Session 4aAA

Architectural Acoustics: Speech Intelligibility in Reverberation and Noise

Roger W. Schwenke, Chair
Meyer Sound Laboratories, 2832 San Pablo Ave., Berkeley, CA 94702

Chair's Introduction—8:25

Invited Papers

8:30

4aAA1. Speech intelligibility studies in a historic multipurpose room. Ana M. Jaramillo (Olson Sound Design, LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu), Peggy B. Nelson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Bruce C. Olson (AFMG Services North America, LLC, Brooklyn Park, MN)

The Speech-Language-Hearing Sciences department at the University of Minnesota is located in Shevlin Hall, a historic building dating from the 1920s. Shevlin Hall has a large room on the first floor with a high coffered ceiling that is often used for receptions and presentations but also serves as a classroom. Students with hearing loss often complained about attending classes in Shevlin 110 due to its high reverberation times, combined with window AC unit noise, resulting in very degraded speech intelligibility. In 2013, the room went through renovations that addressed room acoustics, sound system design, as well as lighting and equipment (not including AC). Post-renovation measurements and subjective impressions have shown a significant improvement in speech intelligibility in the room. For this paper, we are looking at subject-based intelligibility tests to compare with prediction metrics and simulations, as well as collecting a qualitative narrative on the department's impression on the room before and after renovations.

8:55

4aAA2. A comparison of speech coherence index and measured speech intelligibility. Tobi A. Szuts and Roger W. Schwenke (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94610, tobi@meyersound.com)

Perceptual tests were taken in real and simulated spaces to determine whether Speech Coherence Index (SCI) predicts human performance better than Speech Transmission Index (STI). SCI is a proposed method of estimating speech intelligibility during event conditions, that is, in real time with program material. The complex valued coherence function is used to estimate the signal-to-noise ratio on a per frequency basis, and is calculated with short time windows at high frequencies and longer time windows at low frequencies to mimic the multi-resolution nature of human hearing. SCI has been shown to track STI very closely under simplified conditions. Under more realistic conditions, SCI is more sensitive to reverberant energy and is always less than STI. Under certain conditions, such as long source to listener distance or low SNR, SCI predicts significantly lower intelligibility values than STI.

9:20

4aAA3. Acoustical renovation and sound system design in a church in buenos aires, argentina. Fernando M. del Solar Dorrego (Elec. Eng., Instituto Tecnológico de Buenos Aires, 445 Waupelani Dr., Apt. B10, State College, PA 16801, fsolar@gmail.com) and Pablo Gardella (Elec. Eng., Instituto Tecnológico de Buenos Aires, Buenos Aires, Ciudad de Buenos Aires, Argentina)

Church "Nuestra Señora de la Paz" suffered from an excessive reverberation (4 seconds at mid-frequencies) and an ill-suited sound reinforcement system. Measurements of the Speech Transmission Index (STI) in the congregation area indicated that speech intelligibility was greatly impaired. Thermal insulation on the roof was insufficient and impacted in air conditioning costs. Renovations included the design of a sound-absorbing and thermally insulating covering in the roof area. Subsequent measurement of room acoustic parameters showed that the proposed goals were met. A new sound system, which increased speech intelligibility and audio quality substantially, was designed and tested. The possible appearance of acoustical defects after the renovation, such as echoes and flutter echoes, was studied using objective tests on the measured Room Impulse Responses (RIRs).

9:45

4aAA4. Study of architectural design & acoustic conditions of primary schools in Singapore. Siu Kit Lau and Geok Ling Lee (Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Dr., Singapore 117566, Singapore, slau@acousticsresearch.com)

Urban noise problems are a growing concern in rapidly urbanizing cities such as Singapore. Studies have revealed that children are particularly susceptible to the harmful effects of noise. Children in Singapore schools are particularly susceptible considering the long hours spent in naturally ventilated classrooms. This study aims to assess the architectural design and planning of schools in Singapore based on their acoustic qualities. External ambient noise, façade sound insulation, and reverberation time were measured in three schools

4a THU. AM

about the architectural design to characterize their acoustic environments and seek areas for improvements. It is revealed that all the classrooms studied experience internal noise levels above the guideline values recommended by the World Health Organization (WHO) of 35 dB, suggesting poor acoustic conditions and harmful effects on children's' health, behavior, and learning abilities. Reverberation time standards were only met by the classrooms in one school out of three, implying that speech intelligibility and learning performance are hindered elsewhere. Two overseas case studies are explored to demonstrate how better acoustic environments can be achieved through planning and design in unbuilt schools and abatement strategies for built schools. Architects and planners can adapt and incorporate them into the design of primary schools in Singapore.

10:10–10:25 Break

Contributed Papers

10:25

4aAA5. Active learning classroom configurations and the effects on speech intelligibility. Edwin S. Skorski (Interior Design, Central Michigan Univ., CMU - Human Environ. Studies, EHS Bldg. 228, Mount Pleasant, MI 48859, skors1es@cmich.edu)

For high academic achievement, it is critical that the acoustic environment of classrooms and other learning spaces provide the proper conditions for good speech communication between teacher and students. While there is a significant body of research regarding traditional classroom seating arrangements, less is known regarding the acoustical impacts of "active learning classroom" configurations. Active learning classrooms are typically furnished with mobile tables and chairs that are easily reconfigured in a variety of learning arrangements. Additionally, in an active learning environment, there is typically no fixed position from which the teacher will lecture. While active learning spaces provide flexibility and mobility, which is advantageous for student engagement, they also create a complex acoustic environment where both source and receiver positions have a continually shifting spatial relationship. Using design guidelines found in ANSI/ASA S12.60-2010/Part 1, this study will use computer simulated environments to explore various active learning space configurations and compare the results to generally accepted acoustical performance criteria for RT, C80, STI, and RASTI.

10:40

4aAA6. Results from the summer sound lab: Exploring acoustical options for a multiuse space on campus. Daniel Butko, Zachary Maggia, Collin Abdallah, and Lindsey Trout (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

This paper summarizes the progress of an active multiphase acoustical research project emphasizing implementation of adaptive interactive acoustical responses in multiuse spaces to transform unhealthy learning environments. A multiuse space on campus, which suffers from poor speech intelligibility as a result of excessive reverb and flutter echo revealed through finish materials, room volume, and shape, functions as an architectural sound lab and location for data collection. The project, launched Spring 2017, will span multiple phases and disciplines for data collection, design, prototyping, validation, and construction. Faculty and students are collecting and analyzing acoustical data, adding to a literature review and database of comparable acoustical challenges and results, and validating findings in both computer and physical model form. Collected data will lead researchers to design applications for adaptive and resilient learning environments with minimized acoustical distractions and improved speech intelligibility. Their work will culminate in a final full-scale built form, providing scholarly merit applicable to similar spaces and/or functions. Established industry partnerships will provide development of physical panel-based prototyping, followed by acoustical testing, full-size fabrication, and installation for public interaction and academic validation.

10:55–11:40 Panel Discussion

Session 4aAB

Animal Bioacoustics: General Biosonar

Rolf Müller, Chair
Mechanical Engineering, Virginia Tech, ICTAS Life Sciences District (Mail Code 0917), Blacksburg, VA 24061

Contributed Papers

8:00

4aAB1. Bone conducted sound in a dolphin's mandible: Experimental investigation of elastic waves mediating information on source localization. Michael Reinwald (Lab. for Biomedical Imaging, Université Pierre-et-Marie-Curie, 15 rue de l'Ecole de Médecine, Laboratoire d'Imagerie Biomédicale, Paris 75006, France, michael.reinwald@upmc.fr), Jacques Marchal (Institut d'Alembert - UMR 7190, Université Pierre-et-Marie-Curie, SAINT-CYR-L'ECOLE, France), Lapo Boschi (Laboratoire iSTeP, UMR 7193 UPMC-CNRS, Université Pierre-et-Marie-Curie, Paris, France), and quentin Grimal (Lab. for Biomedical Imaging, Université Pierre-et-Marie-Curie, Paris, France)

Marine mammals, such as dolphins, use audition as a tool to navigate and hunt. It is widely accepted that the main auditory pathway for sound reception in a dolphin's skull is through the lower jawbone and the connected fatty tissues—both possibly acting as a waveguide. Elastic waves traveling inside the mandible therefore likely contain information on the position of sound sources. The objective of the work was to investigate to which extent elastic wave signals in a mandible of a common dolphin (*Delphinus delphis*) can be used to localize a source. Experiments were conducted in a water pool by placing a sound source emitting pulses in the range of 10–100 kHz at different angles in the horizontal plane of the mandible. The elastic waves propagating in the mandible were measured at two positions close to the hypothetical ears by piezoelectric discs glued to the bone. We evaluated energy directivity patterns, interaural level differences, and applied a time-reversal localization algorithm to simulate acoustic source localization of the sound source. The findings should give insight into the directivity of the auditory pathway of sound in a dolphin's mandible and will be substantiated by 3D simulations in future work.

8:15

4aAB2. Temporal features of conditioned reduction in dolphin auditory brainstem responses. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

Conditioned changes in the hearing of two bottlenose dolphins were elicited by pairing a 10-kHz conditioned stimulus (i.e., a warning sound) with a more intense, unconditioned stimulus at 80 kHz. Hearing was assessed by measuring transient auditory brainstem responses (ABRs) to tone burst stimuli presented with inter-stimulus intervals drawn from a random distribution with mean interval of 2 ms and 2-ms jitter. Tone burst frequencies were 57, 80, 113, and 133 kHz. ABRs at discrete times within each trial were obtained by synchronously averaging epochs of the instantaneous electroencephalogram time-locked to tone burst onsets over 2 to 3-s time intervals. In initial testing with one dolphin, ABRs were reduced after the conditioned stimulus but before the unconditioned stimulus, demonstrating a conditioned suppression of the ABR. In subsequent testing, and in all testing with the second dolphin, ABRs were suppressed (relative to baseline values) throughout the entire 31-s trial. When the unconditioned stimulus was removed, ABRs returned to baseline values, demonstrating extinction of the conditioned response. The results support the hypothesis that some toothed whales can "self-mitigate" the effects of noise if warned of an impending exposure. [Work supported by SSC Pacific NISE Program.]

8:30

4aAB3. Range perception in dolphins: Discrimination of sounds with distance-dependent features other than amplitude. Carolyn E. Schlundt (Government IT Services / US Navy Marine Mammal Program, 4045 Hancock St., Ste. 210, San Diego, CA 92110, carolyn.melka@harris.com), Jason Mulsow, Ryan A. Jones (National Marine Mammal Foundation, San Diego, CA), and James J. Finneran (U.S. Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr. Pacific, Code 71510, San Diego, CA)

Although sound pressure level (SPL) is typically used to estimate behavioral responses of marine mammals to anthropogenic sound, other factors, such as source distance, may play a role in an animal's ability to assess the inherent risk of a sound (and thereby affect its response). Factors that contribute to range perception include a decrease in received SPL due to spreading loss and absorption, a relative decrease in the amplitude of high-frequencies from frequency-dependent differences in absorption, and the presence of reverberation due to boundary interactions. In the current study, three bottlenose dolphins (*Tursiops truncatus*) learned to discriminate tones with simulated reverberation and high-frequency absorption (*degraded stimuli*) from tones without those features (*undegraded stimuli*). Tones were frequency modulated, with fundamental frequencies of approximately 10 kHz, and roving SPLs of 120 dB ± 10 dB. A progression of experiments examined the dolphins' performance when presented with novel, degraded stimuli containing either reverberation or high-frequency attenuation. The results provide experimental evidence that marine mammals can differentially classify sounds with similar SPLs, but differing transmission-related acoustic cues of reverberation and frequency-dependent absorption, and may use signal degradation characteristics to judge the range of acoustic sources. [Work supported by US Fleet Forces Command.]

8:45

4aAB4. Analysis of bats' gaze and flight control based on the estimation of their echolocated points with time-domain acoustic simulation. Taito Banda (Faculty of Life and Medical Sci., Doshisha Univ., 1-3, Miyakotani, Tatara, Kyotanabe, Kyoto 610-0394, Japan, dmq1001@mail4.doshisha.ac.jp), Miwa Sumiya (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe-shi, Japan), Yuya Yamamoto (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), Yasufumi Yamada (Organization for Res. Initiatives and Development, Doshisha Univ., 1-3, Miyako-tani, Tatara, Kyotanabe, Kyoto, Japan), Yoshiki Nagatani (Dept. of Electronics, Kobe City College of Technol., Kobe, Hyogo, Japan), Hiroshi Araki (Adv. Technol. R&D Ctr., Mitsubishi Electric Corp., Amagasaki, Japan), Kohta I. Kobayasi (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), and Shizuko Hiryu (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Bats detect objects by echolocation. With those echoes, they can describe the shape of objects as well as position, and then approach or avoid them. To reveal which part of objects bats gaze at, we simulated the echoes that bats hear, and estimate the bats' echolocated points by time-domain

2D-acoustic-simulation based on the behavioral data during obstacle-avoiding flight. First, we constructed a microphone array system in an experimental chamber and attached telemetry-microphones to Japanese horseshoe bats. With that measurement system, we obtained the timing, positions and directions of emitted pulses during obstacle-avoiding flight. Using these data, we simulated the echoes returning from the obstacles (acrylic boards) and investigated how the bats show spatial and temporal changes in the echolocated points of objects as they became familiar with the space. In a comparison between before and after the habituation in same obstacle layout, there are differences in the wideness of echolocated points on objects. By flying same layout repeatedly, their echolocating field become narrower. This study suggests the help of acoustic simulation to understand the way bats see the world. [This research was supported by Scientific Research on Innovative Areas of JSPS, and the JST PRESTO program.]

9:00

4aAB5. Doppler shifts in bat biosonar: The good, the bad, and the unexpected. Rolf Müller, Xiaoyan Yin, and Peiwen Qiu (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu)

Most models for explaining bat biosonar are based on linear systems theory. A notable exception are frequency shifts due to the Doppler effect that occur whenever there is relative motion between the wave emitter and receiver. Bat species such as horseshoe bats, Old World leaf-nosed bats, and mustached bats have evolved particularly sophisticated biosonar systems that are able to exploit Doppler shifts for detecting prey in clutter. In this case, "good Doppler shifts" induced by the wing beat velocity of an insect prey give the prey echoes unique signatures that stand out among echoes from stationary targets such as reflecting facets in dense vegetation. Since these "good Doppler shifts" are small, bats exploiting them had to evolve specialized biosonar mechanisms to detect them. However, these mechanism are also sensitive to "bad Doppler shifts" that result, e.g,. from a bat's own flight velocity. It has been generally assumed that bats with Doppler-based biosonar will eliminate such nuisance Doppler shifts through compensation behaviors to allow them to detect the good Doppler shifts. However, a complete picture of Doppler shifts in bat biosonar systems may also have to account for other sources of Doppler shifts that do not fall into the above categories.

9:15

4aAB6. Integration of dynamic emission and reception in the biosonar system of horseshoe bats. Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Tech, 3719 Parliament Rd., Apt. 22, Roanoke, VA 24014, josephs7@vt.edu), Hiroshi Riquimaroux (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Jinan, Shandong, China), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bats have evolved unique mechanisms by which to navigate and hunt in their environments. Some species of the microchiroptera, such as members of the horseshoe bat family (Rhinolophidae), which live in particularly challenging, cluttered environments, have been observed to move their peripheral structures for emission and reception (noseleaf and pinna) during echolocation very rapidly, e.g., within a tenth of a second. With these behaviors, the bats could create time-variant properties in their biosonar systems to provide additional degrees of freedom to be used for enhancing the encoding of sensory information. To make full use of this opportunity, a tight sensorimotor integration is likely to be required. In order to explore the possible role of sensorimotor integration in a dynamic biosonar system, we have developed biomimetic robotic models to replicate the effects of the dynamics of the baffle structures. Data obtained with these systems indicates that a coordinated emitter and receiver dynamics can both contribute to the embedding of time-variant signatures in the signals that encode information about target class. Ongoing research seeks to transition insights from work on the biomimetic design paradigms to data from brain recordings made in from the auditory brainstem and the superior colliculus of horseshoe bats.

9:30

4aAB7. Quantification of fast pinna motions in rhinolophid and hipposiderid bats. Xiaoyan Yin, Peiwen Qiu, and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, xiaoyan6@vt.edu)

As part of their biosonar behaviors, rhinolophid (family Rhinolophidae) and hipposiderid bats (family Hipposideridae) both show conspicuous motions of their outer ears (pinna). These motions coincide with pulse emission and echo reception in time and could hence have a functional relevance for the encoding of sensory information. However, a quantitative in-depth characterization of these motions is still needed to derive detailed hypotheses for their function. To accomplish this, dense sets of landmark points have been placed on the pinna to provide for dense spatial coverage of its surface over the course of a motion cycle. Occlusion-free stereo tracking of the landmarks was accomplished with an array of four high-speed video cameras. Customized methods based on motion prediction have been used to track landmark points across video frames. The results have been used to construct accurate, continuous estimates of pinna surface motion. These estimates reveal that the pinna surface is subject to heterogeneous patterns of displacements and velocities within each motion cycle. Experiments with simplified robotic reproductions to understand the acoustic implications of these pinna surface motions are currently in progress. Once the signal transformations that result from the pinna motions are understood, the question of functional relevance can be addressed.

