much thicker passive bending structure (e.g., a silicon beam or plate). In a first common configuration, parallel plate electrodes reside above and below the piezoelectric film to realize a 3-1 mode device. In a second configuration, a top electrode is patterned in the form of an interdigitated transducer (IDT) to realize a parallel network of 3-3 mode cells. A theoretical comparison of figures-of-merit for each configuration has been presented by research teams in the past. Figures-of-merit include coupling coefficient, actuator strength, and signal to noise ratio for sensing applications. Less work has been performed directly comparing these configurations experimentally using micro-processed thin films. In this presentation, a micromachined accelerometer structure employing a set of multiple springs is used to experimentally compare the two configurations. Each silicon spring contains a 1micron thick lead zicronate titanate (PZT) film along the top surface. Aside from electrode type, the springs are identical in dimension—providing an opportunity for direct comparison.

2:30

3pEA5. A micro-machined Tonpilz hydrophone for audible frequency sounds. Min Sung (Mech. Eng., POSTECH, PIRO416, San31, Hyoja, Namgu, Pohang city, Kyungbuk, Pohang, Kyungbuk 790784, South Korea, smmath2@postech.ac.kr), Haksue Lee (Underwater Acoust. Lab., Agency for Defense Development, Changwon, Kyungnam, South Korea), and Wonkyu Moon (Mech. Eng., POSTECH, Pohang, Kyungbuk, South Korea)

A micro-machined Tonpilz hydrophone based on the piezoelectric thickness mode was designed for audible frequency range of 20 Hz~20kHz. The structure of the sensor was motivated by conventional Tonpilz transducers, but two different design approaches were adopted to enhance the sensitivity and to endure the high hydrostatic pressure in deep-sea. For improved sensitivity, the area ratio of the head-mass and the piezoelectric body was designed to be several hundreds to one, which amplifies the input of the transduction body due to acoustic pressure. Since this approach is adopted in order to develop a miniaturized hydrophone manufactured by a batch process, the size of the piezoelectric transduction body becomes too small to generate the amount of charge enough for detecting the signal using the conventional pre-amplifier at low frequencies below 500 Hz. We have developed and validated the lumped parameter model and used it to identify the requirements for the pre-amplifier circuits and the available sensitivity at low frequencies and to search for the appropriate design for miniaturized hydrophone. Then, the designed hydrophone was fabricated by micromachining and assembled with the custom-made pre-amplifier inside with caster-oil filled rubber housing. The evaluation of the hydrophone will be presented. [Research supported by MRCnD.]

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Natalia Sidorovskaia, Chair
Dept. of Phys., Univ. of Louisiana at Lafayette, UL BOX 44210, Lafayette, LA 70504-4210

Chair's Introduction—2:00

Invited Paper

2:05

3pED1. Time-frequency analysis for acoustics education and for listening to whales in the Gulf of Mexico. Juliette W. Ioup (Dept of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu)

Time-frequency plots continue to be used in many varied applications. One particularly advantageous use is in acoustics courses accessible to non-science majors, students who are often frightened of mathematics and/or physics. All musicians as well as many others can read and understand music scores (time-frequency plots). Time-frequency plots are extremely useful in explaining the differences in timbre of the same pitch coming from different musical instrument families, from individual instruments themselves, and from different human voices. Examples are given from the first of a UNO two-semester sequence on the Physics of Music (textbook by Rossing!). The second semester of this sequence includes recording and reproduction of music, and time-frequency plots are again very useful. Investigation of acoustic signals for research also benefits from the use of time-frequency plots. The study of marine mammals is enhanced by analysis of underwater acoustic recordings. Examples of both the sounds of and the time-frequency plots for sperm whale clicks in the northern Gulf of Mexico are presented. Seismic airgun shots from oil industry exploration can be heard on the recordings as well as the whale clicks.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Frederick J. Gallun, Chair
National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Chair's Introduction—1:15

Invited Papers

1:20

3pID1. Hot topics in architectural acoustics: Classroom acoustics and archaeoacoustics. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

When ASA members discovered that typical American classrooms were too noisy or reverberant for serious learning in 1988 they began a successful grassroots movement to fix them. By 2002, ASA volunteers produced the first-ever ANSI standard for classroom acoustics. The effort was led by ASA's TCAA and supported by three other TCs, the S12 Standards Committee, courageous ASA staffers, and elected officers. ANSI Standard S12.60 was adopted fully or partly by school districts, states, and architectural authorities including the Green Building Council. Classroom acoustics research reporting remains active at ASA meetings. We show how this standard helps to make a better world. The new fields of archaeoacoustics and historical acoustics are "hot". They employ scientific

acoustics to study the past. Their novel hypotheses and discoveries attract young investigators to acoustical careers. Sound was more important in the quiet ancient world. Many are fascinated by the 1988 discovery of strong correlation between the locations of Paleolithic cave paintings and cave resonance. Why? A pyramid at Chichen Itza, Mexico, chirps like a bird revered in Mesoamerican cultures. The chirp is explained by applying the convolution theorem to pyramid architecture. Was it intentionally designed? Other archaeoacoustic and historical acoustic examples are addressed.

1:45

3pID2. Single beam acoustic tweezer. K. K. Shung (Biomed. Eng., Univ of S. Calif., 136 DRB, 1042 Downey Way, Los Angeles, CA 90089, kkshung@usc.edu)

Single beam acoustic tweezer, a distant cousin of optical tweezer, has been recently experimentally validated. A prerequisite of acoustic tweezer as in optical tweezer is a sharply focused beam with a steep intensity variation within the dimension of a particle. As the frequency of an acoustic beam reaches 100 MHz or higher, the beam diameter approaches cellular level allowing acoustic tweezing or trapping of a cell. Recent experimental results have shown that at 200 MHz it is possible to trap particles as small as 1 μm and red blood cells (RBC). These results along with the experimental arrangement and potential biomedical applications of acoustic tweezer including measuring RBC deformability will be discussed in detail in this talk.

2:10

3pID3. Hot topics in musical acoustics applied to real-time sound synthesis. Julius O. Smith (Music / CCRMA, Stanford Univ., Stanford, CA 94305, AbstractCentral@w3k.org)

(Invited paper for an Interdisciplinary session) New activities in any field are often precipitated by new enabling technologies. In musical acoustics applied to real-time sound synthesis, one exciting new development is smart-phones and tablets having high audio quality, multicore processing power, and display screens with multitouch controls. For example, it is possible to implement a complete virtual-acoustic guitar on current smart-phones and tablets, playable in real time, with plenty of processing power left over for real-time display of chord charts and vibrating-string graphics. (See presentations of the moForte Guitar at this conference and/or on the Web.) A second enabling technology area is evolving high-level domain-specific languages for audio signal processing such as Functional AUdio STream (FAUST). For example, the source code for the moForte virtual acoustic guitar, written in the FAUST language, is about an order of magnitude smaller (and faster to write) than conventional C++, while the compiled performance is generally within a factor of 2. A third enabling technology area is advancement in methods for convex optimization along with advances in the development of convex formulations of important problems. For example, a convex formulation has recently been developed for approximating 2D bowed-string bridge admittances as ”passive” (positive-real) digital filters. (See presentations by Esteban Maestre.)