9:45

4aAB8. Noseleaf motions impart dynamic signatures on bat biosonar pulses. Liujun Zhang, Ru Zhang (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Shanda South Rd. 27, Jinan 250100, China, sdujune@gmail.com), Shuxin Zhang (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Jinan, Shandong, China), Luhui Yang, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Old-World leaf-nosed bats (Hipposideridae) are echolocating bats with nasal biosonar pulse emission. The nostrils in these animals are surrounded by baffle shapes ("noseleaves") that have been shown to function as beamforming devices. In addition to their elaborate static geometries, the noseleaves can also undergo non-rigid shape changes that coincide with biosonar pulse emission. Prior work by the authors based on biomimetic baffle shapes has demonstrated that baffle motions similar to the ones seen in bat noseleaves can result in time-variant device properties, i.e., distributions of signal power over direction and frequency that also change with time. However, it remained to be established whether similar effects occur in bats. To answer this question, biosonar pulse emission in hipposiderid bats (*Hipposideros armiger* and *Hipposideros pratti*) has been studied with synchronized arrays of high-speed video cameras and ultrasonic microphones. Noseleaf motion was characterized by tracking the position of nine landmark points on the noseleaf. Temporal variability in the distribution of signal energy across spatial directions and ultrasonic frequencies was quantified using pair-wise measures of similarity applied to time-windowed segments of the pulse waveforms. Pulses that were accompanied by unequivocal noseleaf motions showed a significantly larger variability across time than those that were not.

Session 4aAO

Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics: Biological Effects on Seabed Geoacoustic Properties I

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, 10000 Burnet Road, Austin, TX 78758

Megan S. Ballard, Cochair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, 10000 Burnet Road, Austin, TX 78758

Kelly M. Dorgan, Cochair
Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528

Chair's Introduction—8:00

Invited Papers

8:05

4aAO1. Some startling effects of biologic factors on seafloor physical properties. Thomas F. Wever (WTD71, Eckernförde, Schleswig-Holstein, Germany) and Chris J. jenkins (INSTAAR, Univ. of Colorado, Boulder, 4001 discovery drv, Boulder, CO 80309-0450, jenkinsc0@gmail.com)

Recently, the biological impact on marine geophysics has received much interest. The question now is What types of impacts, how intense, and how changing? How can these impacts be understood and quantified, and then enacted in our practical predictive systems? Two dramatic examples from NW Europe. The acoustic response of the sandy seabeds is being altered by dense populations of species such as the invasive Ensis (i.e., a change of seabed properties). The same species forms a baffle that alters the seafloor response to flows, causing sediment trapping (i.e., a change of processes). At the sea surface, massive drifting rafts of detached bladder algae change the acoustic response of the sea surface and can attenuate surface waves. The rafts have seasonal and longevity attributes and can be tracked remotely, so effects on long-range underwater communications may be manageable. The Volkswagenstiftung, Office of Naval Research, NATO, and others have understood the significance and started to finance meetings, projects, and expert teams. To improve prediction capabilities, especially before at-sea operations, a close cooperation of biology and physical sciences is necessary. This can be achieved by two parallel approaches: (1) make existing bio- and geo-information compatible and (2) design new combined bio-geo experiments.

8:25

4aAO2. The trait-based approach as a powerful tool to study the bio-geo seafloor system. Anna Törnroos (Environ. and Marine Biology, Åbo Akademi Univ., BioCity, Tykistönkatu 6, Turku 20520, Finland, anna.m.tornroos@abo.fi)

One of the major causes of heterogeneity on the seabed is the biology. The presence of organisms creates voids and frameworks within and on the sediment, and their behavior may layer or sort the entire seafloor. Making use of the biological information would be powerful for improving acoustics. Likewise, embracing, to a greater degree, acoustic techniques and measurements to understand the biology would be favorable. However, to tackle this bio-geo diversity in a cross-disciplinary way requires a common language and approach. Here, I present and propose the trait-based approach as a way forward. Because there are simply too many species to describe and include in one model, reducing this complexity is essential and can be done by considering individuals characterized by a few key characteristics, or *traits*. Relevant biological traits span morphology, behavior and life-history of organisms and can be applied on single individuals and scaled up to whole communities, incorporating the density of organisms. By exemplifying the progress in benthic ecology, I outline where we currently stand, possible key traits of value for both fields and ideas on how to progress.

8:45

4aAO3. Relating functional diversity of infauna and sediment geoacoustic properties. W. Cyrus Clemo and Kelly M. Dorgan (Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528, wcc1622@jagmail.southalabama.edu)

Marine sediments cover most of the seafloor and vary in physical structure from unconsolidated gravel or sand grains to cohesive, gel-like muds. Sediments also provide a habitat to diverse and abundant animals (infauna). Infauna are impacted by but can also affect the sediment's physical and biogeochemical properties and fluid flow through and across the sediment. Through activities such as burrowing, feeding, and construction of tubes and rigid body parts, many infauna act as ecosystem engineers, directly and indirectly creating and changing their habitats. Characterization of infauna using functional groups based on how their activities affect their sediment environments can simplify the broad diversity of infauna. For example, some infauna create habitat structure in addition to modifying surface topography through tube construction. Others change sediment properties through bioturbation (animal-induced particle mixing) and bioirrigation (fluid mixing) as they burrow. Structure creation and bioturbation/irrigation can affect sediment geotechnical properties such as porosity, bulk density, grain size heterogeneity, fracture toughness, and stiffness. Changes in these properties can influence acoustic wave speed, attenuation, and backscatter. Thus, acoustics tools can be used to observe and quantify animal-sediment interactions relatively noninvasively by relating the activities of different functional groups with changes in acoustic properties of sediments.

9:05

4aAO4. Biogenic control on rheological and geoacoustic properties of aquatic sediments: Past research and future directions.
Samuel J. Bentley (Geology and Geophys. and Coastal Studies Inst., Louisiana State Univ., E235 Howe Russell GeoSci., Dept. of Geology, LSU, Baton Rouge, LA 70803, sjb@lsu.edu)

Our understanding of biogenic controls on rheological and geoacoustic properties of aquatic sediment lags behind other topics in seafloor science. For nearly a century, the ability of sedimentary infauna to influence fabric and diagenesis of aquatic sediments has been widely recognized. These insights result from many studies (both observation and modeling) of bioturbation and its control on mass transport and destruction of primary sedimentary layering in the shallow seabed. Nevertheless, our detailed knowledge infaunal behavior below the intertidal zone remains very limited for most taxa. Our understanding of infaunal controls on sediment physical properties and rheology such as shear strength, porosity, or sound-wave propagation below the intertidal zone remains even more sparse. Some pioneering studies over the decades have demonstrated that biogenic influences on sediment porosity, strength, and rigidity can be significant and are related to bioturbation style and rate. However, the number of these studies remains small compared to the breadth of the problem. Several recent numerical models of muddy-sediment seabed dynamics and consolidation hold promise as frameworks for potential experimentation, in which biogenic forcing can be implemented to dilate, compact, or reshape sediment, and so alter sediment rheological and geoacoustic properties. This presentation will explore potential new approaches in this line of research, based on past successes and present knowledge gaps.

9:25

4aAO5. Fast-marching methods to model the acoustic responses of biological seafloors.
chris j. jenkins (INSTAAR, Univ. of Colorado, Boulder, 4001 Discovery Dr., Boulder, CO 80309-0450, jenkinsc0@gmail.com)

Biologically colonized seafloors have been difficult to model for their acoustic interaction. The distributions of geoacoustic properties above and beneath the nominal bottom can be extremely complex. We have implemented workflows in software which: (i) efficiently construct many patterns of growth forms, bioturbation, organism aggregations, and flesh-skeleton morphologies, and (ii) submit them to “Eikonal” wavefront acoustic modeling. Ray-based finite-element and statistical methods can, in theory, be applied but they run into problems of the strongly inhomogeneous geoacoustics, ray-chaos (of several causes), absence of a clear division into the usual rough surface/inhomogeneous volume halfspaces, and heavy computational costs. Wavefront methods, namely, Eikonals implemented via Fast-Marching Algorithms, are able to deal with complex situations—as has been realized lately in deep seismic geophysics, in moving object tracking, and interestingly, in urban settings, for locating gunshots and mapping Wi-fi fields. They also to offer a new way to understand the acoustic backscatter results from multibeam, sidescan and single beam sonar systems over complex biologic seafloor types. This direction of research requires close knowledge of both the “bio” and the “geo” of seafloors.

9:45

4aAO6. Do environmental stress-induced changes in faunal composition and bioturbation affect sediment geoacoustic properties? A test case in the Gulf of Mexico dead zone.
Kevin Briggs (Div. of Marine Sci., Univ. of Southern Mississippi, 1020 Balch Blvd., Stennis Space Ctr., MS 39529, Kevin.B.Briggs@usm.edu), Michael Richardson (Inst. for Defense Analyses, Alexandria, VA), and Shivakumar Shivarudrappa (Louisiana Universities Marine Consortium, Chauvin, LA)

Four sites on the continental shelf off Louisiana, each subjected to different historical exposures to low concentrations of bottom-water dissolved oxygen, are investigated in terms of their macrobenthos species composition, sediment physical properties, and sediment geoacoustic properties (sound speed, attenuation, and impedance). From macrobenthos species composition, feeding type is identified, which allows categorization of some bioturbation activity as either dilation or compaction of sediment. Dilation and compaction should affect sediment properties of bulk density and porosity, which are significant predictor variables of geoacoustic properties. Different levels of oxygen stress correspond with statistically separable macrobenthos assemblages, abundance and diversity of biogenic structures (burrows and voids), and ratios of dilators to compactors. Sediment sound speed and attenuation values measured in subcores from box cores are compared at each of the four sites with different faunal composition and average bottom-water dissolved oxygen concentrations to test the long-term (historical) effects of different bioturbation regimes on geoacoustic properties. Sediment acoustic properties are regressed on sediment physical properties and compared with established empirical fits from a worldwide database to identify possible anomalies. The impedance of surficial sediment is measured at each site with the Acoustic Sediment Classification Profiler to examine possible large-scale (side-wide) effects on geoacoustic properties.

10:05–10:25 Break

10:25

4aAO7. Time-dependent seafloor roughness simulations using a coupled spectral ripple-biological decay model. Allison Penko and Joseph Calantoni (Marine GeoSci. Div., U.S. Naval Res. Lab., 1005 Balch Blvd., Code 7434, Stennis Space Ctr., MS 39529, allison.penko@nrlssc.navy.mil)

Ever changing hydrodynamic conditions in the nearshore continuously affect biological communities and sediment features on the seafloor. Seafloor sediments have complicated interactions with the co-existing biological flora and fauna on the seabed. For example, ripple formation by waves and ripple degradation from biological activity are opposing forces occurring continuously. Simulating this complex natural environment is challenging due to these dynamic interactions of processes that span across a wide range of length and time scales. Existing seafloor models are often based on equilibrium conditions and neglect any change due to biological effects. Additionally, typical equilibrium ripple models provide only geometric dimensions, not a roughness spectrum necessary for acoustic applications. We present a time-varying spectral seafloor model, NSEA, that predicts the spatial and temporal evolution of seafloor roughness accounting for both the evolution and degradation of ripples due to hydrodynamics and biology. Model predictions are compared to high-frequency sector-scanning sonar data collected off the coast of Panama City, FL, during the Target and Reverberation Experiment (TREX13) and at the US Army Corps of Engineers Field Research Facility in Duck, NC. The sector-scanning sonar imaged ~15 square meters of the seafloor every 12 min for up to four weeks in depths up to 20 m.

10:45

4aAO8. Predicting acoustic backscattering from bioturbated seafloors. Shawn Johnson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, shawn.johnson@psu.edu)

Seabed roughness is an important property for high-frequency scattering. However, for certain littoral environments, the surficial roughness of the seabed is often continuously being modified by hydrodynamic and biologic forces. For example, ocean surface waves may create orbital ripples on sand seafloors which are characterized by anisotropic roughness, and acoustic scattering from such a seafloor would be strongly aspect-dependent. Yet bottom-dwelling organisms rework the sediment, returning the structured seabed relief to random isotropic roughness, where acoustic scattering is more uniform with respect to the aspect ensonified. In this talk, we will utilize both analytic expressions as well as procedural generation methods to predict the two-dimensional roughness equilibrium power spectra of rippled and bioturbated seafloors, and the resulting impact on acoustic scattering prediction.

11:05

4aAO9. Sonar observations of biological activity in marine sediments. Darrell Jackson (Appl. Phys., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

This talk will begin with a review of past experiments using bottom-mounted, rotating sonars to observe temporal change in the seafloor. These measurements were conducted at 40 kHz and 300 kHz, and provide images of change as represented by ping-to-ping correlation. Time scales of change vary greatly with frequency and sediment type, with changes occurring over weeks at 40 kHz at a silty site and over minutes at 300 kHz at a sandy site. Finally, the level of scattering due to relic tubes, shells, etc., will be compared to that of live animals. The conclusion is that it is easiest to observe the cumulative effects of biological activity, but techniques are available to isolate current activity.

11:25

4aAO10. Local suppression of bioturbation and its acoustic consequences. Brian T. Hefner, Steven G. Kargl, and Kevin Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

Seafloor roughness in shallow waters can be both spatially and temporally dynamic. While wind-driven waves and bottom currents can generate bedforms such as ripples on sand sediments, bottom-feeding fish tend to limit the lifetime of these anisotropic features. This greatly reduces the time during which high frequency sonars can exploit the ripples to coherently penetrate the sediment at subcritical angles. Sonar performance prediction in these environments therefore relies heavily on sediment transport models to generate roughness predictions that accurately reflect changes on these time scales. The competition between the hydrodynamics and biology of the environment can be further complicated by the natural introduction of materials which can locally suppress the bioturbation. Two examples are discussed here: The first comes from the Sediment Acoustics Experiment in 2004 where a storm deposited a layer of mud over some areas of seafloor protecting the bedforms from fish. The second is from the recent Target and Reverberation Experiment where there is evidence that spatially-varying shell content can also reduce the effects of bioturbation. The consequences for sonar performance in these two cases are discussed. [Work supported by the U.S. Office of Naval Research and the Strategic Environmental Research and Development Program.]

Session 4aBA

Biomedical Acoustics: Ultrasound-Mediated Neuromodulation

Parag V. Chitnis, Chair
Department of Bioengineering, George Mason University, 4400 University Drive, 1G5, Fairfax, VA 22032

Chair's Introduction—8:25

Invited Papers

8:30

4aBA1. Ultrasound neuro-stimulation effects of peripheral axons *in-vitro*. Nader Saffari, Christopher J. Wright (Mech. Eng., Univ. College London, UCL, Torrington Pl., London WC1E 7JE, United Kingdom, n.saffari@ucl.ac.uk), and John Rothwell (UCL Inst. of Neurology, Univ. College London, London, United Kingdom)

Neurological disorders are a huge disease burden for individuals and the economy worldwide. For a few of these disorders, successful new treatments have been found in neuro-stimulatory techniques. Ultrasound (US) can target the brain in a spatially precise, non-invasive manner, stimulating or suppressing neuronal activity. Much success has been reported with US in small animals, primates, and human subjects. Despite all the recent progress, very little is known about the molecular and cellular mechanisms behind the observed neural responses. This study uses a controlled *in-vitro* environment, directly stimulating and recording CAPs from excised crab nerves (*Cancer pagurus*). The aim is to create an environment where both the biological and ultrasound environment can be measured and modelled accurately to gain insight into the mechanism by which mechanical forces are transduced into propagating electrical activity in nerve fibres. The results demonstrate that the constituents of unmyelinated axonal tissue are sufficient to generate de-novo action potentials in response to US stimulus. The threshold for this stimulation was much higher than similar procedures performed on CNS models but in good agreement with other PNS focused studies. They also provide the first clear evidence for the involvement of cavitation as an ultrasound stimulation mechanism.

9:00

4aBA2. Ultrasonic stimulation and metabolic stress in neuronal systems. John R. Cressman (Phys. and Astronomy, George Mason Univ., 4400 University Dr., MSN2a1, Krasnow Inst., Fairfax, VA 22030, jcressma@gmu.edu), Monica Gertz (Neuroscience, George Mason Univ., Fairfax, VA), and Parag V. Chitnis (Dept. of BioEng., George Mason Univ., Fairfax, VA)

We report on computational and experimental investigations into the effects of high-frequency focused ultrasound stimulation on neuronal tissue. We simultaneously measuring local field potentials as well as extracellular oxygen and potassium concentrations in hippocampal brain slices to investigate physiological responses, including metabolic demand, during ultrasonic stimulation. To better interpret our observations, we utilized computational models that incorporate dynamics for transmembrane ionic and osmotic flows, membrane fluctuations, and metabolism. Our experiments and models suggest that ultrasonic stimulation can produce substantial ionic redistribution that leads to a significant metabolic burden.