2:35

3pID4. Hot topics: Recent discoveries using new recording methods in animal bioacoustics. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

Recent research has successfully addressed a variety of difficult questions about animal acoustic communication and orientation by taking advantage of two related techniques—on-board recording or telemetry of animal sounds or neurophysiological signals during behavior, and acoustic tracking of the locations of animals using arrays of microphones or hydrophones coupled with efficient software for processing the time-of-arrival of sounds. These methods have great power for relating the locations and movements of animals to their acoustic signals and auditory responses, but they have limitations that have to be taken into account if their potential is to be realized. The examples to be presented highlight discoveries that have been made using these methods. They have broadened our understanding of how animals interact acoustically with each other according to their spatial distribution and proximity, how they dynamically regulate auditory sensitivity.

Session 3pNS

Noise and Architectural Acoustics: Double Duty: The Added Value of Coupling Acoustics with Other Functionality

David S. Woolworth, Chair
Oxford Acoust., 356 CR 102, Oxford, MS 38655

Chair's Introduction—1:25

Invited Papers

1:30

3pNS1. Supporting dynamic infrastructure loads from a vibrating structure. Alexander Salter (Charles M. Salter Associates, Inc., 130 Sutter St., Fifth Fl., San Francisco, CA 94104, alex.salter@cmsalter.com)

A large ballroom at a downtown hotel in San Francisco sits directly below the parking garage with ramps located both under and over the Ballroom's parlor areas on either side. In 2011 Charles Salter Associates, Inc., together with the project architect and structural engineer, developed an isolated structural grid to support the ballroom infrastructure. The grid also allows for variable loading at select rigging points to accommodate a wide range of events. Prior to the renovation, significant vibration and structure borne noise was generated inside the ballroom due to vehicular activities. The ceiling structure, ductwork, and piping, as well as the chandeliers visibly shook and rattled during vehicle pass-bys. As a result, the hotel received complaints and needed to mitigate the issue in order to provide a first class ballroom facility. This paper discusses the background and design process taken in order to mitigate the issue, and describes how the isolated grid provides both acoustical mitigation as well as utility to users of the hotel's ballroom.

1:50

3pNS2. Combining acoustical and fire-resistive design in separating assemblies. John LoVerde and David W. Dong (Veneklasen Associates, 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

In most project types, separating assemblies must be fire rated, which means the assembly design and materials must be listed by one of the approved fire rating bodies. This can limit the acoustical design options to assemblies and materials that have previously been listed. For large developers, it can be advantageous to design new assemblies that are not currently listed that meet both the acoustical goals and the fire code requirements. Acoustical testing is performed on design iterations and with various products, in coordination with architectural elements and product manufacturers. The completed assembly is burn testing and agency listed. Examples and lessons of the process are shared.

2:10

3pNS3. The importance of the relationship of acoustics and sustainability. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Sustainability has become a keyword in the architecture and the building industry, in no small part driven by the development of standards such as IGBC and LEED. The refined definition of sustainability, however, depends on the viewpoint: economic, social, or environmental. Acoustics of buildings (and the environment) has an intrinsic link to sustainability that should be highlighted to prevent the sidelining of architectural acoustics in building budgets and architectural programs of study. This paper examines the concept of sustainability as it relates to the building industry and architectural acoustics, and provides a summary of work in this area to outline this relationship to illuminate the importance of acoustics in sustainability.

2:30

3pNS4. Relating acoustics and thermal performance. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Good thermal performance of a building produces a tangible financial return to the owner that comes from energy savings. While some not-as-directly tangible returns such as functionality or productivity can be exacted from good acoustics, some acoustic treatment materials or techniques also exhibit beneficial thermal properties. This paper examines and quantifies some of these acoustic approaches that allow thermal benefits to "piggy back" and help embed the acoustic design into the project so that it is not so easily removed in the value engineering phase of the project.

Session 3pPAa

Physical Acoustics, Noise, Structural Acoustics and Vibration, and Engineering Acoustics: Jet and Other Aeroacoustic Noise Source Characterization II

Kent L. Gee, Cochair
Brigham Young Univ., N243 ESC, Provo, UT 84602

Tracianne B. Neilsen, Cochair
Brigham Young Univ., N311 ESC, Provo, UT 84602

Invited Papers

1:00

3pPAa1. Sensitivity analysis of an equivalent source model for a military jet aircraft. Tracianne B. Neilsen, Kent L. Gee, David M. Hart (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Research and Consulting, LLC, Asheville, TN)

The noise from a tied-down F-22A Raptor is modeled with an equivalent source consisting of two line arrays of monopole sources and their image sources, to represent the interference from the ground. These arrays, one correlated and one uncorrelated, with Rayleigh-distributed amplitudes, mimic properties of fine and large-scale turbulent mixing noise. [Morgan *et al.*, Noise Control Eng. J. **60**, 435-449 (2012)]. The equivalent source modeling parameters (the distributions' peak locations, amplitudes, widths, and the relative phase angle between correlated sources) are selected using Bayesian optimization implemented with simulated annealing and fast Gibbs sampler algorithms. The resulting equivalent source model reasonably predicts the radiated midfield up to 1250 Hz [Hart *et al.*, POMA **19**, 055094 (2013)]. In this study, the relationship between the correlated array's peak location and its phase angle has been further analyzed. Although sensitivity analysis of the results reveals non-uniqueness of the model, it also yields additional physical insight in the form of bounds for the dominant aeroacoustic source region as a function of frequency. The far field sound radiation predicted by the equivalent source model for a wide range of frequencies will be compared to measured far-field directivities. [Sponsored by the Office of Naval Research.]

1:20

3pPAa2. Acoustical holography and proper orthogonal decomposition analyses of full-scale jet source properties. Alan T. Wall (Dept. of Phys. and Astronomy, Brigham Young Univ., Bldg. 441, Wright-Patterson AFB, Ohio 45433, alantwall@gmail.com), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Research and Consulting, Asheville, NC)

Efforts to characterize and reduce noise emissions from high-performance fighter aircraft are often focused on the modeling and control of large-scale turbulence structures within the jet exhaust. In past investigations, these structures have been represented as oscillatory functions whose amplitudes grow then decay with distance from the jet nozzle, or wave packets. Recently, acoustical holography methods have been used to reconstruct the radiated field of a full-scale jet, including field properties at the source. Proper orthogonal decomposition of the reconstructed source region, based on the multiple signal classification (MUSIC) algorithm, provides a physically intuitive set of partial sources. Taken together, this set is an equivalent source representation of the large-scale structures. Individually, each partial source exhibits similar qualitative behavior to that of wave packet source models.