Contributed Paper

9:30

4aBA3. Imaging of tissue displacement induced during focused ultrasound neuromodulation *in vivo*. Stephen A. Lee, Matthew Downs, Niloufar Saharkhiz, Yang Han, and Elisa Konofagou (Biomedical Eng., Columbia Univ., 500 W 120th St., New York City, NY 10027, sal2212@columbia.edu)

Our group has recently shown feasibility of FUS modulation of the peripheral nervous system (PNS) *in vivo*. However, the interplay between FUS and the PNS is not completely understood and has never been imaged. To unveil both the mechanism as well as provide an image-guided approach to modulation monitoring, a new modulation system was designed to simultaneously image the mechanical perturbation of the tissue during modulation *in vivo*. The system consisted of a 4.5 MHz HIFU confocally aligned with a 7.8 MHz imaging transducer. Activation of the sciatic nerve of the mouse was induced with parameters previously reported for successful modulation. 200 RF frames at a 10 kHz pulse repetition frequency were used for 1D cross-correlation (20 lambda window, 90% overlap) to calculate the inter-frame axial displacement before, during, and after modulation. Displacement maps overlaid on the B-mode images illustrate that once FUS is applied, downward displacement was detected where the highest displacement is located at the focus (9.8 micron average peak displacement). After FUS modulation, displacement steadily reduced to baseline (0.5–0.8 ms). Our findings indicate that FUS neuromodulation is associated with the radiation force effect and therefore successful application is dependent upon sufficient displacements induced.

9:45–10:00 Break

Invited Papers

10:00

4aBA4. Intersections of neuromodulation, focused ultrasound, and gene delivery with brain-penetrating nanoparticles. Brian Mead (Biomedical Eng., Univ. of Virginia, Box 800759, Health System, Charlottesville, VA 22908), Namho Kim, Karina Negron (Johns Hopkins Univ., Baltimore, MD), Wilson Miller (Radiology, Univ. of Virginia, Charlottesville, VA), Panos Mastorakos, Jung Soo Suk (Johns Hopkins Univ., Baltimore, MD), Alexander Kilbanov (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), Justin Hanes (Johns Hopkins Univ., Baltimore, MD), and Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA, rprice@virginia.edu)

Focused ultrasound (FUS) can modulate CNS tissue function by facilitating MR image-guided transfection of selected brain structures with therapeutic genes. Conversely, we hypothesize that modulation of CNS tissue with FUS may be used to enhance transfection. In previous studies, our group has (i) transfected selected brain structures by delivering systemically administered, non-viral, "brain-penetrating" nanoparticles (BPN) across the blood-brain barrier (BBB) with FUS and microbubbles (MBs) and (ii) employed this platform to restore dopaminergic motor neuron structure and function in a rat Parkinson's disease model via delivery of a neurotrophic factor (GDNF) gene to striatum. Here, we show that pre-conditioning of rat neural tissue with pulsed FUS or pulsed FUS + MBs before intrastriatal convection-enhanced delivery of BPN complexed with a reporter gene (CMV-ZsGreen) serves to enhance transfection volume by 36% or 151%, respectively. Pre-conditioning with both FUS and FUS + MBs proved effective in mice as well, with both FUS protocols yielding ~75% increases in transfection. We conclude that modulation of CNS tissue plays an important role in dispersing BPN after they are delivered across the BBB using pulsed FUS and MBs. Pre-conditioning of CNS tissue with FUS may eventually represent an attractive means for enhancing transfection.

10:30

4aBA5. Central and peripheral nervous system modulation using ultrasound. Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Focused ultrasound (FUS) neuromodulation has previously been proposed as a promising technique to drive neuronal activity and has been shown throughout a breadth of applications including in mice, rats, non-human primates, and humans as a novel technique for the noninvasive manipulation of neuronal activity using ultrasound. Our group has demonstrated excitation of both the central (CNS) and peripheral nervous system (PNS). In the CNS, motor- and cognitive-related brain regions of mice were induced by targeting specific brain structures. Higher acoustic pressures increased the success rate. Pupil dilation was observed when neuromodulating regions in the brain covering the superior colliculus and other anxiety-related structures such as hippocampus and locus coeruleus. In the PNS, we showed for the first time stimulation of the sciatic nerve with FUS eliciting a physiological motor response was recorded *in vivo*. Clipping the sciatic nerve downstream of stimulation eliminated EMG activity during FUS stimulation. Peak-to-peak EMG responses and delay to signal were comparable to conventional electrical stimulation methods. Histology along with behavioral and thermal testing did not indicate damage to the nerve or surrounding regions. Our studies demonstrated the capability of FUS to modulate target specific regions in both the brain and the peripheral *in vivo*.

Contributed Paper

11:00

4aBA6. Ultrasonic neuromodulation of the mouse cortex increases cardiorespiratory activity. Christian Aurup and Elisa Konofagou (Biomedical Eng., Columbia Univ., 630 West 168th St., P&S 19-418, New York, NY 10032, ca2625@columbia.edu)

Focused ultrasound (FUS) has demonstrated its ability to modulate neuronal activity in both cortical and subcortical brain regions in a noninvasive and safe fashion in mice, non-human primates, and humans. Rodent studies have shown that ultrasonic neuromodulation (UNMOD) can elicit motor responses in limbs and trigger pupil dilation. Little is known about the effect of ultrasonic neuromodulation of the CNS on the autonomic activity in mice, primarily due to the cardiorespiratory depression caused by the anesthesia used. However, urethane has limited effects on autonomic activity and brain hemodynamics. In this paper, we demonstrate that UNMOD of the cortex increases cardiorespiratory activity in mice. A 1.9-MHz single-element focused ultrasound transducer was used to apply pulsed ultrasound to the mouse cortex. Each animal was injected intraperitoneally with urethane (1500 mg/kg), allowing for a stable plane of anesthesia and experimental efficacy window often exceeding 4 hours with minimal effects on autonomic activity. Sonications were performed in a grid spanning the cortex centered at the Bregma skull suture and heart and breathing rates were acquired using a pulse oximeter. FUS resulted in significant increases in both breathing and heart rates immediately following sonication. This study demonstrated that FUS can have an autonomous excitatory effect *in vivo*.

4a THU. AM

Invited Paper

11:15

4aBA7. High resolution modulation of human brain circuits using transcranial focused ultrasound. Wynn Legon (Neurosurgery, Univ. of Virginia, 426 Church St., SE MMC 8388, Minneapolis, MN 55406, wlegon@umn.edu)

Current non-invasive electric and electromagnetic non-invasive neuromodulatory approaches have proven effective for inducing transient plastic changes in human cortex. However, these technologies have poor spatial resolution and suffer from a depth-focality tradeoff. Transcranial focused ultrasound (tFUS) is an emerging non-surgical low-energy technique for inducing transient plasticity in sub-cortical and cortical areas with high spatial resolution and adjustable focus. tFUS has been successfully employed in small and large animal preparations and our work has also demonstrated tFUS to be effective for human cortical and subcortical neuromodulation. Here, we will discuss tFUS current approaches as well as challenges and future directions for this technology as a tool for neuroscience research and human brain mapping and potential clinical applications.

THURSDAY MORNING, 7 DECEMBER 2017 STUDIO 2, 7:55 A.M. TO 12:00 NOON

Session 4aNS

Noise and ASA Committee on Standards: Urban Planning Using Soundscape II

David Woolworth, Cochair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

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

Chair's Introduction—7:55

Invited Papers

8:00

4aNS1. Urban design with noise in mind—A model for future ASA outreach workshops for city planners and decision makers. Kerrie G. Standlee (DSA Acoust. Engineers, Inc., 15399 SW Burgundy St., Tigard, OR 97224, stanhartk@comcast.net)

In early 2009, the Acoustical Society of America funded the development of materials that could be used to introduce urban planners and decision makers to the concept of using soundscaping as a tool in land-use planning. After several months of discussion among interested persons, a symposium outline was generated and speakers were solicited to develop materials that would address specific topics considered important in training urban planners. The materials were developed and first used at a symposium co-sponsored by the ASA and the City of Portland during the ASA May, 2009 meeting in Portland, Oregon. The materials were again presented at a symposium co-sponsored by the ASA and the City of Baltimore during the ASA April, 2010 meeting in Baltimore, Maryland and finally at a symposium co-sponsored by the ASA and the City of Seattle during the ASA May, 2011 meeting in Seattle, Washington. This paper presents a summary of the materials used at the three symposiums and discusses how the material can continue to be used as an ASA outreach event at future meetings around the country.

8:20

4aNS2. An exploration of the urban design possibilities offered by soundscape theory. Gary W. Siebein, Hyun Paek, Gary Siebein, Keely Siebein, Marylin Roa, and Jennifer R. Miller (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

The basic concepts of soundscape theory including soundmarks which are the acoustical equivalent of landmarks, keynote sounds which are the typical sounds in a vicinity; and sound signals or the specific acoustic events that comprise the ambient sound were presented by Schafer (1977) and Truax (2001). These concepts offer ways to document, analyze, and design the acoustical qualities of urban places similar to the way that the concepts developed by Kevin Lynch in the Image of the City (1960) did for the visual character of cities. The urban design potential of soundscape theory lies in identifying the ecological relationships linking sources of sounds,

prospective listeners of sounds, and the "coloration" of the sounds by the urban environment that gives unique identities to localities within a city. An acoustical community is formed by the parties who are related by the need for various forms of tangible and intangible communication in an urban setting. Case studies of an entertainment district in a city, the initial planning of a new town and the redevelopment of a city park will be used to illustrate the methods and applications of this theory to enhance the aural and urban design of cities and towns.

8:40

4aNS3. Quietness as a commons: Integrating soundscape in urban planning for the environmentally just city. Antonella Radicchi (Institut für Stadt- und Regionalplanung, Technische Universität Berlin, Hardenbergstraße 40 a, Sekr. B 4, Berlin, Berlin 10623, Germany, antonella.radicchi@tu-berlin.de)

Today, urbanization and environmental pollution represent major issues of the urban planning agenda and one of the most relevant challenges is constituted by planning environmentally and socially just cities to protect human health and ensure quality of life and well-being. To achieve this goal, noise pollution and the scarcity of quietness in cities have also to be taken into account. In Europe, the importance of quiet areas has been first recognized by the 2002 Environmental Noise Directive and, since then, numerous projects have been developed by the EU Member States to define a common methodology to protect quiet areas. Notwithstanding, according to the European Environment Agency, there is still the need for in-depth research in the field. This paper argues that integrating soundscape in urban planning processes, through the paradigm of "quietness as a commons," could significantly contribute to fill this gap of knowledge. This assumption is discussed by presenting a novel citizen-driven methodology to analyze, assess and plan urban quiet areas, implemented in a pilot study in Berlin. In detail, this paper illustrates the methods applied, the findings and the first draft of planning guidelines developed with the community to protect existing quiet areas in the pilot area.

9:00

4aNS4. Basic concepts on sound sculptures design in soundscapes. Fernando J. Elizondo - Garza (Acoust. Lab., FIME-UANL, Calle 8 #422, Col. Villazul, San Nicolas, N.L. 66420, Mexico, fjelizon@hotmail.com) and Cesar Guerra -Torres (Acoust. Lab., FIME-UANL, Monterrey, N.L., Mexico)

The primitive development of visual sculptures was born for the rite and evolved toward the artistic object, that in parallel with the development of music and musical instruments. Currently with a broad sense conceptualization of the art object, the sculpture has evolved from an aesthetic object towards the concept of installation, a multisensory interactive act, within which the sound sculpture, or installation, has taken predominance. In this paper some of the basic (mental) conceptual schemes to be considered in the design of the acoustic dimension of sound sculptures will be presented, e.g., the sculpture as a sound source and its acoustic field; its location and their primary and secondary effects; acoustic or audio; the to think-design-build process in that order, among others. Some examples of the various types of objects that can be considered sound sculptures will be shown and the importance of the sound sculptures in the design of soundscapes will be discussed.

9:20

4aNS5. Soundscapes in historic urban environments—A case study. Pamela Jordan (HEAD-Genuit-Foundation, Technische Universität Berlin, Institut für Strömungsmechanik und Technische Akustik Fachgebiete der Technischen Akustik, Sekr. TA7, Einsteinufer 25, Berlin 10587, Germany, pam.f.jordan@gmail.com)

In current soundscape practice, soundscape descriptors tend to focus on present conditions, either toward an assessment of site conditions or as predictive tools for future interventions. But does the same approach work when the acoustic environment reflects the past as well, such as at historic sites? How does the complexity of a place representing both past and present inform visitors' reactions to the soundscape? Results from a recent study at the Berlin Wall Memorial will be presented, where a guided soundwalk, informed by archival accounts of the area, was combined with a survey to scrutinize visitor impressions of the soundscape in conjunction with the site's sonic past. Descriptors common to soundscape studies were used in semantic scales alongside new proposals aimed directly at understanding the role that historic information plays in assessments by visitors and local experts. Results from the study will be presented, showing how historic information can subvert some of our fundamental assumptions about soundscape, such as what constitutes "appropriate" or "pleasant" conditions, and how results may differ across expert groups. Indeed, this research demonstrates the powerful impact that historic context can exert on soundscape assessment, with implications for soundscape practice, heritage interpretation, and urban planning alike.

4a THU. AM

9:40

4aNS6. The advantage of binaural recording for soundscape assessment. Klaus Genuit (Head Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

In the beginning, binaural recording was used for music productions. For more than 40 years, applications are being intensively used in the automotive industry to optimize vehicle interior sound. Only a binaural recording system like an artificial head enables listeners using calibrated equalized headphones to perceive hearing events which are very realistic to the original sound field. Nowadays, within the scope of soundscape standardization ISO 12913, recommendations are being discussed regarding the use of binaural technology for recording environmental sounds, where in the past normally only single microphones were used. For an overall estimation of the total sound pressure level, this may be sufficient, but in a complex sound situation, like soundscapes with several different sound sources at different locations, the selectivity of the human hearing is only given properly by binaural listening. This paper will explain the technology of binaural recording and playback systems and its advantages in comparison with other sound recording systems.

10:00

4aNS7. Evaluation of urban soundscapes: Field vs. laboratory assessments using binaural recordings. Roland Sottek and André Fiebig (Head Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

It is well-acknowledged that soundscape investigations must be carried out in the original context. In order to investigate the influence of acoustic cues on the perceptual measurements, binaural recordings were performed during soundwalks in the center of Gothenburg (Sweden) using a calibrated binaural headset. The participants were 20 students from the Chalmers Technical University. The evaluations were carried out *in-situ* regarding, e.g., loudness, appropriateness, pleasantness, and eventfulness at eight different outdoor gathering spaces. The group of students was divided into four groups. Two groups went from location 1 to 8, separated in time by 15 minutes. The other two groups went from location 8 to 1. At all eight locations each group recorded the acoustic environment (overall 32 recordings). The laboratory assessments were performed with the same students the next day using four recordings measured at four different locations. First, the evaluations were performed simultaneously in the classroom using Sennheiser HD 414 headphones, and second in groups of four students using four calibrated binaural headsets in playback mode. The paper describes the experiments performed in the original and laboratory context, and discusses the results of the different approaches. Additionally, instrumental parameters such as representative values for loudness are compared to the perceptual results: a relevant aspect regarding the second part of the international standard on soundscape dealing with data collection.

10:20–10:40 Break

10:40

4aNS8. e-Appraisal of soundscape for public squares in China. Andy Chung (Smart City Maker, Smart City Maker, Copenhagen, Denmark, ac@smartcitymaker.com) and W. M. To (Macao Polytechnic Inst., Macao, Macao)

There have been activities undergoing in public squares in China changing the soundscape of the venues as well as their neighborhood. Such activities include group dancing with music, martial arts playing sound generating tools, etc., which are not the anticipated activities during the planning and design stage. Soundwalk and surveys using an app were conducted to have an appraisal of the situation. This paper presents the findings of the case studies and recommendations for future planning of public squares with unanticipated sound generating activities.

11:00

4aNS9. Going beyond noise in urban planning—Human perception will be the trusted guide. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de) and Andrè Fiebig (Head Acoust., Herzogenrath, Germany)

When it comes to urban planning "people's perceptions or experiences and/or understanding of an acoustic environment" should be the guiding feature. The use of a variety of data collection methods related to human perception, acoustic environment and context is recommended according to ISO 12913. Following this standard, it is human perception which is paramount determining and stimulating any physical measurement. The international standard ISO 12913 will allow the collection of perceptual data in a harmonized way to get access to the key components in Soundscape: people, acoustic environment, and context. This paper will introduce and discuss techniques of interviews and guidelines, exploration of areas through soundwalks, but also the collaboration platform within a soundscape approach regarding urban planning. It will present strategies for "measurement by persons," and those applied in "measurement by instruments" to overcome the significant gaps that still exists between the both.

11:20

4aNS10. Sensitivity to sounds produced in the city, sound ordinance issues, and cultural planning in New Orleans. Helene Stryckman (Universite Catholique de Louvain, Ave. du Haut-Pont 18, Brussels 1050, Belgium, str.helene@gmail.com) and David Woolworth (Roland, Woolworth & Assoc., Oxford, MS)

In an effort to address sound ordinance issues and better quantify and qualify the soundscape of New Orleans, multiple soundwalks have been held in the older areas of the city over the last few years, specifically in the densest tourist areas (Vieux Carre/French Quarter and Marigny). Sound level data and participant response from several soundwalks are considered with respect to age, sex, occupation, stakeholder status, and language of descriptive used by the respondents. This soundscape inquiry technique has been complemented by qualitative ethnographic fieldwork performed in 2015 that describes the sound ordinance issues and the communities historically involved: musicians, music venues owners, cultural advocates, newcomers, and neighborhood associations. This qualitative case study reveals a process-based dynamic that evolves around different stakeholders' sensitivity to sounds produced in the city. We propose that a sound ordinance should attempt to be neutral in regard to cultural effects and be based on the community-asset knowledge of local experts.