1:40

3pPAa3. Wavepacket noise source model for microphone array data analysis of hot supersonic jets. Philip Morris (643 Berkshire Dr., State College, PA 16802, pjm@psu.edu), Robert Dougherty (OptiNav, Inc., Bellevue, WA), Chris Nelson (ITAC, LLC, Lynnwood, Washington), Alan Cain (ITAC, LLC, Chesterfield, MO), and Kenneth Brentner (State College, PA)

Phased arrays of microphones have proved themselves to be a powerful tool for aeroacoustic investigations. There are many different algorithms for processing the resulting data, including classical beamforming, its modern derivatives, Linear Programming, and Generalized Inverse Methods. The current work stems from a recognition that, for configurations with extended coherent sources (such as hot supersonic jets), the Generalized Inverse Method may be preferred, but that its accuracy can be improved by improving the underlying source model that it uses. We examine a wavepacket-based source model for analysis of the noise emitted from hot supersonic jets. This approach provides a more physically realistic representation of the jet noise sources than previously used. The model is tested using data obtained from numerical simulations as measured at a "virtual" array of microphones. The resulting generalized inverse method analysis is then used to predict noise at a farfield arc, and this prediction is compared with that from a conventional Ffowcs Williams-Hawkings acoustic analogy prediction. Initial results with the new wavepacket source model are encouraging, with improved directivity predictions and elimination of some spurious noise sources. The results of the ongoing model development will be included in the final paper.

2:00

3pPAa4. Large eddy simulation of crackle noise in hot supersonic jets. Joseph W. Nichols (Aerosp. Eng. and Mech., Univ. of Minnesota, 107 Akerman Hall, Minneapolis, MN 55455, jwn@umn.edu) and Sanjiva K. Lele (Aeronautics and Astronautics, Stanford Univ., Stanford, CA)

Crackle noise from heated, supersonic jets is investigated through high-fidelity large eddy simulation (LES). Simulations of a military-style nozzle reveal that N-shaped waves responsible for crackle noise emerge directly from the supersonic jet turbulence, and thus do not depend solely on nonlinear acoustic propagation effects. Conditional averaging using a backtracking algorithm furthermore reveals intermittent large-scale flow structures embedded in the jet turbulence as a likely source of the N-shaped waves. The skewness of pressure (Ffowcs Williams *et al*., 1975) and the skewness of the time derivative of pressure (Gee *et al*., 2007) are evaluated from full-field simulation data, and assessed as metrics of crackle. We test the hypothesis that the crackle level depends solely on the jet velocity (Ffowcs Williams *et al*., 1975) by comparing three simulations sharing the same geometry, but having different operating conditions so that the velocity and temperature are varied independently.

2:20

3pPAa5. Unstructured large eddy simulations of heated supersonic twin jets. Guillaume A. Brès, Frank E. Ham (Cascade Technologies, 2445 Faber Pl., Ste. 100, Palo Alto, CA 94303, gbres@cascadetechnologies.com), and Sanjiva K. Lele (Stanford Univ., Stanford, CA)

Unstructured large eddy simulations are performed with the compressible flow solver ”Charles” developed at Cascade Technologies, for heated supersonic over-expanded jets issued from a twin nozzle. The study focuses on the modeling of the internal flow and its effects on the flow-field in the jet plume and ultimately on the radiated noise. In this work, near-wall adaptive mesh refinement, synthetic inflow turbulence and wall modeling are used inside the nozzle. In addition to the converging-diverging nozzle geometry, the computational domain includes the Y-duct, S-ducts, and angle adapter upstream of the nozzles, to realistically reproduce the elements of the experimental configuration where the internal flow conditions and wall modeling can be expected to affect the external flow-field and sound-field predictions. Comparisons between the numerical predictions and experimental PIV and far-field noise measurements from NASA Glenn Research Center will be presented and discussed.

Contributed Papers

2:40

3pPAa6. Infrasonic emissions from aircraft wake vortices: Field installation. Qamar Shams, Howard K. Knight (NASA Langley Res. Ctr., Hampton, VA), and Allan J. Zuckerwar (Analytical Services & Mater., Hampton, VA)

An infrasonic field installation was set up at Newport News-Williamsburg International Airport in early 2013. The system is made up of three PCB 377M06 microphones installed into non-porous subsurface windscreens [POMA 1pNS9, **18**, 040005 (2013)], which limit the bandwidth to 100 Hz. The microphones are placed 250 ft (76.2 m) orthogonal to the runway and 200 ft (60.96 m) apart. The data acquisition system is the B&K Pulse, from which time histories, spectra, and coherence between microphone channels are derived. The system is placed inside an instrumentation vehicle just behind the center microphone. Perforated drainage hoses are installed from the subsurface windscreens to adjacent drainage ditches and weight is added to the windscreens for additional stability. The drainage system has proved successful even on occasions of heavy downpour, revealing a truly all-weather system. A pistonphone calibration at 14 Hz in the field reveals that the three channels are matched to within 2 dB. This capability permits long-term monitoring of the health of the system. A sample time history of signals received from an aircraft takeoff will be presented.

2:55

3pPAa7. Infrasonic emissions from aircraft wake vortices: Experimental results. Allan J. Zuckerwar (Analytical Services & Materials Inc., 107 Res. Dr., Hampton, VA 23665, ajzuckerwr@yahoo.com), Qamar Shams, and Howard K. Knight (NASA Langley Res. Ctr., Hampton, VA)

Infrasonic emissions from aircraft wake vortices were investigated at the Newport News-Williamsburg International Airport early in the year 2013. Signals received by the microphones situated along an airport runway were processed in 10-s intervals. As an aircraft accelerates toward takeoff, it produces a large pressure burst as it passes each microphone. Following the burst, there appear low-frequency signals of high coherence among microphone pairs. These are interpreted as emissions from the aircraft wake vortices, as suggested by theory. In successive 10-s intervals, the coherence gradually diminishes to background levels, signifying the disappearance of the vortices. On landing the intervals of high coherence precede the bursts at aircraft touchdown, and then diminish. The pressure burst serves as a time stamp for the ensuing vortex emissions and thereby permits the tracking of successive takeoff or landing events on the same runway or on adjacent runways. The emission spectrum is essentially broadband, lacking spectral features (e.g., tones). Data were taken for takeoff of Airbus 319, DC-9, MD-88, CRJ, Lear Jet, Corporate Jet, and Dash-8 aircraft, and for landing of the Airbus 319. The pattern of pressure burst, high-coherence intervals, and diminishing-coherence intervals was observed for all takeoff and landing events without exception.