11:40

4aNS11. The soundwalk as a tool to improve community relations and aid in (sound) conflict resolution. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Helene Stryckman (Universite Catholique de Louvain, Brussels, Belgium)

The measurement of soundscapes (under development as ISO/DIS 12913-2) is often performed with local experts in the form of a soundwalk, and for the sake of comparable results requires standardization. One issue stemming from the standardization of the soundwalk is its applicability and usefulness to all situations/locations from a holistic point of view (social, cultural, political, etc.) and whether it considers all significant factors that affect the participants. This paper examines potential benefits of a less structured approach that permits/encourages interaction between the participants who may be at odds in regard to their views on the soundscape, and also provides observations on the impressions of repeat participants. Additionally, several observations of non-auditory or personal factors from soundwalks will be provided. The work referenced in this paper was performed in New Orleans.

Session 4aPA

Physical Acoustics: Infrasound, Atmospheric Sound Propagation, and Turbulence

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

Contributed Papers

9:00

4aPA1. Infrasound from small explosions from near to intermediate ranges. Luis De Jesus Diaz, Richard D. Costley, Sarah McComas, Christopher Simpson, James Johnson (Geotechnical and Structures Lab., U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, Luis.A.DeJesus-Diaz@usace.army.mil), and Chris Hayward (Dept. of Geophys., Southern Methodist Univ., Dallas, TX)

An experiment was performed near the U.S. Army Engineer Research and Development Center (ERDC) site near Vicksburg MS on May 2014. Explosive charges were detonated and the shock and acoustic waves were detected with pressure gauges, infrasound sensors, and seismometers stationed at various distances from the source, from 3 m to 15 km. One objective of the experiment was to compare the effectiveness of different wind filter strategies. Toward this end, several sensors were deployed near each other, approximately 8 km from the site of the explosion. These sensors used different types of wind filters, including different length of porous hoses, a bag of rocks, a foam pillow, and no filter. Signal-to-noise estimates made from signals recorded with these different sensors will be used to evaluate the effectiveness of the different strategies. A second objective was to compare the infrasound and seismic signals recorded with collocated infrasound sensors and geophones. Results from this experiment will be presented. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.

9:15

4aPA2. Effects of simplified atmospheric profiles on short range infrasound propagation. Andrew Lammers (Atmospheric Sci., Univ. of Illinois at Urbana-Champaign, 1301 W Green St., Urbana, IL 61801, alammers719@gmail.com) and Michelle E. Swearingen (US Army ERDC, Champaign, IL)

Meteorological profiles strongly influence the propagation of infrasound signals. Much of the infrasound literature has focused on long-range (>250 km) propagation; however, there is interest in short-range (<150 km) propagation from an infrastructure monitoring standpoint. Therefore, understanding the effects of meteorological profiles is crucial. This paper focuses on simplified vertical temperature and wind profiles up to 20 km altitude and their effect on an infrasonic signal emanating from an arbitrary point source. A wide-angle, finite-element PE model that correctly handles discontinuities in wavenumber is used for calculating transmission loss. A large number of simulations were performed to investigate the effect that temperature and wind profiles in different layers of the atmosphere have on surface transmission loss and to assess the sensitivity of varying these profiles. A discussion of the results of these simulations are presented as well as an overview of temperature profiles used in this study.

9:30

4aPA3. On the influence of the jet stream variations on infrasonic propagation. Maxwell B. Willis, Roger M. Waxler, and Claus Hetzer (National Ctr. for Physical Acoust., Univ. of MS, 145 Hill Dr., University, MS 38677, mwillis@go.olemiss.edu)

The jet stream is an eastward flowing wind jet seen around 10 km with its largest magnitude in the winter months, lessening in the summer. Its direction, while predominantly easterly, can vary from northeast to southeast on a day-to-day basis. It can produce ducting of infrasonic signals when its speed is of sufficient magnitude. Infrasonic signals were observed on a network of sensor arrays from a series of explosions detonated throughout the year. Each series consisted of 3 explosions, 1 series for each season of the year. Sensors were deployed in arrays of 4 sensors per array. The network consisted of 4 lines of arrays extending in the westerly, easterly, northeasterly, and southeasterly directions. These arrays were located at distances ranging from 20 km to 80 km from the source. The data from each array was analyzed to determine if infrasonic signals from the explosions were detected. A propagation model was produced for each event using a parabolic equation and an atmospheric profile corresponding to the time of each event. Correlations between the observations and the direction of the jet stream are investigated.

9:45

4aPA4. Sound propagation above a flat ground with random spatially varying properties. Didier Dragna and Philippe Blanc-Benon (Ctr. Acoustique, Laboratoire de Mécanique des Fluides et d'Acoustique, 36, Ave. Guy de Collongue, Ecully 69134, France, didier.dragna@ec-lyon.fr)

The acoustic properties of ground surfaces vary in space. Usually, these variations are not accounted for in predictions of outdoor sound propagation for two main reasons. First, they are not known as it is costly to obtain finely sampled spatial measurements of these ground properties. Second, there is no existing simple and quick-to-compute analytical solution for a ground with a spatially-varying admittance, contrary to the well-documented case of a ground with a homogeneous admittance. This imperfect knowledge of the ground characteristics leads to uncertainties about the sound pressure level. This paper aims at characterizing these uncertainties. Propagation of acoustic waves over the ground with a spatially-varying surface admittance is therefore considered. Using the diagram technique, the average Green's function is determined. An asymptotic expression in the form of a Weyl-Van der Pol formula is obtained at long range and at grazing incidence. Modifications of the reflection coefficient and of the surface wave contribution due to randomness are analyzed. Numerical simulations using a linearized Euler equations solver are then carried out. Comparison of the mean pressure obtained from ensemble-average over 200 realizations and from the analytical solution is performed. Finally, the mean intensity and intensity fluctuations are investigated.

10:00

4aPA5. Efficient modeling of long range impulse sound propagation in three dimensions. Matthias Cosnefroy, Sylvain Cheinet, Loic Ehrhardt (French-German Res. Inst. of Saint-Louis, 5 Rue du General Cassagnou, Saint-Louis 68300, France, matthias.cosnefroy@isl.eu), Daniel Juvé, and Didier Dragna (Ctr. Acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Écully, France)

Sound characteristics close to the ground strongly depend on the atmospheric and ground properties in terms of amplitude, shape, and time of arrival. Time-domain numerical modeling is able to accurately account for outdoor sound propagation and is valuable to decipher the interactions between the refractive, scattering, and ground effects. It remains computationally challenging for three-dimensional long range simulations. A high-order parallel Finite-Difference Time-Domain solver with the moving frame approach is presented, featuring accurate time-domain impedance boundary conditions and very efficient convolutional perfectly matched layers to artificially truncate the computational domain. The design of the absorbing boundaries is based on a stability analysis of the time integration scheme and is shown to optimize the absorption properties for grazing waves. The model allows for accurate propagation of impulse sounds in 3D over several hundreds of meters and up to 1000 Hz using a personal laptop with a simulation duration of a few hours. The numerical predictions are compared to experimental measurements under different weather conditions, with a special focus on the sensitivity to the mean vertical wind profile and the ground properties.

10:15–10:30 Break

10:30

4aPA6. Performance of Green's function retrieval methods for various sound source distributions: Application to outdoor acoustic propagation. Max Denis (RDRL-CIE-S, U.S. Army Res. Lab., 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com), Sandra L. Collier, John Noble, W. C. Kirkpatrick Alberts, David Ligon, Leng Sim, Christian G. Reiff, and Deryck D. James (RDRL-CIE-S, U.S. Army Res. Lab., Adelphi, MD)

In this work, Green's function retrieval methods for an outdoor acoustic propagation channel are presented. Green's function retrieval methods by multidimensional deconvolution and crosscorrelation are compared for different source distributions. Of particular interest is the accuracy of the retrieved Green's function of an arbitrary sound source with that of an impulsive sound source. Also, the signal-to-noise ratio of both methods will be investigated. To this end, outdoor acoustic experiments are conducted in a clearing and wooded area in southern Maryland. Results will be obtained for various source-receiver ranges up to 400 m.

10:45

4aPA7. Atmospheric effects on acoustic vector sensing. Sandra L. Collier, Max Denis, Latasha Solomon, David Ligon, John Noble, W. C. Kirkpatrick Alberts, Leng Sim, Christian G. Reiff, and Deryck D. James (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil)

Two-dimensional acoustic particle velocity and acoustic pressure, concurrent with atmospheric data, were collected during a series of field tests. It was found that for short propagation distances, the particle velocity field remained unsaturated (little variation in amplitude and phase) for higher frequencies than did the pressure field. Theoretical comparisons are made to examine the physical effects of the propagation environment on the particle velocity field statistics.

11:00

4aPA8. Numerical analysis on the effect of the sea surface roughness on the acoustic transmission loss in the atmosphere. Diego Turo (Mech. Eng., Catholic Univ. of America, The Catholic University of America, 620 Michigan Ave. NE, Washington, DC 20064, turo@cua.edu), Teresa J. Ryan (Engineering, East Carolina Univ., Greenville, NC), John Judge, and Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, Washington, DC)

Atmospheric sound propagation over the sea surface is affected by several factors including wind, temperature profile, and sea state. Preliminary numerical studies using a Crank-Nicolson parabolic equation solution approach have demonstrated that sea roughness introduces excess transmission loss. This excess loss induced by the sea roughness can be represented instead by a flat surface with a compensatory effective impedance. Calculation of transmission loss for the flat surface is computationally more efficient but requires the evaluation of the effective impedances for any given sea state. In this study, a 2-D finite-difference time-domain solver is used to determine the effective impedances for several sea states. The effect of sea roughness on transmission loss is then quantified by using the Crank-Nicolson parabolic equation.

11:15

4aPA9. Statistical characterization of sound propagation along near-vertical paths in a turbulent atmosphere. Vladimir E. Ostashev, D. Keith Wilson, and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu)

Most previous research on sound propagation in the atmosphere has focused on nearly horizontal propagation paths, with the sources and receivers close to the ground. Along such paths, refraction by wind and temperature gradients, and scattering of sound by small-scale turbulence, are known to be important and have been studied extensively during the past couple decades. Sound propagation from elevated sources is fundamentally different because the atmospheric motions are larger, more coherent, and driven by different types of instabilities. The corresponding research is needed to address the ability of ground-based acoustic microphone arrays to detect and track elevated sound sources, as well as the ability of elevated arrays to detect and localize ground-based sources. In this paper, near-vertical sound propagation through a turbulent atmosphere is studied using approaches and methods developed in wave propagation through random media. The statistical characteristics of sound signals for near-vertical propagation such as the log-amplitude and phase fluctuations, angle-of-arrival variance, transverse and longitudinal spatial coherence, and temporal and frequency coherence are calculated and analyzed. The results obtained are compared with those for near-horizontal sound propagation.

11:30

4aPA10. Nonlinear propagation of *N*-waves through kinematic turbulence: Statistics of peak overpressure and shock front steepness. Petr V. Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Maria M. Karzova (Phys. Faculty, Moscow State University, Moscow, Russian Federation; LMFA - UMR CNRS 5509, Univ. Lyon, Ecole Centrale de Lyon, Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Sébastien Ollivier, Didier Dragna (LMFA - UMR CNRS 5509, Univ. Lyon, Ecole Centrale de Lyon, Ecully, France), Vera Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (LMFA - UMR CNRS 5509, Univ. Lyon, Ecole Centrale de Lyon, Ecully, France)

Nonlinear propagation of short *N*-waves (wavelength of 1.4 cm) through a turbulent layer with outer scale of about 16 cm was simulated based on the 2D parabolic KZK-type equation. A modified von Kármán spectrum was used to generate random fluctuations of wind velocity associated with the presence of turbulence. Statistics of peak overpressure and shock front steepness were analyzed for linear and various degrees of nonlinear effects in *N*-wave propagation. It was shown that in case of linear propagation, the turbulence mainly led to smearing the shock fronts and resulted in low probabilities of observing steepened shocks. At higher pressure level, nonlinear effects resulted in steepening of the shock fronts and hence counteracted the effects introduced by the turbulence. Due to nonlinear enhancement of focusing gain in random foci, up to twofold increase of the cumulative

probability of observing highly peaked waveforms was shown for nonlinear wave propagation comparing to the linear case. However, it was also shown that for stronger nonlinearities, saturation of the focusing gain led to saturation of the overpressure cumulative probabilities. [Work supported by RSF-17-72-10277 and by the Labex CeLyA of Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ ANR-11-IDEX-0007).]

11:45

4aPA11. Deturbing supersonic signatures. William Doebler (The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, wfd5057@psu.edu) and Victor Sparrow (The Penn State Univ., University Park, PA)

The process of deturbing (removing effects of turbulence from) experimentally measured sonic boom pressure waveforms from aircraft would provide a baseline sonic boom signature. This baseline signature could be used to calculate perception metrics that may be part of a supersonic aircraft noise certification standard. The traditional deturbing process has been shown to remove turbulence reasonably well from experimentally measured N-wave sonic boom pressure waveforms whose tail shock generates the atmosphere's impulse response. N-wave booms such as that from Concorde were deemed unacceptable and prohibited for over land operations due to their startling nature and due to public annoyance. As a consequence, newly designed low-boom aircraft are being tailored for lower amplitude sonic booms that will not have any strong shocks. Since a tail shock is needed to generate the atmospheric impulse response for traditional deturbing, new deturbing techniques are needed. Two methods are presented using cross correlation averaging and frequency domain filtering for determining a baseline supersonic signature for computing sonic boom metrics. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

THURSDAY MORNING, 7 DECEMBER 2017 — STUDIO 4, 9:00 A.M. TO 11:30 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Psychoacoustics of Speech Perception in Noise, Localization, and Frequency Selectivity

Eric Hoover, Chair
University of South Florida, 16458 Northdale Oaks Dr., Tampa, FL 33624

Contributed Papers

9:00

4aPP1. Auditory object formation from background noise improves speech perception. Maury Lander-Portnoy (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, landerpo@usc.edu) and Jason Zevin (Psychology, Univ. of Southern California, Los Angeles, CA)

Auditory object formation allows listeners to isolate a speech stream from a complex acoustic environment. While most studies focus on foregrounded speech, here we focus on whether listeners can form auditory objects from background noise to aid in speech perception. We use an open-set word recognition task with fluctuating speech-shaped-noise maskers, created by randomly interrupting the noise in each 125 ms window with a long (100 ms) or short (25 ms) silence. On each trial, three maskers were played, with the word occurring during the third. We previously showed that participants perform significantly better when the patterns repeat, compared to when the first two maskers differed from the third. Only the ordering of long or short within preceding patterns differs. To examine whether this advantage is due to auditory object formation, we leverage previous studies showing that inserted silences interfere with object formation. We insert 433 ms silences between patterns and observe defeat of the advantage we previously observed. We therefore attribute the advantage in the repeated masker condition to formation of an auditory object from background noise. Moving forward, we will manipulate other parameters affecting object formation and observe their affect on speech perception in noise.

9:15

4aPP2. Background noise interacts with spectral context effects during speech categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Speech perception is heavily influenced by surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects (SCEs)* that bias categorization of later sounds. For example, context sounds with a low-F1 bias produce more high-F1 responses to the target vowel and vice versa. SCEs have been studied for decades but only for speech perception in quiet. Background noise is expected to interfere with these effects, but the extent of this interference is unclear. On each trial, listeners heard a context sentence and target vowel both combined with speech-shaped noise. Sentences were filtered to add a + 20-dB spectral peak at low-F1 (100–400 Hz) or high-F1 frequencies (550–850 Hz). Target vowels varied from /ɪ/ to /ɛ/. Background noise was unmodulated or sinusoidally modulated at 2/4/8 Hz at variable SNRs. In general, SCEs were larger at better SNRs and in unmodulated noise, but all SCEs were smaller than those reported for speech in quiet. Complex patterns of results were observed at different noise modulation rates, with likely ties to modulations in F1 frequency regions in the context sentence. In sum, background noise diminishes context effects in speech perception, but the extent depends on noise characteristics.

9:30

4aPP3. The effect of listener motion on localization of tones in a room.
Eric J. Macaulay, Jack C. Magann, and William M. Hartmann (Phys. & Astronomy, Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, macaula5@msu.edu)

Listeners reported the perceived azimuths of 36 sine-tone sources in a room. For some trials, listener motion was forbidden and for others unstructured head and torso motion was encouraged with the listener remaining seated. Tone frequencies were 500, 1000, 2000, and 4000 Hz. Onsets and offsets were masked, leaving a steady-state duration of 7.6 s for exploration. It was found that the benefits of motion on source localization accuracy were clearly frequency dependent. Dynamic interaural cues were obtained from probe-microphone ear-canal recordings. Interaural cues were combined with head-tracker orientation data to predict perceived azimuths according to diverse hypotheses. Responses and interaural cues for trials without motion were used to predict a dynamic inferred location within the room for runs with motion and to identify stable inferences. Analysis of front/back confusions was conducted based on the regression between changes in interaural level differences and changes in head angle sampled at 76 instants. Except for 4000 Hz, the slope of the regression was significantly different depending on whether responses were in the front or back of the room. For interaural time difference, significance was limited to 500 and 1000 Hz. [Work supported by the AFOSR.]

9:45

4aPP4. Effects of listener's whole-body rotation and sound duration on sound localization accuracy.
Akio Honda (Yamanashi-Eiwa College, 888 Yokone-machi, Kofu, Yamanashi 400-8555, Japan, honda@yamanashi-eiwa.ac.jp), Sayaka Tsunokake, Yôiti Suzuki, and Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Miyagi, Japan)

Listener's head movement is known to facilitate sound localization, which creates dynamic changes to the information input to both ears. For this study, we used a digitally controlled spinning chair to examine the effects of a listener's whole-body rotation and sound duration on sound localization accuracy. We measured their sound localization accuracy at locations from left 30 deg to right 30 deg with respect to the listener. Stimuli were 1/3-octave band noise burst (fc = 1 kHz, SPL = 65 dB) with duration of 50, 200, and 1000 ms. Each stimulus was presented from a loudspeaker in a circular array. The listener, sitting on the chair at the circle center, reported the position of the presented stimulus in chair-still (0 deg/s) and chair-rotation (10 deg/s) conditions. Results showed superior sound localization accuracy of chair-rotation condition to that of a chair-still condition. Moreover, a significant effect of sound duration was observed, but interaction of the test condition and the sound duration was not significant. Although the listener's head movement might influence the localization performance more for long stimuli than for short stimuli, results suggest that the effects of a listener's whole-body rotation were less influenced by the sound duration.

10:00

4aPP5. Diagnostic value of experimental parameters for measurement of localization ability.
Angelique A. Scharine, Morgan C. Domanico, Ashley N. Foots, and Timothy J. Mermagen (Human Res. and Eng. Directorate, US Army Res. Lab., 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005-5425, angelique.a.scharine.civ@mail.mil)

This research examines the test parameters in two methods proposed by an ANSI/ASA working group for measurement of auditory localization with tactical communications and hearing protection systems (TCAPS). The localization task consisted of responding to pink noise bursts (0.250s and 7000s). Participants responded by pointing to the perceived source with a laser pointer mounted on an instrumented chair. 12 participants completed the study—half of the trials while wearing no headset and half while wearing an active Invisio X5 headset. Method 1 used four pairs of loudspeakers, separated by 8°, 10°, or 12°, and the listener was tested in two orientations (0° and 45°). Intended for use in untreated acoustic environments, Method 1 provides only a coarse estimate of localization accuracy and proportion of front/back errors. Method 2 used 36 equally spaced loudspeakers, a sound treated environment, and a chair instrumented with a laser pointer. Performance is compared as a function of separation between loudspeaker pairs in Method 1 and test method. The analyses include measures of bias, magnitude, and whether errors are due to poor acuity or front/back confusion. The utility of the proposed test configurations to assess the effects of headgear on auditory spatial awareness will be discussed.

10:15–10:30 Break

10:30

4aPP6. Distribution of spectral modulation transfer functions in a young, normal-hearing population.
David A. Eddins, Eric Hoover, and Ann C. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu)

Common measures of auditory spectral shape perception include notch-noise masking methods to estimate auditory filter shape and width (i.e., equivalent rectangular bandwidth or ERB) and spectral modulation detection to estimate across-channel spectral processing. The notch noise masking method has been widely used and the distribution of ERBs in a young, normal hearing population is well established. The spectral modulation detection method is growing in popularity yet the distribution of spectral modulation transfer functions among young normal hearing people is unknown. Here, we report the distribution of spectral modulation detection thresholds for spectral modulation frequencies from 0.25 to 8 cycles/octave for 49 young, normal hearing listeners who inexperienced in performing psychoacoustic listening tasks. The carrier was Gaussian noise with a passband from 400 to 3200 Hz and −24 dB/octave rolloff outside the passband. The presentation level was 80 dB SPL. The results reveal the typical bandpass sensitivity to spectral modulation with a minimum near 2.0 cycles/octave. Thresholds at each frequency are tightly distributed across the population and normative values are computed including mean, median, and corresponding confidence intervals. These reference values can be very useful for future investigations involving normal controls or patient populations and when comparing among previously published datasets.

10:45

4aPP7. Detection of frequency glides in single tones and single formants: Are tones appropriate analogs of formants in psychophysical experiments?
Michelle R. Molis and Amie Roten (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland HCS, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

Psychophysical studies of the limits on detection of frequency glides often have used single tones as simple, non-speech analogs of more complex speech stimuli. However, the perceptual equivalence of a single component falling at a nominal formant-peak frequency and a multi-component spectral resonance has not been firmly established. The distinction may be especially relevant when formant frequencies change over time. Detection thresholds for upward and downward frequency change were measured for single-component tonal glides and multi-component spectral resonances as a function of center frequency (500 or 1500 Hz) and duration (30 or 120 ms). The fundamental frequency of the formant stimuli was held steady at 104.5 Hz. The tones were presented at 15 dB SL and the formants were presented at approximately 15 dB SL for a 1/3-octave frequency region around the center frequency. Results suggest that thresholds for frequency glide detection depend on the complexity of the stimuli, as well as stimulus direction, center frequency, and duration. [Work supported by NIH/NIDCD.]

11:00

4aPP8. Comparison of scoring methods for spatial release from masking for speech based on analysis of psychometric function slope.
Eric Hoover (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 16458 Northdale Oaks Dr., Tampa, FL 33624, erichoover@usf.edu), Anna Diedesch (Otolaryngol. Head and Neck Surgery, Oregon Health and Sci. Univ., Portland, OR), Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), and David A. Eddins (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

The ability to benefit from the spatial separation of talkers in a multi-talker environment is critical to speech communication. A headphone-based

test of spatial release from masking for speech (SR2) has been shown to be sensitive to differences between young, healthy listeners and listeners with various auditory deficits. Existing studies relied on a simplistic method to score SR2, comparing the number of correct responses in conditions of collocated and spatially-separated background talkers. Psychometric functions were fit to trial data compiled across multiple studies in order to evaluate if an alternative scoring method could improve diagnostic power. We hypothesized that differences in informational masking between collocated and spatially separated conditions would result in a difference in condition-specific slope estimates, thus leading to a relationship between scoring method and diagnostic power. Results showed a small but significant difference in slope between conditions, consistent with expectations, but the difference between scoring methods was not clinically significant. This information will be used to guide the development of a portable, rapid implementation of SR2 for clinical and research applications. [Funding provided by NIH R01 DC015051.]

11:15

4aPP9. Establishing the response of low frequency auditory filters. Menachem Rafaelof (National Inst. of Aerosp., M.S. 463, 100 Exploration Way, Hampton, VA 23681-2199, menachem.rafaelof@nasa.gov), Andrew Christian, Kevin P. Shepherd, Stephen A. Rizzi (NASA Langley Res. Ctr., Hampton, VA), and James Stephenson (US Army Aviation Development Directorate, Hampton, VA)

The response of auditory filters is central to frequency selectivity of sound by the human auditory system. This is true especially for realistic complex sounds that are often encountered in many applications such as modeling the audibility and annoyance of sound, voice recognition, noise cancelation, and the development of advanced hearing aid devices. The purpose of this study was to establish the response of low frequency (below 100 Hz) auditory filters. Two experiments were designed and executed; the first was to measure subjects' hearing threshold for pure tones (at 25, 31.5, 40, 50, 63, and 80 Hz), and the second was to measure the Psychophysical Tuning Curves (PTCs) at two signal frequencies (Fs= 40 and 63 Hz). Experiment 1 involved 36 subjects while experiment 2 used 20 subjects selected from experiment 1. Both experiments were based on a 3-down 1-up 3AFC adaptive staircase test procedure using either a variable level tone or variable level, narrow-band, noise masker. A summary of the results includes masked threshold data in the form of PTCs, the response of auditory filters, their distribution across subjects, and comparison with similar recently published data.

THURSDAY MORNING, 7 DECEMBER 2017 — STUDIO 7, 8:00 A.M. TO 10:50 A.M.

Session 4aSCa

Speech Communication: The Southern States: Social Factors and Language Variation I

Wendy Herd, Cochair
Mississippi State University, 2004 Lee Hall, Drawer E, Mississippi State, MS 39762

Irina A. Shport, Cochair
English, Louisiana State University, 260-G Allen Hall, Baton Rouge, LA 70803

Chair's Introduction—8:00

Invited Papers

8:05

4aSCa1. Sociophonetic trends in studies of Southern English. Erik R. Thomas (English, North Carolina State Univ., Box 8105, Raleigh, NC 27695, erthomas@ncsu.edu)

Studies of Southern English have largely kept pace with advancements in sociophonetics in other parts of the world. They have expanded away from single-point studies of vowel nuclei to cover such topics as vowel-inherent spectral change, intonation, consonantal acoustics, and neurolinguistic coding of phones, especially with regard to ethnic identification. The ethnic diversity of the South, with African Americans, Latinos, enclaves of Asian Americans, and other groups such as Cajuns, facilitates growth in these new directions. Among the current linguistic trends of which sociophoneticians are staying abreast are the expansion of Latinos, social divergence within the African American population, and rapid dialect leveling among urban white Anglos. Integrating these developments with other advancements in sociolinguistics has proved more difficult. Empirical studies have incorporated various sociological methods with regard to social networks, but researchers are only now beginning to utilize them in conjunction with current phonetic techniques. There is also a need to examine wider arrays of phonetic variables in studies.

8:30

4aSCa2. Northern vowel features within the deep south: The new orleans yat dialect. Katie Carmichael (Virginia Tech, 407 Shanks Hall, 181 Turner St. NW, Blacksburg, VA 24061, katcarm@vt.edu)

Linguists have noted the distinctively Northern sound of the White, working class New Orleans English dialect locally referred to as Yat, though the features of this dialect have not been well documented phonetically. Vowel features attested include a split short-a system, BOUGHT raising, and Canadian raising of /ɑʊ/—common to Northeastern dialects but rare within the American South. In this study, I analyze the realization of these sounds within a corpus of interviews, reading passages, and word lists completed with 57 local Yat speakers, situating Yat within broader American English trends. Results indicate that in comparison to descriptions of these features in Northeastern dialects, the Yat split short-a system features slightly different phonetic triggers, BOUGHT is less raised and diphthongal, and Canadian raising appears to occur across the diphthong and not just in the nucleus. Thus, while these features do resemble those found in Northeastern dialects, there are some local distinctions. Moreover, BOUGHT raising and the split short-a system are in decline amongst younger speakers, while Canadian raising is a relatively recent innovation within the New Orleans area. Thus, Yat will likely remain a peculiar dialect within the South, despite its changing qualities.

8:55

4aSCa3. The emergence of a new pronunciation variant in a culturally changing appalachian community. Ewa Jacewicz and Robert A. Fox (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

The dialect spoken in the small southern Appalachian community in Western North Carolina, formally a part of the Inland South, represents the most archaic features of Southern English, including monophthongization of the diphthong /ai/ to [a:]. While Southerners produce the monophthong before voiced consonants ("prize") all across the South, pre-voiceless monophthongization ("price") is more restricted and still occurs in the Inland South. However, as the local culture is gradually shifting toward new mainstream sociocultural norms, so does the local dialect. To document the sound change in progress, this acoustic study examined the /ai/-monophthongization across several generations of local speakers ranging from 8 to 91 year olds, 58 males and 60 females, using a style shifting paradigm. Older generations produced the monophthong irrespective of speaking style and consonant context. Young adults introduced the diphthongal variant in pre-voiceless context. The actual departure from the monophthong occurred in children—more so in the girls—who created an intermediate, slightly diphthongized variant of /ai/, halfway between [a:] and /ai/. Bridging the two worlds, the children adjusted their pronunciation of this intermediate diphthong according to speaking style, also rejecting the archaic pre-voiceless monophthongization. Strategies for children's association of the spectral enhancement/reduction with communication context will be discussed.

9:20–9:35 Break

9:35

4aSCa4. The influence of regional identity on appalachian intonation. Paul E. Reed (Dept. of Communicative Disord., Univ. of Alabama, 909 Welsh Humanities Bldg., University of South Carolina, Columbia, SC 29208, reedpe@email.sc.edu)

In a study of intonation in Appalachian English (AE), Greene (2006) suggested that relative frequency and phonetic realization of pitch accents might reflect a stronger regional identity. The present study tests these observations, using rootedness—defined as one's orientation to place—as a means to describe observed intonational variation. Data were drawn from sociolinguistic interviews with 25 AE speakers from northeast Tennessee. In addition to the interview, every participant completed a Rootedness Metric, a psychometric survey designed to quantify place-based orientation. To consider the extent how AE speakers compared to the broader South, this cohort was also compared to a non-Appalachian Southern cohort. A section of speech from each speaker was labeled following MAE-ToBI conventions (Beckman *et al.*, 2005). Pitch accent frequencies were totaled. Additionally, tonal peak alignment for the L+H* pitch accents was measured. Results indicate that AE speakers have a greater occurrence of L+H* pitch accents ($p<0.001$), and that the tonal peak occurs earlier in the syllable ($p<0.005$) compared to speakers of other Southern varieties. Within the AE cohort, both older and more rooted speakers have more frequent L+H* occurrence ($p<0.001$), while males ($p<0.01$), and more rooted speakers ($p<0.004$) have earlier peak alignment.

10:00

4aSCa5. An acoustic and phonological description of /z/-devoicing in Southern American English. Abby Walker, Amy Southall, and Rachel Hargrave (Virginia Tech, 409 Shanks Hall, 181 Turner St. NW, Blacksburg, VA 24060, ajwalker@vt.edu)

In this paper, we investigate whether z-devoicing is a feature of Southern American English. 37 students from around Virginia, half of whom identified as Southern, half of whom did not, were recorded completing a picture naming task and a reading task designed to elicit /z/-final tokens. They also answered questions about their orientation and attitudes toward their hometown, Appalachia, and the South in general. Participants' final /z/ tokens were automatically categorized as being [z] or [s] using the FAVE Aligner (Rosenfelder *et al.* 2011) and then acoustically analyzed in terms of duration, spectral energy measures, and voicing. The results of the categorical analysis and voicing analysis suggest that people who identify as Southern /z/-devoice at higher rates than those who do not, but only in pre-pausal environments. In fact, there appears to be significantly more voicing in Southern speech in pre-voiced environments. Additionally, within the Southern identifying participants, those who express more positive attitudes towards the South /z/-devoice more than those who do not. The results suggest that pre-pausal /z/-devoicing can be considered a socially meaningful feature of Southern American English.

10:25

4aSCa6. Prosodic and segmental cues to regional dialect variation in American English. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX) and Cynthia G. Clopper (Linguist, Ohio State Univ., 1961 Tuttle Park Pl., 108A Ohio Stadium East, Columbus, OH 43210, clopper.1@osu.edu)

Historically, most research on regional variation in American English (AE) examined lexical and segmental sources of variability. Focusing on prosodic variation, we have demonstrated significant effects of regional dialect on overall articulation rate, distributions of pauses, pitch accents, phrasal-boundary tone combinations, and syllable-to-syllable vowel and consonant duration variability (Clopper & Smiljanic, 2015, 2011). In this talk, we examine whether native AE listeners utilize prosodic and segmental cues to classify AE talkers by region and whether listener familiarity with different dialects affects classification. Using free classification with unmodified, monotone (flattened F0), and low-pass filtered (removed segmental information) stimuli, listeners from Ohio and Texas grouped 60 talkers based on perceived regional similarities. Listeners were more accurate in classifying talkers with the unmodified and monotone stimuli than with the low-pass filtered stimuli. There were no differences between the two listener groups in accuracy. Multidimensional scaling analyses indicated that listeners used gender and region as salient dimensions in classifying talkers in the unmodified and monotone conditions, but not in the low-pass filtered condition. The results suggest that segmental cues facilitated classification whereas intonation and global temporal cues on their own did not. The relationship between acoustic-phonetic features and classification patterns will be discussed.

THURSDAY MORNING, 7 DECEMBER 2017 STUDIOS FOYER, 11:00 A.M. TO 12:00 NOON

Session 4aSCb

Speech Communication: The Southern States: Social Factors and Language Variation II (Poster Session)

Wendy Herd, Cochair
Mississippi State University, 2004 Lee Hall, Drawer E, Mississippi State, MS 39762

Irina A. Shport, Cochair
English, Louisiana State University, 260-G Allen Hall, Baton Rouge, LA 70803

All posters will be on display and all authors will be at their posters from 11:00 a.m. to 12:00 noon.

Contributed Papers

4aSCb1. Incipient /ay/-Raising in Baton Rouge. Kelly Berkson (Linguistics, Indiana Univ., 1020 E. Kirkwood Ave., Dept. of Linguist - Ballantine 844, Bloomington, IN 47405, kberkson@indiana.edu) and Wendy Herd (MS State Univ., MS State, MS)

Canadian raising targeting the /ay/ diphthong has been reported for a number of dialects of U.S. English. In the United States and elsewhere, however, the incipient—or purely phonetic—stage of diphthong raising, wherein it is triggered only by consonants that are phonetically voiceless, has been notoriously difficult to capture. In such a stage, raising is expected before the /t/ in *cite* (/saɪt/ → [sʌɪt]) but not before the flapped-/t/ in *citing* (/saɪtɪŋ/ → [saɪɾɪŋ]). Berkson, Davis, and Strickler (in press) recently discovered incipient phonetic raising in northeastern Indiana, however, and have suggested that extremely short diphthongs immediately preceding a primary stress (as in *citátion, psychólogy*) may be the very first to undergo raising. We investigate this hypothesis with data from Louisiana. While Yat English spoken near New Orleans has been reported as an /aw/-raising dialect (Carmichael, 2012), we also find /ay/-raising in New Orleans. Furthermore, the question of whether raising is present in Baton Rouge remains unanswered. Data presented here reveal that incipient phonetic raising is present in some—but not all—talkers from Baton Rouge. We consider the variation in Baton Rouge /ay/-raising in the context of gender differences and of /ay/-monophthongization typical in Southern varieties of English.