Session 3pPAb

Physical Acoustics: Topics in Geophysical Acoustics

Bradley Goodwiller, Cochair
Mech. Eng., The Univ. of Mississippi, 1 Coliseum Dr., Rm. 1079, University, MS 38677

Tiffany Grey, Cochair
National Ctr. for Physical Acoust., 1 Coliseum Dr., University, MS 38677

Contributed Papers

1:00

3pPAb1. Backscattering and attenuation mechanisms in solids and solid-liquid suspensions. Paul Panetta (Applied Research Associates, Inc., Mail: P.O. Box 1346, Shipping: Rte. 1208 Greate Rd., Gloucester Point, VA 23062, ppanetta@ara.com)

The ultrasonic backscattering and attenuation are commonly used to characterize the properties of solids and solid-liquid suspension to determine grain morphology for solids and particle size and solids loading for solid liquid suspensions. An ultrasonic field is attenuated by absorption and scattering mechanisms as the field traverses a material. However, the relative strength of the absorption, single scattering and multiple scattering contributions are often unknown. In solids, the grain morphology and the dislocation properties are especially important contributions, and in solid-liquid suspension, the particle size and concentration control the attenuation. This paper will present a study of the attenuation mechanisms in solids and solid-liquid suspensions utilizing traditional attenuation, backscattering, and resonance or diffuse field measurements of the attenuation. The results provide the potential to separate the multiple scattering, single scattering, and absorption contribution to the various ultrasonic attenuation measurements on stainless steel alloys and solid liquid suspension. Results for solids and solid-liquid suspensions which elucidate the interrelationship between these energy loss mechanisms will be reported. Where appropriate, the experimental measurements will be compared with theoretical predictions.

1:15

3pPAb2. Sound speed and frequency-dependent attenuation determination in highly attenuating lubrication fluids. Blake Sturtevant and Dipen N. Sinha (Mater. Phys. and Applications, Los Alamos National Lab., MPA-11, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Acoustic characterization of lubricating drilling muds is essential for the design of acoustics-based sensors and imaging devices for downhole petroleum or geothermal well environments. This work reports on the measurement of sound speed and the determination of acoustic attenuation in drilling muds with densities ranging from 10 to 15 pounds/gallon. Measurements were made in a two transducer pitch-catch configuration as a function of distance up to 40 cm and as a function of frequency up to 1 MHz. Corrections for diffraction will be discussed. Experimentally determined data are compared with previously reported attenuation values at selected frequencies, specifically 180 kHz and 280 kHz, and found to be in good agreement. The dB/cm attenuation values for the muds studied in this work were found to be three and five orders of magnitude greater than those values for silicone oil and water, respectively, at the same frequencies

1:30

3pPAb3. Design and implementation of a passive acoustic bedload monitoring surrogate system. Bradley Goodwiller, James Chambers (Mech. Eng., The Univ. of MS, 1 Coliseum Dr., Rm. 1079, University, MS 38677, btgoodwi@olemiss.edu), Wayne Carpenter (National Ctr. for Physical Acoust., The Univ. of MS, University, MS), Daniel Wren, Roger Kuhnle, Jr. Rigby (National Sedimentation Lab., Oxford, MS), and Robert Hilldale (Tech. Service Ctr., U.S. Bureau of Reclamation, Denver, CO)

Various methods of employing passive acoustics to monitor bedload transport by listening to noise generated by colliding gravel have been explored in both the lab and field. Expanding upon this research, a hydrophone-based passive bedload-monitoring system was designed, tested, and deployed by researchers at the University of Mississippi and the National Sedimentation Laboratory in Oxford, MS. A series of laboratory experiments was used to measure the dependence of acoustic propagation on various parameters such as depth of the water and placement of the hydrophones. These tests employed a calibrated acoustic transmitter to broadcast tones of various frequencies across a gravel bed. Additional flume tests used a motorized carrier to drag rocks across the gravel bed, allowing a known sediment flux to be measured acoustically without any flow noise. The hydrophone-based system was deployed on the Trinity River, Weaverville, CA, in the summer of 2012 and the Elwha River, Port Angeles, WA, in the summer of 2013. Both deployments were co-incident with physical bedload measurements taken by Graham Mathews and Associates. The design of the acoustic system, methods used to analyze the data, results from laboratory experiments, and preliminary results from field data collection will be presented.

1:45

3pPAb4. Measurement and modeling of pulsed bi-frequency, nonlinear acoustic excitation of buried landmines. Benjamin J. Copenhaver, Justin D. Gorhum, Charles M. Slack, Martin L. Barlett, Thomas G. Muir, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, bcopenhaver@utexas.edu)

To help resolve certain practical issues with acoustical methods for humanitarian landmine detection, we have researched using a pulsed, standoff source method for acoustical excitation of the buried mine [J. Acoust. Soc. Am. **130**, 2541 (2011); J. Acoust. Soc. Am. **133**, 3457 (2013)]. Pulses consisting of two primary frequencies are used in order to search for induced nonlinear vibrations at interaction frequencies such as the sum frequency, which arise due to nonlinear interaction at the mine/soil interface. To model the pulsed excitation, we employ a fully nonlinear time-domain

3p WED. PM

implementation of the lumped-element model of nonlinear soil/mine interaction introduced by Donskoy *et al*. [J. Acoust. Soc. Am. **117**, 690 (2005)]. Modeling is compared with experimental results, which are obtained with bi-frequency pulses exciting a soil with a buried landmine replica, instrumented with a geophone and a nearby microphone. Cases investigated include: (1) target only, (2) buried target under disturbed soil, (3) disturbed soil only, and (4) undisturbed soil. Excitation both on and off the resonance of the buried mine is also investigated, as is burial in different soil types at various depths. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:00

3pPAb5. A unified model of hysteresis and relaxation in geophysical materials. Lev A. Ostrovsky (PSD, NOAA ESRL, 325 Broadway, R/PSD99, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov) and Andrey V. Lebedev (Inst. of Appl. Phys., Nizhni Novgorod, Russian Federation)

A physical model of stress-strain dynamics and long-time relaxation (slow time) in structured media such as the rock is suggested. It is based on an adhesion mechanism (JKR model) for inter-grain contacts which implies the surface force potential with a barrier. This model results in a unified description of the classical nonlinearity, stress-strain hysteresis, and logarithmic relaxation law for sound velocity and, hence,for the frequency of nonlinear resonance in samples of structured materials. Estimates of a characteristic volume of interacting contacts give close values for a variety of consolidated materials. Propagation of waves in such materials is briefly considered. For a weak (linear) testing wave, the logarithmic relaxation occurs if the classical quadratic nonlinearity is added to the stress-strain relation.