4aSCb2. Perception of phonetic imitation of Southern American English by Midwesterners. Cynthia G. Clopper and Ellen Dossey (Ohio State Univ., 1961 Tuttle Park Pl., 108A Ohio Stadium East, Columbus, OH 43210, clopper.1@osu.edu)

Phonetic imitation has been observed across regional dialects in English, French, and Mandarin. This cross-dialect imitation is affected by social factors, including dialect prestige and social salience of the linguistic variables. The goal of the current study was to explore the perception of phonetic imitation of Southern American English by native speakers of Midwestern American English in a word shadowing task. The magnitude of the Midwesterners' phonetic imitation was assessed using an AXB perceptual discrimination task, in which a different group of Midwestern listeners was asked to identify either a baseline read token or a shadowed token as more similar to the model token. The results of the AXB task revealed evidence of overall imitation, consistent with acoustic measures of imitation in word durations, vowel formant frequencies, and vowel formant trajectories. Moreover, perceived imitation was weaker for shadowers who were told where the model talker was from (i.e., Louisville, Kentucky) than for shadowers who were not provided with any information about the model talker, suggesting that explicit information about the model talker's dialect background can shape the magnitude of cross-dialect phonetic imitation. The specificity of this cross-dialect phonetic imitation across acoustic domains and vowel variables will be discussed.

4aSCb3. Acoustic correlates of perceived southernness ratings. Kaylynn Gunter, Charlotte Vaughn, and Tyler Kendall (Linguist, Univ. of Oregon, 1585 E. 13th Ave., Eugene, OR 97403, kgunter@uoregon.edu)

The Southern U.S. dialect and the Southern Vowel Shift (SVS), in particular, have been the subject of extensive research (e.g., Feagin, 1986; Labov, 1991; Fridland & Kendall, 2015), though there is limited work examining what acoustic cues trigger listeners to judge a speaker as sounding southern (cf. Fridland, Bartlett, & Kreuz, 2004; Allbritten, 2011). Fridland & Kendall (2012), and others, have used the Euclidean distance (ED) between the front vowels /e/ and /ɛ/ as a gradient metric of speakers' degree of SVS shiftedness. While this was demonstrated to be a useful diagnostic in production, it has not been tested against listeners' percepts of speakers' southerness. This study asks: are listeners' perceptions of southernness predicted by a speaker's /e/-/ɛ/ ED, or other such measures? To test this question, we presented listeners with isolated words from both southern and western speakers. Listeners rated words on a 1–9 scale of how southern they sound. We assess whether southernness ratings are predicted by speakers' /e/-/ɛ/ ED and other acoustic correlates. We examine both the perception of vowels implicated in the SVS, in addition to those less associated with the shift. Results shed light on the acoustic correlates of perceived southernness in vowel production.

4aSCb4. Southern stops: Phonation type differences in Mississippi. Wendy Herd (Mississippi State Univ., 2004 Lee Hall, Drawer E, MS 39762, wherd@english.msstate.edu)

While the voicing contrast between American English word-initial stops is often described as relatively uniform across speakers (e.g., voiced segments are produced with short positive VOTs while voiceless segments are produced with long positive VOTs), considerable sociophonetic variation exists. The current study investigates variation in the VOT of voiced and voiceless word-initial stops in *pot, bot, tot, dot, cot, got* produced in isolation and in carrier sentences. Participants included 40 native English speakers from Mississippi grouped according to their self-identified gender and ethnicity. As previously reported, African American speakers produced significantly more voiced stops with negative VOTs and more fully voiced closures preceding voiced stops than Caucasian American speakers. While speakers did not differ in their production of voiceless VOT when words were read in isolation, African American speakers maintained closure voicing preceding voiceless stops far more often than Caucasian American speakers. No gender differences were found. These data suggest this voicing variation is due to robust dialectal differences. The possibility that speakers who exhibit more prevoicing and more fully voiced closures are prolonging closure voicing through laryngeal lowering or nasal venting is explored through the examination of pitch and intensity trajectories within the closure and within the following vowel.

4aSCb5. Pitch accents in read speech: Black and White southern women. Yolanda F. Holt and Balaji Rangarathnam (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu)

Prosodic variation between African American English and General American English has been attested to in numerous works, yet few studies have collected measures of F_0 in African American English and fewer have examined F_0 beyond the word level. Additionally, the analysis of prosodic variation in regional dialects of American English is not well studied. F_0 movement at the level of the Intonational Phrase (IP) is known to convey both local and global information. Research on F_0 movement in General American English has analyzed combinations of H(igh) and (L)ow pitch accents as categorical markers of prosodic alignment to the segmental string. Understanding the alignment of F_0 contours provides key information on phonetic realization and phonological alignment in the creation of intonational categories. This pilot data explores the interaction of F_0, vowel duration and word duration of prenuclear and nuclear pitch accents in the read speech of Black and White southern women. This study seeks to determine if group differences exists in the expression of pitch accents between the regionally defined socio-ethnic dialects used by the two groups. Results will be discussed in terms of dialect variation.

4aSCb6. Comparing lexical decision reaction times and error rates for Southern and British English. Mairym Llorens (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu)

The English spoken in the southern states of the US as well as the "received pronunciation" of British English are two varieties that the majority of students at the University of Southern California are relatively unfamiliar with. Some of the popular stereotypes surrounding the two groups of speakers, however, diverge: while it has been shown that certain listeners consistently down-grade Southern speakers on positive attributes like intelligence and competence, no such effect is observed for British speakers. This study aimed to investigate the impact of negative stereotypes on lexical decision reaction times and error rates. A list containing high frequency words, low frequency words and non-words was constructed and a trained voice actor recorded the stimuli in both British and Southern English. Participants heard half of stimuli in each variety. Participants tended to incorrectly reject more real words in Southern English than British. These errors occurred in trials with relatively short reaction times. British English stimuli tended to have slower reaction times. The implications of these findings in light of other experiments comparing lexical decision in these and other varieties of English are discussed.

4aSCb7. Acoustically quantifying /ai/ monophthongization in four southern dialect regions. Rachel M. Olsen, Michael Olsen, and Margaret E. Renwick (Linguist, Univ. of Georgia, 142 Gilbert Hall, University of Georgia, Athens, GA 30602-6205, rmm75992@uga.edu)

The *Atlas of North American English* describes four southern U.S. dialect areas: Inland South (IS; interior Appalachia), Texas South (TS; around Dallas), Florida (FL), and South (S; remainder of Southern US). These areas are distinguished by degree of /ai/ monophthongization, the Southern Vowel Shift's (SVS) triggering feature (Labov *et al*., 2006). IS is argued to be most advanced in the SVS, and /ai/ weakens in all phonetic environments. While not as advanced generally, TS also features all-environment weakening. In S, /ai/ weakens only in syllable-final and pre-voiced conditions, and in FL it remains diphthongal. This study uses the *Digital Archive of Southern Speech (DASS)* to acoustically measure /ai/ weakening in each region. Speech was force-aligned and F1 and F2 values collected at five time points. Related work on F2 formant angle in *DASS* speech has suggested /ai/ weakens most in IS (Renwick and Stanley, 2017). Here, trajectory length (TL) measurement (Fox and Jacewicz, 2009) corroborates evidence for regional effects on diphthong dynamics, where longer TL corresponds to more diphthongal production. Mixed modeling suggests that IS and TS speakers monophthongize /ai/ more than those in S and FL. The effects of phonetic environment, race, and gender are also explored.

4aSCb8. Dynamic trajectories of tense vs. lax vowels in the American South. Margaret E. Renwick (Dept. of Linguist, Univ. of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu)

Regional variation in American English speech is often described in terms of vowel shifts, in which one or more sounds "move" within the X,Y plane of height and backness captured by vowels' first and second formant values. Such shifts indicate which sounds are converging or diverging, providing predictions for language change and variation. Static characterizations of shifting, with a single pair of F1/F2 values taken near the vowels' midpoint or intensity peak, provide an approximation of vowels' relations, but fail to capture acoustic information during the rest of the vowels' time course. In Southern dialects, where front tense and lax vowels may "swap places," static measures show strong overlap between these pairs and thus predict mergers between, e.g., *beet-bit*, *bait-bet*; instead, these mergers' clear lack of occurrence indicates the inadequacy of static methods. Using data from 64 semi-spontaneous linguistic interviews in the Digital Archive of Southern Speech, we model the shape and movement of vowels' dynamic trajectories. Formant values taken at five locations show that tense and lax vowels have distinct beginning and endpoints, though they pass through shared acoustic space near their temporal midpoints. Sociolinguistic factors affecting trajectories' length and shape are investigated using Generalized Additive Mixed Models.

4aSCb9. A preliminary examination of the gender role in back-vowel fronting in Central Louisiana. Irina A. Shport (English, Louisiana State Univ., 260-G Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu)

In addition to the Southern Vowel Shift that involves /ɛ ɪ/ fronting, high back vowels also tend to be fronted in white Southern U.S. speech (Labov, Ash, & Boberg, 2006; Thomas, 2001). The tense /u/ is the first to shift in the back-vowel system, followed by the lax /ʊ/. Gender effect on the degree of fronting has been reported for one vowel but not the other: Male speakers lead in /u/fronting, whereas /ʊ/ fronting occurs more or less uniformly across speakers of different genders (Clopper, Pisoni, & de Jong; Fridland, 2001). This study provides a preliminary examination of the role of gender in the relative degree of back vowel fronting in young adult speakers from Central Louisiana. They were recorded producing words with target vowels in a variety of tasks: word list and passage reading, sentence creation, and informal conversation. The formant values were analyzed with a reference to other vowels in each speaker and with a reference to average values for Southern U.S. English reported in previous research. The data are discussed in the context of variation in the South, adding Central Louisiana to the linguistic map.

4aSCb10. Flapping before a stressed vowel: The case of *whatever*. Irina A. Shport, Gregory Johnson (English, Louisiana State Univ., 260-G Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu), and Wendy Herd (English, State Univ., MS)

In English, word-internal intervocalic alveolar stops are predominantly flapped when preceding an unstressed vowel (*water, charity*) and optionally flapped at word boundaries preceding unstressed and stressed vowels (*that is, private airplane*). In this study, we show that /t/ is flapped in *whatever* although it is word-internal and precedes a stressed vowel. The data were elicited in a sentence reading task, with four speakers of Appalachian English. The duration of /t/ and the acoustic correlates of stress were examined. A comparison of vowel duration and amplitude patterns in *whatever* versus *everwhat* (both words are relative pronouns in free relative clauses in this dialect, $N = 699$) showed that the second syllable is stressed in *whatever*. A comparison of /t/ durations showed no significant differences among *whatever*, *watermelons, waterlilies, water buffalo* ($N = 533$, $M = 30.1$ ms). These results may be interpreted as: (a) *whatever* is an exception to the word-internal flapping environment, or (b) the word-internal flapping environment must be modified to include preceding stressed vowels at morpheme boundaries, or (c) *whatever* consists of two phonological words and falls within the word-final flapping environment. Prosodic and syntactic analyses of free relative clauses consistent with the last interpretation are discussed.

4aSCb11. Social and linguistic influences on the phonetic imitation of Southern American English vowels. Ellen Dossey (The Ohio State Univ., 108A Ohio Stadium East, 1961 Tuttle Park Pl., Columbus, OH 43210, dossey.1@osu.edu)

This study investigated the effect of regional biases on phonetic imitation, specifically exploring imitation of Southern American English by Midwesterners in a lexical shadowing task. To manipulate the regional biases at play, participants received different information about the model talker's regional origins. They were either told she was from the Midwestern city of Columbus, Ohio; the Southern locations of Eastern Kentucky or Savannah, Georgia; or nothing at all. After shadowing, participants completed a survey about their perceptions of the talker and her supposed region of origin. Vowel formant frequencies and trajectories were analyzed for imitation. Significant imitation was observed, although not all vowels were imitated equally. The survey revealed that the Southern locations differed in the biases associated with them, and that the Midwest was more socially desirable than the South. However, the survey responses were not predictive of imitative behavior. The vowel selectivity in imitation may be due to linguistic factors, such as the baseline distance between the model talker's and participants' vowels, or social factors such as the relative level of social stigma associated with the vowel variants. The results of this study suggest that imitation may be partly automatic but also influenced by certain language-external social information.

4aSCb12. Vowel acoustic characteristics of Southern White English produced by speakers from New Orleans area, Louisiana. Hyunju Chung and Lauren A. de Mahy (Dept. of Commun. Sci. and Disord., Louisiana State Univ., 81 Hatcher Hall, Field House Dr., Baton Rouge, LA 70803, hchung@lsu.edu)

This study aims to characterize vowels produced by speakers from the New Orleans area, Louisiana. The state of Louisiana is well known for its various dialects spoken across the regions. Despite the possible methodological issues and challenges researchers face for studying speech and language of young children (e.g., Oetting, & McDonald, 2002), vowel characteristics of different dialects, especially those of the New Orleans area, have not been well investigated. In the current study, vowels produced by speakers of Southern White English of New Orleans will be compared to those of two other regions of Louisiana. Participants included a total of 30 female adults, ranging in age from 18 to 25. There were 10 participants in each of the New Orleans, other Southern, and Northern Louisiana area. Each participant was asked to produce English words containing 11 different monophthongs three times in a randomized order. Vowel duration, the first three formant frequencies at the vowel midpoint, and formant trajectories were measured and analyzed. Acoustic differences in vowels are expected among speakers of three different regions. The outcome of the current study will provide normative data necessary for evaluating children's vowel articulation skill of diverse dialectal backgrounds.

Session 4aSP

Signal Processing in Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Signal Processing in Acoustics Metamaterials

Jeffrey S. Rogers, Cochair
Acoustics Division, Naval Research Lab., 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375

Matthew Guild, Cochair
University of Texas at Austin, Austin, TX 78712

Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. Computational imaging with a frequency-diverse metasurface aperture. David Smith (Duke Univ., Box 90291, Durham, NC 27708, drsmith@ee.duke.edu)

We demonstrate image formation with radio frequency (RF) waves using a frequency-diverse metasurface antenna. The metasurface antenna consists of a planar cavity—formed from a double-layer, copper-clad printed circuit board—fed from one side by an RF source, and with a series of irises patterned into the opposite side. At any particular frequency, the cavity mode excited by the feed in turn excites the irises, each of which radiates as a magnetic dipole element. The cavity walls (fabricated using vias patterned into the circuit board) are irregular, so that the excited cavity modes exhibit considerable variation as a function of frequency. Thus, the radiated fields from the cavity-backed metasurface tend to have a series of randomly directed lobes, the number and directions of which vary significantly as a function of frequency. These radiation patterns can be used to form measurements of a scene, with the scene then estimated using computational imaging techniques. We have applied the metasurface aperture in a prototype imaging system capable of acquiring high resolution images of human-scale targets. The metasurface aperture design and reconstruction methods are general, and can be easily adapted for acoustic imaging.

8:25

4aSP2. Non-reciprocal acoustic metamaterials for full-duplex communications and imaging. Andrea Alu, Curtis Wiederhold, Li Quan, and Dimitrios Sounas (The Univ. of Texas at Austin, 1616 Guadalupe St. UTA 7.215, Austin, TX 78701, alu@mail.utexas.edu)

In this talk, we discuss our recent efforts in the area of non-reciprocal metamaterials based on moving media or spatiotemporally modulated acoustic elements, aimed at realizing efficient and compact isolation and circulation in acoustic devices. We discuss the relevance of these concepts for new-generation acoustic devices that realize isolation in guided and radiating structures, with relevance for acoustic imaging applications, including ultrasound and sonar technology, and for sound communications. Non-reciprocal responses and built-in isolation and circulation in waveguides and in radiating elements offers the opportunity of translating the advantages recently enabled by non-reciprocal components in electromagnetics, based on full-duplex signal transmission, to acoustic technology.

8:45

4aSP3. High speed acoustic communication with orbital angular momentum multiplexing. Chengzhi Shi, Marc Dubois, Yuan Wang, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, CA 94720, chengzhi.shi@berkeley.edu)

Acoustic communication is critical for underwater application such as deep ocean scientific explorations, off-shore industrial controls, and ocean environment monitoring. This is because other techniques using electromagnetic waves such as RF communications are difficult for underwater applications due to the strong absorption of water in such a frequency. Optical communication, on another hand, suffers from the light scattering from micro-particles or marine life making long range underwater optical communication very challenging. Therefore, using acoustic waves to transmit information is currently the dominate technique for underwater applications including data collection and remote control of off-shore benthic stations. However, the low frequency bandwidth available for acoustic communication limits the data transmission rate and information capacity or content. We propose and experimentally demonstrate here a new approach using the orbital angular momentum (OAM) of acoustic vortex beams which provides a new and independent channel that enhances the data transmission rate by eight fold. The OAM multiplexing method demonstrated here will impact significantly on future underwater communications.

9:05

4aSP4. Metamaterial-based passive phased arrays: Resolution, losses, and characterization. Likun Zhang, Joel Mobley (Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38677, zhang@olemiss.edu), Xue Jiang (Nanjing Univ., Nanjing, China), and Yong Li (Tongji Univ., Shanghai, China)

Metamaterial and metasurface-based passive phased arrays provide novel means for the manipulation of acoustic fields. As passive elements, metasurfaces can provide local phase delays which enable the steering of sound fields and facilitate extraordinary wave phenomena. Some aspects that affect the performance of these passive elements include the spatial resolution/aliasing effect, the component resonances that shape the transmission spectra, and the thermoviscous losses that limit the transmission efficiency [X. Jiang, L. Yong, and L. Zhang, *J. Acoust. Soc. Am.* **141**(4), EL363–368, April 2017]. Efficient acoustic field characterization is essential for investigating these issues with the ultimate aim of optimizing the geometrical parameters of the structure. This talk will address some details of the physics of metamaterial-based passive arrays and the broadband approach for assessing their transmission properties, with the aim of improving the design cycle and exploring novel uses.

9:25

4aSP5. A diffraction based acoustic single pixel imager. Jeffrey S. Rogers, Charles Rohde, David Smith (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil), Christina J. Naify (Jet Propulsion Lab., Pasadena, CA), and Gregory Orris (Acoust. Div., Naval Res. Lab, Washington, DC)

We present a method for localizing an acoustic source with a single, omni-directional receiver paired with shaped aperture screens that allow for a spatially diverse set of measurements. Traditionally this is accomplished by patterning the screen with a series of subwavelength openings that allow the acoustic transmission an otherwise sound opaque material. Here we consider screens that have openings on the order of an acoustic wavelength or larger and incorporate a diffraction model into the single pixel imaging framework to account for these larger openings. The method is demonstrated on experimental data taken in air and an analysis of the error as a function of receiver position is presented. [This work was supported by ONR.]

9:45–10:00 Break

10:00

4aSP6. Tailoring the flow of acoustic waves by architectured metamaterials. Nicholas X. Fang (MechE, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, nicfang@mit.edu)

Today, sound is an indispensable component in numerous industrial and consumer products, such as musical instruments, cars, building technology, medical diagnostics, and many others. Acoustic characteristics are among their most important properties, greatly influencing their function and our society at large. Recent development of acoustic metamaterials opens a door to an unprecedented large design space for acoustic properties such as negative bulk modulus, negative density, and refractive index. These novel concept expands paves the way for the design of a new class of acoustic materials and devices with great promise for diverse applications, such as broadband noise insulation, sub-wavelength imaging and acoustic cloak from sonar detection. In this invited talk, I will present our development of advanced design and micro/nanofabrication techniques, to enable exploration architectured meta structures for acoustic waves. These structures show promise on focusing and rerouting ultrasound through broadband metamaterials. As example, our study on the sound absorption of thin composite aerogel foams using a bimodal porous structure predicts a possible route to perfect thin film absorber by increasing the amount of epoxy resin. In a second case, bifunctional acoustic lens can be implemented in practice with subwavelength unit cells exhibiting effective anisotropic parameters. Lastly I will report our study on a prototype hydraulic hydrogel actuators with excellent optical and sonic transparency.

10:20

4aSP7. Acoustic characterization of silica aerogel clamped plates for perfect absorption purpose. Alan Geslain (ISAT, Dr. EA1859, Universite Bourgogne Franche Comte, Nevers, France), Vicente Romero-Garcìa, Jean-Philippe Groby (LAUM UMR CNRS 6613, Universite du Maine, Le Mans cedex 9, France), Francisco Cervera, and Jose Sanchez-Dehesa (Electron. Dept., Universitat Politecnica de Valencia, Camino de vera s.n., Valencia 46022, Spain, jsdehesa@upv.es)

Silica aerogel has been widely studied as bulk material for its extremely low density and thermal conductivity. Plates or membranes made of this extremely soft materials exhibits interesting properties for sound absorption. A novel signal processing method for the characterization of an acoustic metamaterial made of silica aerogel clamped plates is presented. The acoustic impedance of a silica aerogel clamped plate is derived from the elastic theory for the flexural waves, while the transfer matrix method is used to model reflection and transmission coefficients of a single plate. Experimental results are obtained by using an acoustic impedance tube. The difference between the measured and modeled reflexion and transmission coefficients is minimized under constraints to recover the acoustic parameters of the silica aerogel plate. Once the properties of the silica aerogel plate are obtained, the perfect absorption condition is derived by studying the reflection coefficient of a aerogel plate rigidly backed with a cavity in the complex frequency domain. Reflection measurements with a varying cavity length from 1 mm to 65 mm are performed to valid the perfect absorption condition. It is found that the use of silica aerogel plate exhibit perfect absorption condition for several configurations.

10:40

4aSP8. Constant amplitude sound waves in non-Hermitian metamaterials. Etienne Rivet (Inst. of Elec. Eng., EPFL, Lausanne, Vaud, Switzerland), Andre Brandstötter (Inst. for Theor. Phys., TU Vienna, Vienna, Austria), Konstantinos Makris (Phys. Dept., Univ. of Crete, Heraklion, Greece), Stefan Rotter (Inst. for Theor. Phys., TU Vienna, Vienna, Austria), Herve Lissek (Inst. of Elec. Eng., EPFL, Lausanne, Switzerland), and Romain Fleury (Inst. of Elec. Eng., EPFL, 1 University Station C0803, Austin, TX 78712, romain.fleury@epfl.ch)

We investigate the possibility for acoustic waves to propagate with a constant amplitude in disordered media. We find that this remarkable property is possible if one adds a tailored distribution of gain and loss on top of the disorder, making the medium non-Hermitian. We present the theory of constant-amplitude acoustic waves in both cases of continuous and discrete media, and provide an experimental demonstration in a metamaterial at audible frequencies.

11:00

4aSP9. Breaking the fundamental limit of space-coiling metamaterial: Toward simultaneous phase and amplitude modulation. Reza Ghaffarivardavagh, Jacob Nikolajczyk (Mech. Eng., Boston Univ., Boston, MA), Stephan Anderson (Radiology, Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., One Silber Way, Boston, MA 02215, xinz@bu.edu)

Acoustic metasurfaces represent a family of planar, wavefront-shaping devices garnering increasing attention due to their capacity for novel acoustic wave manipulation. Despite the successful demonstration of phase engineering using metasurfaces, amplitude modulation remains overlooked. This work explores the feasibility of simultaneous phase and amplitude modulation using space-coiling metamaterials. In the case of conventional space-coiling metamaterials, we observed a fundamental bound on the transmission coefficient, which precludes full wavefront manipulation. Herein, we present a novel class of metasurfaces featuring a modified space-coiling structure and enabling full acoustic control with simultaneous phase and amplitude modulation. The functionality of this class of metasurfaces, featuring a gradient in channel spacing, has been theoretically and numerically investigated and an equivalent model simplifying the structural behavior is presented. Furthermore, a metasurface featuring this novel geometry has been designed and its functionality in modifying acoustic radiation patterns simulated. The class of acoustic metasurface demonstrated in this work provides a new design methodology enabling complete acoustic wave manipulation, which may find utility in a range of applications including biomedical imaging, acoustic communication and non-destructive testing.

11:20

4aSP10. Deconvolution methods to obtain the impulse response of acoustic metamaterial samples of finite extent. Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, sprattkyle@gmail.com), Colby W. Cushing, Kevin M. Lee, Preston S. Wilson, Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Xiaoshi Su, and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

One complication of characterizing the response of acoustic metamaterials is that models often assume the medium is of infinite extent. On the other hand, material property measurement can only be done with samples of finite size. The result is that acoustic field measurements include the effects of edge diffraction and scattering from fixtures. This work presents deconvolution methods to extract the frequency-dependent reflection and transmission behavior of acoustic metamaterial samples. Measurements using broadband chirp signals are obtained and subsequently post-processed using deconvolution techniques to obtain high-time-resolution impulse responses. Additionally, the chirp signals can be modified to compensate for the frequency response of the transducers being used. Examples will be shown for the transmission behavior through a two-dimensional pentamode gradient index lens and the reflection response of a flat square plate. [Work supported by ONR.]

11:40

4aSP11. The use of acoustic resonators for characterization of underwater acoustic metamaterials. Preston S. Wilson and Michael R. Haberman (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

The characterization of acoustic metamaterials used in underwater applications at low frequencies (below a few kilohertz) is complicated by the associated long wavelengths (1 m and greater), which render free field methods impractical for a range of frequencies of significant interest. One approach used to address this issue is the water-filled acoustic resonator. The key benefit is that a resonator tube with a length of as little as one half wavelength can be used for material characterization, compared to free-field techniques, which require a spatial extent of several wavelengths in two or more dimensions. Another benefit is that sample size can be orders of magnitude smaller when using a resonator. These benefits are enabling, but come at the expense of designing a system to overcome the fact that practical tube materials such as steel or glass cannot be considered rigid when water-filled. Another caveat is that an appropriate model is required to relate the observed resonance frequencies to the acoustic properties of interest. An overview of resonator-related measurement techniques will be presented, along with recent efforts to apply these techniques to the measurement of underwater acoustic metamaterial properties. [Work supported by ONR.]

Session 4aUW

Underwater Acoustics: Underwater Soundscapes and Noise: Measurement and Abatement

Kathleen E. Wage, Cochair
George Mason University, 4400 University Drive, MSN 1G5, Fairfax, VA 22030

Aleksander Klauson, Cochair
Civil Engineering and Architecture, Tallinn University of Technology, Ehitajate tee 5, Tallinn 19086, Estonia

Contributed Papers

8:00

4aUW1. Low frequency ambient noise modeling in the North Pacific: Simulation versus experiment. Mehdi Farrokhrooz and Kathleen E. Wage (George Mason Univ., 4211 Ridge Top Rd., Apt. 2114, Fairfax, VA 22030, farrokhrooz1@gmail.com)

Farrokhrooz *et al.* [*J. Acoust. Soc. Am.*, 2017] measured the seasonal variations of ambient noise in the 40–50 Hz band using long vertical arrays deployed during SPICEX, a 2004–2005 experiment in the northeastern Pacific. This talk compares simulations of low frequency noise from distant sources with the SPICEX data. The dominant sources of distant noise in the 40–50 Hz band are ships and wind. Noise from surface shipping and wind sources is transferred into the deep sound channel by several mechanisms [Dashen and Munk, J. Acoust. Soc. Am. 1984] and propagates long distances. The simulations use shipping density maps from exactEarth, Ltd., and measured wind speeds from European Centre for Medium-Range Weather Forecasts (ECMWF). A parabolic equation model that includes internal wave effects is used to generate Monte Carlo trials for different realizations of the shipping and wind environment. Results indicate that the seasonal variations in the SPICEX data are dominated by distant wind sources, including storms and persistent high latitude winds during the winter season. This comparison of simulation and experiment supports Bannister's conclusions [J. Acoust. Soc. Am., 1986] regarding the role of high latitude winds in generating low frequency ambient noise. [Work sponsored by ONR.]

8:15

4aUW2. Assessment of the proportion of anthropogenic underwater noise levels in passive acoustic monitoring. Mirko Mustonen, Aleksander Klauson (Dept. of Civil Eng. and Architecture, Tallinn Univ. of Technology, Ehitajate tee 5, Tallinn 19086, Estonia, mirko.mustonen@gmail.com), Mihkel Tommingas, and Julia Berdnikova (Thomas Johann Seebeck Dept. of Electronics, Tallinn Univ. of Tecnology, Tallinn, Estonia)

The marine environment faces a pressure from the increasing shipping intensity in the form of rising levels of continuous anthropogenic underwater noise. Current underwater noise monitoring guidelines advise to measure the long-term trends in the overall noise levels in the frequency bands where ship noise is most prevalent. However, the natural noise is omnipresent and prolongs the time period required for the detection of statistically significant trends in the overall noise. The monitoring efficiency can be improved by finding the proportions of the anthropogenic and the natural noise levels and by measuring the changes in the proportions over time. These proportions can be found by differentiating between the two types of ambient noise in the recordings according to both proximity of the ships and the variability of the environmental conditions. This is achieved by using the AIS ship traffic data along with the ship noise detection algorithms. The AIS data enable to determine the position of ships around a noise monitoring location and calibrate the ship noise detection algorithms. The results and the methods are presented for the passive acoustic monitoring in the Baltic Sea and applicability of the described methods are discussed.

8:30

4aUW3. The marine soundscape off the Isle of Mull in Scotland's Inner Hebrides. Adele Roland, Kathleen E. Wage, and E. C. M. Parsons (George Mason Univ., 4400 University Dr., Fairfax, VA 22030-4444, roland.adele@gmail.com)

The waters around the Isle of Mull, Scotland are home to minke whales, basking sharks, common and bottlenose dolphins, harbor porpoises, and many fish and marine invertebrates. This talk describes the soundscape of this region, measured as part of a study on minke whale (*Balaenoptera acutorostrata*) acoustic habitat use, conducted June–September 2016. The study area consists of shallow (<100 m depth) shelf waters between and around the Islands of Mull, Coll, and Muck. Measurements were made with a Soundtrap 300 from multiple points over the study area (~560 km^2) at a sampling rate of 288k Hz. Recordings were taken at a depth of ~30 m from a whale watching vessel when it was stopped (usually observing marine wildlife). The study includes 131 recordings with durations of 5–40 min. Spectral density varies with location and time over the study. Temporal and regional variation and the identity of various sounds are analyzed. Most of the acoustic energy in the region is concentrated below 50 kHz. The soundscape is dominated by wave noise, snapping shrimp, and distant dredging. Local areas or times may be dominated by transient biological sounds, e.g., the calls of common dolphins. Regular but infrequent ferry passage temporarily alters the base soundscape.

8:45

4aUW4. The acoustic energy conversion efficiency of a single raindrop. Dajing Shang, Qi Li, Shu Liu, and Fangzhou Deng (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Nangang District Nantong St. No. 145 Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn)

The measurement of rainfall in the ocean is more difficult than in land, but the noise signals produced by rainfall can be used for measuring the rainfall in the ocean. In this paper, a single raindrop measurement system was set up to measure the noise of the raindrop, the bubble sound is classified by the radius of the raindrop which was measured by the splash method. The mechanism of the initial impact and the bubble sound are analyzed, and the kinetic energy threshold and acoustic energy conversion efficiency of the single raindrops is also investigated. The results show that the bubble sound is the main compared with initial impact sound, the initial impact sound will become larger with the radius of raindrop, the bubble acoustic energy of large and great raindrop is much larger than that of small one, but the tiny and medium raindrop have no bubble acoustic energy. The kinetic energy threshold is not a constant, but proportional to the raindrop size. The acoustic energy conversion efficiency is about 1.04×10^{-3}% for small raindrops, but 10^{-5}% for heavy rain and great raindrops.

9:00

4aUW5. Underwater ship noise pattern detection and identification. Julia Berdnikova (Thomas Johann Seebeck Dept. of Electronics, Tallinn Univ. of Technol., Ehitajate tee 5, Tallinn 19086, Estonia, julia.berdnikova@ttu.ee), Aleksander Klauson, Mirko Mustonen (Civil Eng. and Architecture, Tallinn Univ. of Technol., TALLINN, Estonia), and Mihkel Tommingas (Thomas Johann Seebeck Dept. of Electronics, Tallinn Univ. of Technol., Tallinn, Estonia)

The anthropogenic underwater noise event could be identified by spectral analysis and acoustic pattern pre-classification of previously measured noise sources. This paper discusses real-time applicable methods for separation of purely natural noise recordings when ships are absent from the data polluted by the ship noise. The natural noise periods are used for the statistical modeling of underwater channel environment and relative levels of marine ambient noise (Wenz curves). The anthropogenic noise periods allow us to identify the noise sources. Moreover, most of the ship identification methods require spectrum component processing and pre-determined ship classification, which could be based on this automatic separation results. The correlational and multiple criteria detection methods compared with cyclostationary feature detection and widely used energy detection. False alarm and misclassification minimization could be achieved by multiple sensor data fusion: weather and Automatic Identification System (AIS). The measured data were collected near the seafloor with an autonomous long-term passive acoustic recorder and were combined with ship passage information from the AIS. Three years of measurements include multiple individual ship observations with different speeds and directions. Analysis of the experimental results emphasizes the importance of pre-classification, especially in multiple target cases.

9:15

4aUW6. Ship source strength estimation in shallow water. Aleksander Klauson and Mirko Mustonen (Civil Eng. and Architecture, Tallinn Univ. of Technol., Ehitajate tee 5, Tallinn 19086, Estonia, aleksander.klauson@ttu.ee)

Continuous underwater noise from the commercial ship traffic is an important pressure on the marine environment. The environmental assessment of this pressure often includes both measurements and modeling. In order to model ship noise based on AIS traffic data, it is essential to provide a better Source Level (SL) input values for the individual ships. The SL can be estimated using underwater noise recordings along with the identified position of the ships. A good estimate of the SL in the deep water can be obtained by application of the spherical spread model. In the shallow water such an approach is inapplicable because of multiple interactions of the sound waves with the sea surface and the bottom and therefore sound propagation modeling should be applied. If the modeling is performed for some particular frequency bands, estimation of the transmission loss needs even more calculation effort. Simplified approach for more efficient calculation of the losses is proposed. The calculated results are compared with the measurements in different geographical positions and sea conditions. The topic of the ship source directionality is addressed. Repeatability of the results is checked for the different passages of the same ship during the year.