2:15

3pPAb6. A hybrid geostatistical-acoustical model for estimating single-event noise levels from noise monitor data. Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil) and Anthony A. Atchley (Acoust., The Penn State Univ., University Park, PA)

A hybrid geostatistical-acoustical (geo-acoustic) model is proposed as a method for estimating single-event noise levels over a large region from data obtained from a small number of noise monitors. The geo-acoustic model is developed using geostatistical theory and an environmental acoustic-propagation-based regression model. The model is compared to several benchmark models and is evaluated under controlled and simulated meteorological conditions, noise monitor geometries, and areal sensor densities. The results show that it is possible to obtain accurate estimates of the SPL and the variance—associated with the SPL estimates—over a large region with a small number of noise monitors. The proposed geo-acoustic model is significantly more accurate than other commonly used spatial interpolation models, especially when there are few noise monitors and when the estimation point is extrapolated from the noise monitor data.

2:30

3pPAb7. Frequency decorrelation of broadband acoustic signals in a turbulent atmosphere. Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 325 Broadway, Boulder, Colorado 80305, vladimir.ostashev@noaa.gov), D. Keith Wilson, and Sergey N. Vecherin (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

The impact of atmospheric turbulence on sound propagation is an important consideration for source localization with acoustic sensor arrays, studies of noise pollution, and the development of new remote sensing techniques. This paper takes as a starting point a set of recently derived, closed-form equations for the spatial-temporal coherence function of a broadband acoustic signal propagating in a refractive, turbulent atmosphere with spatial-temporal fluctuations in temperature and wind velocity. The theory is quite general and enables analysis of many statistical characteristics of the sound field. It has certain advantages in comparison with Monte-Carlo simulations and has already been used to study the spatial-temporal coherence of narrowband signals. In the present paper, this theory is employed to calculate and analyze the frequency decorrelation of broadband acoustic signals for typical regimes of the atmospheric boundary layer: mostly cloudy or sunny, with light, moderate, or strong wind. The results obtained are then used to study the effect of atmospheric turbulence on the temporal broadening of the intensity of an acoustic pulse.

2:45

3pPAb8. A combined finite element/parabolic equation formulation for acoustic wave propagation over and within a rigid porous ground. Hongdan Tao and Kai Ming Li (School of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47906, mmkmli@purdue.edu)

This paper describes a finite-element (FEM) marching scheme for a standard wide-angle parabolic equation (PE) formulation for calculating the sound fields over and within a rigid porous ground. The study has an application for exploring the effect of snow cover on the propagation of horizontal acoustic waves under refractive atmospheric conditions. Instead of using the standard Crank Nicholson method to solve for the finite difference marching scheme, a FEM discretization approach has been advocated in the present study. By using this approach, the boundary conditions, i.e., the continuity of acoustic pressure and velocity, can be incorporated directly at the air/ground and ground/ground interfaces that facilitates the simultaneous calculations of the sound fields over and within the rigid porous ground. The FEM/PE formulation yields a complete set of information on the acoustic pressure and its spatial derivatives at any receiver locations. In addition, the perfectly matched layer (PML) technique has been adopted that simulates a dissipative layer especially designed to absorb sound waves without reflections from the uppermost layers. Numerical simulations have confirmed that the application of such impedance-matched PML has efficiently eliminated reflections and reduced the overall computational time.

Session 3pSA

Structural Acoustics and Vibration, ASA Committee on Standards, and Engineering Acoustics: Structural Health Monitoring II

Tribikram Kundu, Cochair
Civil Eng. & Eng. Mech., Univ. of Arizona, 1209 E. 2nd St., Bldg. # 72, Tucson, AZ 85721

Anthony Croxford, Cochair
Univ. of Bristol, Queens Building, Bristol BS8 1TR, United Kingdom

Invited Papers

1:00

3pSA1. High frequency guided ultrasonic waves for defect monitoring. Paul Fromme (Mech. Eng., UCL, Torrington Pl., London Wc1E 7JE, United Kingdom, p.fromme@ucl.ac.uk)

High frequency guided ultrasonic waves offer a good compromise between area coverage and defect detection sensitivity for the health monitoring of critical sections of structures. Using standard ultrasonic transducers with single sided access to the structure, guided wave modes were generated that penetrate through the complete thickness of the structure. The wave propagation and interference of the guided wave modes depends on the frequency thickness product. Laboratory experiments were conducted using a laser interferometer and the wave propagation dependence on the wall thickness reduction verified against theoretical predictions. Measurements were conducted using accelerated corrosion in a salt water bath. From the measured signal changes due to the wave mode interference the wall thickness reduction due to corrosion was monitored. The energy transfer between the plate surfaces results in a good sensitivity for the detection of small defects on both surfaces. Experimentally the reflected wave was measured using standard pulse-echo equipment. Using a combination of evaluation in the time and frequency domain, the defect location and damaged plate side can be accurately determined. The high frequency guided waves have the potential for damage monitoring at critical and difficult to access locations from a stand-off distance.

1:20

3pSA2. Ultrasonic transducers and monitoring methods for high resolution structural health monitoring. Wolfgang Grill (Inst. of Experimental Phys. II, Univ. of Leipzig, Burgweg 8, Koenigstein im Taunus, Hessen 61462, Germany, wg@analogspeed.de)

An overview is presented covering novel methods for structural health monitoring (SHM) including the scientific background of different, lately developed methods, dominantly based on the transport properties of guided ultrasonic waves. Among the issues presented and discussed are mode and velocity selective contact and non contact transducers including testing methods for array transducers allowing, respectively, adapted optimized operation. Principles of operation, including holographic and tomographic imaging and high resolution integral temporal monitoring, are presented. Special attention is given to wideband excitation in combination with spectroscopic and dispersive imaging based on respective data acquisition and evaluation. Furthermore, compensation methods for temperature variations by acoustic thermometry are presented. The advantages and possible shortcomings of locally selective and integral SHM are discussed and examples are presented for aircrafts, civil structures, and their sections.

1:40

3pSA3. Guided wave structural health monitoring using inductively coupled embedded sensors. Anthony Croxford, Chenghuan Zhong, and Paul Wilcox (Mech. Eng., Univ. of Bristol, Queens Bldg., Bristol BS8 1TR, United Kingdom, a.j.croxford@bristol.ac.uk)

Conventional SHM systems typically rely on permanently attached sensor networks glued on to a structure. These add complexity and weight through either a wiring requirement or the use of wireless sensing nodes. This paper reports on an alternative approach whereby the wire between transducer and ultrasonic equipment is replaced by an inductive coupling. This allows a passive small sensor unit to be embedded into a composite component. Here a model is presented to describe the performance and optimization of such a coupling. The resulting sensors are embedded in a CFRP component and shown to exhibit excellent performance. Signal processing to account for the effects of misalignment is described. Finally, the ability of such a system to detect typical impact damage is demonstrated.

2:00

3pSA4. Inspection vs structural health monitoring: Manual ultrasonic thickness measurements compared to monitoring with permanently installed sensors. Frederic B. Cegla, Peter E. Huthwaite, and Michael J. Lowe (Mech. Eng., Imperial College London, NDE Group, Mech. Eng., London SW7 2AZ, United Kingdom, p.huthwaite@imperial.ac.uk)

Corrosion is a major issue in industry and inspection and monitoring for wall thickness loss are important to assess the structural integrity of pipework and process vessels. Manual ultrasonic thickness measurements are widely used; however, they are also notoriously unreliable because of operator errors. Therefore, automated inspection scans and monitoring at fixed locations with permanently installed sensors are becoming more attractive; they help to remove some of the error introduced by operators. However, this raises the question of what the underlying uncertainties of automated ultrasonic wall thickness measurements are. A key contributor to the uncertainty is the surface roughness condition and the authors have been researching this topic for some time. This talk will give an overview of the physics of scattering of ultrasonic waves from rough corroded surfaces. The different scales of roughness will be discussed, and a simulation technique based on the Distributed Point Source Method (DPSM) to model the scattering and its experimental validation will be presented. The need for statistical results makes both the speed and accuracy of the simulation very important. Finally, based on the simulations, results of the estimated ultrasonic measurement errors due to roughness are presented.

Contributed Paper

2:20

3pSA5. Relationship between internal damping and modal shapes due to fracture growth. Jose Villalobos (Mater. Eng., Univ. Autonomous of Nuevo Leon, Av. Universidad S/N,, Ciudad Universitaria, San Nicolas de los Garza, NUevo Leon 66451, Mexico, villalobosluna@gmail.com), Diego Ledezma (Mech. Design, Univ. Autonomous of Nuevo Leon, San Nicolas de los Garza, Nuevo Leon, Mexico), and Moises Hinojosa (Mater. Eng., Univ. Autonomous of Nuevo Leon, San Nicolas de los Garza, Nuevo Leon, Mexico)

It is shown in this work that the modal shapes and internal damping of simple structures such as bars change when a crack is present in the system, due to geometry changes. The amount of change in damping could be related to crack propagation and detected with traditional damping measurement techniques. Several amorphous materials, metals, and ceramics are subjected to a crack with controlled propagation and their internal damping is measured with modal analysis techniques. It is shown there is a potential application of modal analysis techniques for crack detection in simple geometries.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 PLAZA B, 1:00 P.M. TO 2:30 P.M.

Session 3pSC

Speech Communication: Communication Disorders

Alexander L. Francis, Chair
Purdue Univ., SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Contributed Papers

1:00

3pSC1. The effects of speech compression algorithms on the intelligibility of dysarthric speech. Rene L. Utianski (Dept. of Speech and Hearing Sci., Arizona State Univ., Coor Hall 2211, 10th St. and Myrtle, P.O. Box 870102, Tempe, AZ 85287-0102, rutiansk@asu.edu), Steven Sandoval, Visar Berisha (Dept. of Elec. Eng., Arizona State Univ., Tempe, AZ), and Julie Liss (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Reduced vowel space area has been demonstrated in dysarthria, as a result of a variety of neurodegenerative diseases and subsequent decreased motor control. The effects of this reduced vowel distinctiveness on intelligibility has been studied extensively in speakers who produce less acoustically contrastive vowels, resulting from dysarthria, or even speaking casually. Past results have demonstrated that when examining such speech, listeners better identify acoustically distinctive vowel tokens. Given this, and the notion of expanded vowel space facilitating vowel identification, speech compression algorithms will be utilized to heighten the distinction between tense and lax vowels. Both connected speech and forced-choice hVd context will be tested and the effects on vowel identification and confusion will be assessed. Given the importance of vowels to overall communication, the proposed method may offer improved intelligibility when speakers cannot improve their speech production. Results bear on the possible utility of exaggerated speech compression algorithms as an augmentative communication tool for individuals with motor speech disorders.

1:15

3pSC2. Predictions from"speech banana"and audiograms: Assessment of hearing deficits in Thai hearing loss patients. Nittayapa Klangpornkun (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., 99, Moo 18, Phaholyothin Rd., Klongnung, Klongluang, Pathumthani 12120, Thailand, nittayapa@gmail.com), Chutamanee Onsuwan (Dept. of Linguist, Faculty of Liberal Arts, Thammasat Univ., Pathumthani, Thailand), Charturong Tantibundhit (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Pathumthani, Thailand), and Pittayapon Pitathawatchai (Dept. of Otorhinolaryngology, Faculty of Medicine, Thammasat Univ., Pathumthani, Thailand)

“Speech banana” is a banana-shaped plot of speech power distribution, where the abscissa and ordinate represent frequency and intensity. By superimposing speech banana over an audiogram, tested with pure tones, degrees

of gain or loss of individual speech sound could be predicted. Speech banana has been constructed for English (Northern and Downs, 1984) and Swedish (Liden and Fant, 1954); however, none has been proposed for tonal languages, such as Thai. This work presents a construction of speech banana for Thai, a language with 21 consonants and 5 lexical tones. Specifically, intensity of each phoneme in the speech banana was calculated by differences of sound pressure level between the local maxima of power spectral density and equal loudness contour at 0 dB. Distribution of the 21 consonants is around 170-5700 Hz and 25-65 dB. Predictions of gain or loss of the phonemes from the constructed speech banana and audiograms were evaluated based on perception test results from seven Thai sensori-neural hearing loss patients, where they identified what they heard from a pair of rhyming words (210 stimuli) differing in initial phonemes, equally distributed across phonemes. Interestingly, the results showed high prediction rates of 71.4-85.7% for phonemes predominantly emphasized on frequency below 2000 Hz.

1:30

3pSC3. Effects of vowel duration and increasing dynamic spectral information on identification of center-only and edges-only syllables by cochlear-implant users and young normal-hearing listeners. Catherine L. Rogers, Gail S. Donaldson, Lindsay B. Johnson, and Soo Hee Oh (Commun. Sci. and Disord., Univ. of South Florida, USF, Dept. of Comm. Sci. & Dis., 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

In a previous study, cochlear implant (CI) users' vowel-identification performance was compared to that of young normal-hearing (YNH) listeners. Stimuli included full syllables and two duration-neutralized conditions: center-only and edges-only (silent-center). CI users performed more poorly than YNH listeners overall and showed proportionately larger decrements in performance for partial syllables. Error analyses suggested that at least some CI users rely more heavily on vowel-duration cues than YNH listeners. The present study was designed to test this hypothesis and to determine whether increasing duration of dynamic cues in the edges-only conditions would improve performance, particularly among poorer-performing CI users. Ten YNH listeners and ten adult CI users heard /dVd/ syllables recorded from three talkers. Full syllables were edited to create center-only and edges-only stimuli in which vowel duration cues were or were not preserved, plus edges-only stimuli with different durations of dynamic information. Performance of both groups improved in the duration-preserved condition for center-only, but not edges-only, stimuli. The center-only duration benefit was larger for the CI than for the YNH group. Increasing the duration of dynamic information in the silent-center stimuli improved vowel-identification performance for both groups. Individual differences among CI users and implications for listener-training programs will be discussed.

1:45

3pSC4. Communicative intent and affect in mothers' speech to hearing-impaired infants with cochlear implants. Maria V. Kondaurova, Tonya R. Bergeson (Otolaryngol. – Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr. – RR044, Indianapolis, IN 46202, mkondaur@iupui.edu), and Christine Kitamura (MARCS Lab., Univ. of Western Sydney, Penrith, NSW, Australia)

Emotional properties of infant-directed speech influence normal-hearing (NH) infants' attention to speech sounds. The current study examines communicative intent/affect in speech to hearing-impaired (HI) infants following the first year of cochlear implantation. Mothers of HI infants (HI group, ages 13.3—25.5 months), NH age-matched infants (NH-AM group, ages 13.5—25.7 months) and NH experience-matched infants (NH-EM group, ages 2.3—3.6 months) were recorded playing with their infants at three sessions over the course of one year. 25-second speech samples were low-pass filtered, leaving pitch but not speech information intact. Twelve adults rated stimuli along five scales of communicative intent/affect: Positive/Negative Affect, Intention to Express Affection, Encourage Attention, Comfort/Sooth and Direct Behavior. ANOVAs demonstrated main effects of Group and/or Session for all scales (ps = 0.01 to 0.07). Speech to HI and NH-EM infants was more positive, affective, encouraging, and comforting than speech to NH-AM infants. Mothers decreased affective (NH-EM group) and comforting (HI group) speech qualities over three sessions but increased directive behavior (NH-EM group). The results suggest that affective properties are modified in speech to HI infants depending on their hearing experience rather than chronological age. Mothers adjust these properties to their infant's developmental stage over the 12-month period.

2:00

3pSC5. Feature divergence of pathological speech. Steven Sandoval (School of ECEE, SenSIP Ctr., Arizona State Univ., 2323 E Apache Blvd., Apt. 2120, Tempe, AZ 85281, ssandova@gmail.com), Rene Utianski, Visar Berisha, Julie Liss (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ), and Andreas Spanias (School of ECEE, SenSIP Ctr., Arizona State Univ., Tempe, AZ)

Many state of the art speaker verification systems are implemented by modeling the probability distribution of a feature set using Gaussian mixture models. In these systems, a decision is made by comparing a likelihood of an observation using both a Gaussian mixture model corresponding to an individual, and a Gaussian mixture model universal back ground model. In this study we propose to use a similar framework to instead characterize the divergence of the feature set distribution between healthy and pathological speech. We accomplish this by determining the difference between a universal background model trained on healthy speech and model of an individual's pathological speech. There are several known methods to evaluate the difference between two probability distributions, one example being the Kullback-Leibler divergence. By building a universal background model using healthy speech, we hope to capture the expected distribution of our feature space. Then by computing a difference between a dysathric individual's feature distribution, and the universal background model, we can determine the features that are most likely to capture the effects of a specific motor speech disorder.

2:15

3pSC6. The functional impact of incidental orofacial muscle activity. Lauren R. Johnson, Nancy L. Potter, and Mark VanDam (Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Repetitive use of specific muscle groups is known to increase both strength and the ability to sustain muscle activity (i.e., endurance) of those muscle groups. Certain orofacial muscles are necessarily recruited in the course of playing a brass instrument, and thus regular performers may incidentally gain strength and endurance in the orofacial muscles used to perform. To test this possibility, 16 skilled trumpet players and 16 non-playing controls contributed strength and endurance (at 50% of maximum strength) measures for buccal, lingual, and labial muscle groups. Results indicate that trumpet players had greater cheek strength and lip endurance, but there were no differences between test and control groups for tongue strength or endurance. Findings suggest that incidental orofacial muscle activity may have a positive functional impact on orofacial muscle strength and/or endurance. This finding supports a clinically useful, objective measure for diagnosis and may be useful for functional rehabilitation for patients with orofacial disorders including those with Bell's palsy, complications associated with otitis media, acoustic neuromas, or other facial- or cranial nerve damage due to surgery, trauma, or disease.

Session 3pSP

Signal Processing in Acoustics: Smartphone Acoustic Signal Processing Student Competition (Poster Session)

Kevin Cockrell, Chair
Applied Physical Sciences Corp., 2488 Historic Decatur Rd., Ste. 100, San Diego, CA 92106

Contributed Papers

All posters will be on display and all authors will be at their posters from 1.30 p.m. to 2.30 p.m.

3pSP1. SoundMap: A mobile app for optimizing room acoustics and speaker placement. K. J. Bodon and Zachary Jensen (Phys., Brigham Young Univ., Brigham Young Univ., Provo, UT 84602, joshuabodon@gmail.com)

Improving room acoustics can be expensive and time consuming, even for the most experienced acousticians. SoundMap ™ is a user friendly app that can be used to easily model rooms based upon acoustic properties. A combination of the method of images, ray tracing, and modal analysis will be used to calculate steady state levels for different frequency bands in a room. The modeling will also take several other factors into account, including the room impulse response, absorption coefficients of boundary materials, furniture, and other room elements. The model will incorporate generic loudspeaker systems, with the future design goal of integrating specific loudspeaker systems. Through a contour plot, the user will be able to test the effects of loudspeaker location, loudspeaker type, room layout, and room materials, with the aim of optimizing the listening experience.

3pSP2. Ping-pong: Using smartphones to measure distances and relative positions. Jorge Herrera and Hyung-Suk Kim (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, jorgeh@ccrma.stanford.edu)

A novel system for real-time range and geometry estimation of a group of smartphones co-located in a shared physical space is presented. The system uses off-the-shelf devices and employs audible signals to estimate inter-device (pair-wise) distances. Coordinated sound synthesis and processing of a pair of pitched sounds allows to estimate the distance between two devices based on the travel time. By independently analyzing different harmonics of the sounds used in the measurement, a more robust and precise distance estimation is achieved. To overcome the absence of a centralized clock to coordinate measurements, a synchronous communication channel was used. When four or more devices are present, it is possible to estimate their relative positions in a three-dimensional space, by minimizing an equation error norm. The system works both on closed and open spaces. We believe that such system opens the possibility for new ways of interaction that could benefit musical expression, social interaction and gaming.

3pSP3. Speech assist: An augmentative tool for practice in speech-language pathology. Rene L. Utianski (Dept. of Speech and Hearing Sci., Arizona State Univ., Coor Hall 2211, 10th St. and Myrtle, P.O. Box 870102, Tempe, AZ 85287-0102, rutiansk@asu.edu), Steven Sandoval (Dept. of Elec. Eng., Arizona State Univ., Tempe, AZ), Nicole Lehrer (Dept. of Arts, Media, and Eng., Arizona State Univ., Tempe, AZ), Visar Berisha (Dept. of Elec. Eng., Arizona State Univ., Tempe, AZ), and Julie Liss (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Dysarthria affects approximately 46 million people worldwide, with three million individuals residing in the US. Clinical intervention by speech-language pathologists (SLPs) in the United States is supplemented by high quality research, clinical expertise, and state of the art technology, supporting the overarching goal of improved communication. Unfortunately, many individuals do not have access to such care, leaving them with a persisting inability to communicate. Telemedicine, along with the growing use of mobile devices to augment clinical practice, provides the impetus for the development of remote, mobile applications to augment the work of SLPs. The proposed application will record speech samples and provide a variety of derived calculations, novel and traditional, to assess the integrity of speech production, including: vowel space area, assessment of an individual's pathology fingerprint, and identification of which parameters of the intelligibility disorder are most disrupted (e.g., prosody, vocal quality). The individualized selection of desired information for incorporation into a report template will be available. The reports will mimic those generated manually by SLPs today. The automation of this assessment will allow SLPs to treat patients remotely, allowing for the widespread, worldwide impact of high skilled assessment, something currently lacking in underdeveloped parts of the world.

3pSP4. SpeakerLab: A mobile app for measuring loudspeaker parameters and modeling enclosures. Zachary Jensen and K. J. Bodon (Brigham Young Univ., 934 E 300 S, Provo, UT 84606, zjens1@gmail.com)

Thiele-Small parameters for loudspeaker drivers are essential in modern loudspeaker design. While they are typically given in specifications by a driver manufacturer, their values can vary considerably. SpeakerLab is a loudspeaker parameter measurement and modeling tool, an all-inclusive app for the loudspeaker designer. Using a specially designed cable with a known impedance, a user will be able to plug their mobile device directly into a loudspeaker driver and measure its Thiele-Small parameters using the added mass or known volume method. With the correctly measured parameters, an enclosure can be appropriately modeled and optimized.

3pSP5. "Tone deaf"— The touch based ear-training game. Andrew T. Pyzdek (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

Inspired by the color matching game "Colorblind" by ChewSoft, Tone Deaf is the new game that challenges players to match the pitch and timbre of a note by touching the screen to draw a single cycle of the waveform. The shape of the waveform is then used to determine the appropriate harmonics and their phases, while the length of the cycle determines the frequency. Players will progress through multiple levels of difficulty, starting with simple waveforms such as sine waves, sawtooth waves, and square waves, and finally emulating the complex patterns of instruments such as the piano, trumpet, and violin.

3pSP6. The echolocating phone app. David A. Hague and Saurav Tuladhar (Elec. and Comput. Eng., Univ. of Massachusetts, Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, david.a.hague@gmail.com)

Acoustic echolocation systems have a multitude of applications including test/measurement, noise cancellation, and object detection. Current smartphone technology possesses the necessary computing power to implement sophisticated echolocation systems on smartphones. This provides the smartphone user

with the ability to perform complicated acoustic measurement tasks in any situation and environment. This project implements an active sonar system on an Android smartphone. The sonar system has three main modes of operation: object detection and ranging, object velocity estimation, and joint object range and velocity estimation. Each of these operational modes are accomplished by utilizing one of the phone's speakers as a transmitter and one or more of the microphones as a receiver. The app employs a simple Graphical User Interface that allows the user to switch between the modes of operation and observe/analyze object data. Additional functionality may include the ability to write object data to various data formats for offline processing and analysis. Because this app employs several modes of operations, there are several potential commercial applications including measuring a room's acoustic impulse response or a vehicle speed gun as well as many educational apps exploring various acoustic phenomena.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 CONTINENTAL 4, 3:30 P.M. TO 5:30 P.M.

Plenary Session, Annual Meeting, and Awards Ceremony

James H. Miller, President
Acoustical Society of America

Annual Meeting of the Acoustical Society of America

Presentation of Certificates to New Fellows

Judit Angster – for contributions to the acoustics of the pipe organ

Benjamin A. Cray – for contributions to underwater directional sensing

Kevin D. Heaney – for contributions to ocean acoustic modeling and inversion methods

Marcia J. Isakson – for contributions to modeling shallow water acoustic propagation using the finite element method

Tribikram Kundu – for contributions in nondestructive testing and evaluation

Robert M. Koch – for contributions to the hydroacoustics and structural acoustics of underwater systems

Michael V. Scanlon – for contributions to high speech imaging of fine scale acoustic phenomena

Clark S. Penrod – for service to the Society and leadership in underwater acoustics

Gopu R. Potty – for contributions to ocean acoustic inversion methods in shallow water

Kausik Sarkar – for contributions to the modeling of ultrasound microbubbles

Natalia A. Sidorovskaia – for contributions in research and education in underwater acoustics and animal bioacoustics

U. Peter Svensson – for contributions to room acoustics edge diffraction modeling

Jing Tian – for leadership in promoting American-Chinese cooperation in acoustics

Award Announcements and Presentations of Awards

Announcement of the 2013 Munk Award to W. Steven Holbrook
granted jointly by The Oceanography Society, the Office of Naval Research, and
the Office of the Oceanographer of the Navy

ASA Student Council Mentoring Award to Barbara G. Shinn-Cunningham

Rossing Prize in Acoustics Education to Juliette W. Ioup

Silver Medal in Biomedical Acoustics to Kullervo H. Hynynen

Silver Medal in Musical Acoustics to William J. Strong

Session 3eED

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved

Tracianne B. Neilsen, Cochair
Brigham Young Univ., N311 ESC, Provo, UT 84602

Marcia J. Isakson, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713

This workshop for San Francisco area Girl Scouts (ages 12–17) consists of a hands-on tutorial, interactive demonstrations, and panel discussion about careers in acoustics. The primary goals of this workshop are to expose the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please email Traci Neilsen (tbn@byu.edu) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. – 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. – 7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

WEDNESDAY, 4 DECEMBER 2012 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings beginning at 8:00 p.m. and on Wednesday evening beginning at 7:30 p.m.

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

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Golden Gate 2/3
Signal Processing in Acoustics	Continental 1