9:30

4aUW7. Resonant subwavelength acoustic panels with air inclusions for abatement of noise from underwater machinery and remotely operated vehicles. Colby W. Cushing, Kevin M. Lee, Andrew R. McNeese, Michael R. Haberman, and Preston S. Wilson (Appl. Res. Labs and Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu)

Deeply subwavelength thickness metamaterial panels consisting of encapsulated cylindrical air cavities cut from elastic plates were recently shown to provide broadband, low-frequency underwater noise suppression for frequencies above the resonance frequency of the cavity [*J. Acoust. Soc. Am.* **138**, EL245–EL257 (2015)]. The present work extends previous findings to investigate noise suppression applications associated with underwater machinery and remotely operated vehicles (ROVs). Air inclusions in the panels were sized for maximum attenuation of the frequency spectrum of the ROV and radiated pressure amplitude reduction as a function of frequency and void fraction was determined. To do this, panels were initially secured around an omnidirectional projector in a single discontinuous layer and the panel surface area was varied to quantify the effect of panel air volume fraction on the radiated noise level. Panels were then sized and arranged around an ROV and the measurements were repeated. Results will be presented and compared to an effective medium model. The system has proven to be a simple way to provide significant noise reduction for machinery and ROV-related noise which is of interest for shallow water applications. [Work supported by the US Navy Office of Naval Research.]

9:45–10:00 Break

10:00

4aUW8. Passive, broadband suppression of radiation of low-frequency sound. Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Plane gas-liquid interface becomes anomalously transparent for low-frequency sound, and essentially all energy radiated by a compact sound source in fluid is channeled into the gas when the source is within a fraction of wavelength from the interface [O.A. Godin, Phys. Rev. Lett. **97**, 164301 (2006)]. This is in contrast to high-frequency behavior, where the interface approximates a pressure-release boundary, and only a very small fraction of energy escapes the liquid. This paper investigates whether a similar effect of suppression of unwanted low-frequency sound radiation by underwater sources can be achieved in a wide frequency band by using compact, acoustically soft objects. Low-frequency asymptotics of acoustic Green's functions in the presence of spherical and cylindrical scatterers are used to quantify radiation suppression by simple shapes. Dependence of the radiation suppression efficiency in a homogeneous fluid and in underwater waveguides on mechanical properties of the soft objects is discussed. Feasibility of passive suppression of underwater sound radiation with and without employing acoustic metamaterials is addressed. [Work supported by ONR.]

10:15

4aUW9. In-water impact pile driving sound source spectra comparison and the quest for a "Generic Spectrum". Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Melanie Austin (JASCO Appl. Sci. (USA) Inc., Anchorage, AK), and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

Noise generated from in-water impact pile driving for marine and coastal construction can impact marine mammals. Physiological effects to marine mammals include hearing impairments such as temporary and/or permanent threshold shifts. Because marine mammals' auditory response to sound levels is frequency-dependent, environmental impact analysis from noise exposure needs to consider noise spectrum in addition to broadband received level. In this study, spectral characteristics of measured noise levels from impact driving of 30-inch steel piles at four locations in Alaska and Washington were compared. Results show for all impact pile driving, most acoustic energies were between 100 and 1,000 Hz. Spectral levels below 100 Hz varied among the locations. Above 1,000 Hz, spectral levels decayed at a rate of 15 dB/decade for the three Alaska sites, but at 30 dB/decade in Washington. Despite these general characteristics, however, there does not seem to be a "generic spectrum" for the 30-inch steel pile impact driving. The variations among the spectral characteristics are likely due to differences in substrate, hammer energy, and bathymetry in different locations.

10:30

4aUW10. High-resolution spectral-element computation of underwater noises due to offshore piling. Arvin Manalaysay, Chanseok Jeong (Civil Eng., The Catholic Univ. of America, 20 Michigan Ave., N.E., Washington, DC 20064, manalaysay@cua.edu), and Hom Nath Gharti (Dept. of GeoSci., Princeton Univ., Princeton, NJ)

Offshore piling has been effective in building foundations of offshore structures, such as wind turbines, bridges, and oil rigs. Despite such merits, underwater noise due to offshore piling is considered to be its critical

setback. Pile driving creates a high-level underwater sound that harms marine ecosystems. There have been studies that successfully predicted these noises by using numerical methods, for example, Finite Element Method (FEM). However, there has been no FEM study that considers anti-symmetric irregular domains and complex bathymetry to compute underwater noises in the high-frequency range (>1000 Hz) due to expensive computational costs of FEM. To bridge the gap, this work attempts to explore a novel, powerful simulation tool to efficiently obtain offshore piling noises in complex settings and in the high-frequency range. We adopted and modified an open-source large-scale parallel Spectral Element Method (SEM) wave simulator, SPECFEM3D. SEM is known to be much more efficient than FEM for wave propagation analysis problem of a very large number of elements and time steps without compromising accuracy. Our computational method can be used for prediction of offshore piling underwater noise and investigating novel piling methods, such as optimized shapes of piles or air bubbles curtains to mitigate underwater noise.

10:45

4aUW11. Sound radiation from a finite cylindrical shell with an irregular-shaped acoustic enclosure. Desen Yang, Rui Zhang, Shengguo Shi, and Tengjiao He (Underwater Acoust. Eng., Harbin Eng. Univ., NanTong Str NanGang Dist, Harbin 150001, China, 474996229@qq.com)

In practical situations, large machinery equipments are usually located in underwater vessels and change the regular-shaped cavity into an irregular one. Due to the existence of the machinery equipments, the sound transmission and radiation are difficult to express analytically. This paper models and analyses the noise radiation of a cylindrical shell excited by an internal acoustic source. The cylindrical shell contains a machinery equipment modelled as a rectangular object attached to shell with a spring-mass system. The acoustic field of cavity is computed by an integro-modal approach. Effects of object size on the coupling between acoustic modes and structural modes are investigated. Meanwhile, the relationship between volume of the object and sound radiation is studied. Numerical results show that the existence of the object gives rise to a more effective coupling between the structure and the enclosure compared with a regular-shaped cavity.

THURSDAY AFTERNOON, 7 DECEMBER 2017 — STUDIO 9, 1:00 P.M. TO 5:35 P.M.

Session 4pAA

Architectural Acoustics: Back to the Future: A Look at Multipurpose Spaces, How They've Changed, and What's Next

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

Jennifer Nelson Smid, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. A brief history of the multi-use Theater form. Robin S. Glosemeyer Petrone (Threshold Acoust.com, 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com)

Multi-use theatres provide communities with a venue to accommodate almost any performance type that requires a stage. The advent of structural steel, mechanical ventilation, and amplifications systems are a few advances of the late nineteenth and early twentieth centuries that have given shape to the multi-purpose theater; how have they shaped the form we see today?

1:25

4pAA2. Active acoustics in multi-purpose venues: A ten year retrospective. Steve Ellison and Melody Parker (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com)

Multi-purpose venues better serve diverse programs by tailoring their acoustics for each use condition. Active acoustic systems use electronics to provide wide-ranging acoustic adjustability with the press of a button. Over the past decade, the Constellation acoustic system by Meyer Sound has been deployed in over one hundred venues, supporting the performing arts, worship, education, and corporate sectors. This paper presents a survey of such venues and discusses lessons learned.

1:45

4pAA3. The interactive multipurpose performing arts hall. Hyun Paek, Gary W. Siebein, Jennifer R. Miller, Marylin Roa, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com)

Each multipurpose performing arts hall has a unique acoustic signature. The success of a modern multipurpose hall relies on how the disciplines of architecture, interior design, stage lighting and rigging systems, and acoustical design are coordinated to very small tolerances. The shaping and finish materials are integrated both visually and with precision to create the sound field of the room to accommodate a variety of venues. Two case studies are presented. The first is a 1000 seat multipurpose performance hall in central Florida which accommodates the use of variable acoustics, a specially designed orchestral shell, stage lift, and all visible interior spaces sculpted to provide richness of both amplified and natural acoustics. The second is a performance hall for a magnet high school for the performing arts in western Georgia that has undulating waves of aesthetically integrated panels on the walls that provides a stunning multipurpose space even with economic constraints. The case studies present different approaches to acoustical design of multipurpose theaters. The first multipurpose theater accommodates the needs of varying visiting performers while the second case study focuses on the presentation of students of performing arts that needed an economical but aesthetic center piece of a magnet school. This interactive process between the design team ultimately results in the beauty of the interactive participation of the audience and the performers also.

2:05

4pAA4. Changing an existing performance space from a single purpose use to a multipurpose space or creating a multipurpose space from an existing performance space not acoustically suitable for any performances. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, bob@rccoffeen.net)

Occasionally, a performance space is created for a particular use such as drama and later it realized by the owner and the users that the space should be acoustically suitable for multipurpose use. Also performance spaces are encountered that do not suitably serve performances of any type from an acoustical viewpoint but that must properly serve performances of various types. Examples of both situations are discussed and acoustical data for both situations will be presented.

2:25

4pAA5. Effects of variable acoustic elements on the spatial sound field in multipurpose venues. Michelle C. Vigeant, Matthew T. Neal, and David A. Dick (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vigeant@engr.psu.edu)

The design of multipurpose venues typically includes variable acoustic elements which adjust the acoustics of the space to support the intended use. The most common approach to vary the acoustics of a venue is to use variable absorption, often in the form of heavy drapery and/or acoustic felt banners. These elements are often characterized by changes in reverberation time, but their position also impacts the spatial distribution of room energy. In order to study the effects of variable acoustic elements on the three-dimensional sound field at specific audience locations, spatial impulse response measurements were taken in a number of venues using a 32-element spherical microphone array. The venues ranged from a small recital hall of 400 seats to a typical multipurpose hall with 1300 seats to a 2500-seat concert hall with variable acoustics. Beamforming techniques were used to analyze the effects of the variable acoustic elements on the spatial and temporal distribution of sound energy at several receiver locations in each venue. The sound fields in these venues will be compared to those measured in dedicated concert halls, which have been measured for recent work on the topic of listener envelopment. [Work supported by NSF Award 1302741.]

2:45

4pAA6. Multi-purpose performance spaces as vehicles to enhance the acoustical communities of cities and towns. Gary W. Siebein, Marylin Roa, Jennifer R. Miller, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Soundscape theory can offer insights into methods that can transform the design of multi-purpose performance venues so that they can come closer to fulfilling their potential as a vibrant facility at the artistic and cultural core of a community every day of the year all day long. This is a design theory guided by technical innovation and philosophical insight that can bring together diverse groups of users and performers; experimental building, acoustical and theatrical systems; and sophisticated, integrative design methods. Case studies of several architectural design studio projects that ask questions about the nature of performance in the city will be used to describe an approach to integrated architectural and acoustical design based in soundscape theory that brings forth new approaches to using sound to shape space rather than designing spaces and inserting acoustical envelopes within the spaces based on transformative mapping of interior and exterior space created by sound and the ways that it is enhanced when it interacts with buildings.

3:05

4pAA7. Trends in performing arts center design. Jack P. Hagler (Theatre Planners, Schuler Shook, 325 N Saint Paul St., Ste. 3250, Dallas, TX 75201, jhagler@schulershook.com)

The world of performing arts continues to evolve. Audiences have changed. With every show, contemporary live entertainment delivers an updated, but still unique, experience. Operas of the 18th century are the Broadway shows of the 21st century. Orchestras of the 1700s are the Thievery Corporation of today. Lillie Langtry is now Lady Gaga. Performing arts centers must embrace the concept of popular entertainment inside and outside of the audience chamber to keep audiences excited, engaged, and attending. Contemporary entertainment attracts a contemporary audience—a technology-driven generation with a new set of expectations when it comes to a night at the theatre. Here are trends we've observed in our practice. Convertible audience rooms and floors for more versatility and flexibility; PAC reputation, space utilization, and the decline of attendance for classical symphony orchestra programming; audience immersion; creating a more social experience in the audience chamber and beyond; making a night (and day) of it with better food and beverage offerings; safety, security, and the new norm; animation of architecture. This is a discussion of the shift paradigm of performing arts centers from the classic arts to the contemporary arts and how the acoustician might respond to these contemporary trends.

3:25–3:40 Break

3:40

4pAA8. Case study: Eccles Center for the Performing Arts. Ronald Freiheit and Matthew S. Hildebrand (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

Cinemas and movie theatres have specific room acoustic requirements that are well understood in the industry. Recent technologies such as Dolby Atmos and DTS:X provide an exciting spatial audio experience using a high number of audio tracks dynamically rendered to loudspeakers located in both the overhead and lateral planes. But what happens when the same venue is utilized for both motion picture screenings and acoustic performances? In this paper, active acoustics processing is integrated into existing cinematic surround sound architecture to provide a unique solution that promises flexibility rather than compromise. Design objectives, limitations, and performance of the integrated systems are reviewed.

4:00

4pAA9. Jazz: Multipurpose art form meets multipurpose space. Evan Delli Paoli (Holzman Moss Bottino Architecture, 90 BRd. St., Ste. 1803, New York, NY 10004, edellipaoli@holzmanmossbottino.com)

The contemporary multipurpose performance space is not one that can exist without its evolution from Ancient Greek Amphitheaters to Elizabethan theaters to roaring twenties Movie Palaces. Its accelerating trajectory and relevance are directly correlated to the evolution of the art forms taking place and unfolding from within them. One such art form is Jazz. This session will explore episodic moments throughout the history of the Jazz art form that draw parallels to pivotal moments in the evolution of multipurpose performance spaces as we know them. The session is organized around five specific case studies: (1) Dixieland and The Rise of Armstrong; (2) Ellington, The Jazz Orchestra, and The Music Halls; (3) Miles: Birth of the Cool, Bitches Brew and Jazz Reinvention; (4) Gillespie and the melting-pot of Afro-Cuban Jazz; and (5) Jazz Fusion and found space. In each case study, a moment in Jazz history will be used as an analogy to introduce the architectural and acoustic concepts specific to a particular typology of multipurpose space—accompanied by a specific built building project example of how these concepts physically manifest themselves.

4:20

4pAA10. Hybridization of performance venues. Wendy Pautz and Julie Adams (Architecture, LMN Architects, 801 Second Ave., Seattle, WA 98104, jadams@lmnarchitects.com)

It should come as no surprise that the future of the performance space is wedded to changing demographics and increasingly pluralistic artist expression. New models are needed to survive the rapid societal change—socially, culturally, and economically. While multipurpose functionality has long since become the norm for both civic and community arts venues, we are now seeing the advent of hybrid civic facilities—venues that serve other forms of public gathering to complement arts programming. This proposition entails rethinking performance venue typologies at a fundamental level to ensure a sustainable future. From the multiple perspectives of a diverse panel of industry experts, this session will explore the underlying factors that shape current and future venue trends, including capacity to serve a wide range of artistic content as well as other civic and community programming. The following project case studies will be examined as reference points to the discussion: *The Tobin Center for the Performing Arts, San Antonio Federal Way Performing Arts and Event Center, Federal Way, Washington Octave 9 (experimental performance venue under development for the Seattle Symphony Orchestra)* Audience engagement and spirited dialogue around this topic is encouraged and expected!

4:40

4pAA11. A trio of multi-purpose halls, 1995–2016: Aronoff, Overture, and Hancher. Joseph W. Myers (Kirkegaard Assoc., 801 W. Adams St. 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

The Aronoff Center in Cincinnati OH (1995) includes a 2700-seat multi-purpose hall used as a roadhouse, with a particular focus on touring Broadway shows. The Overture Center in Madison, WI (2005), has a 2250-seat multi-purpose hall that is the home to a professional symphony orchestra and opera company, as well as hosting touring Broadway shows. Hancher Auditorium in Iowa City IA (2016) has an 1,800-seat multi-purpose hall that acts as a roadhouse for a broad variety of performance types, as a presenter for dance, and as the largest performance space for the University of Iowa's arts programs. All three performing arts centers had Pelli Clarke Pelli as design architect, Theatre Projects Consultants as theater consultant, and Kirkegaard Associates as acoustics consultant. This presentation examines what these halls have in common, how they are different, and what drove these differences. It looks specifically at the degree to which the differences are a design response to the particular program of each room and the degree to which they reflect changes over time in design approach or design requirements.

5:00

4pAA12. Multi-purpose halls… Wave of a new future or dead-end? Paul H. Scarbrough and Christopher Blair (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, pscarbrough@akustiks.com)

Multi-purpose halls, that is, performance spaces that can adapt their acoustical and technical features to serve many different performance types are a uniquely North American invention. Europe, where the modern-day symphony orchestra, opera company, ballet company, and drama troupe all originated, favors halls that are purpose-built to serve each art form. In the United States and Canada, even in cities that could afford to field multiple ensembles or companies, building dedicated facilities for each art form was often a bridge too far. This gave rise to the multi-purpose hall and with it the old saw that multi-purpose meant no-purpose. The success of new venues in Charleston, San Antonio, León, México, and elsewhere demonstrates that multi-purpose halls are cost-effective solutions and that they need not be the compromises they once were. The authors will explore the development of multi-purpose halls starting with

Louis Sullivan's Auditorium Theater in Chicago, arguably the forerunner of the contemporary multi-purpose hall, through to the current day. The authors will also explore how the evolving creative impulses of artists and directors is challenging designers to go even further, and incorporate degrees of flexibility into these spaces that would have been unimaginable just two decades ago.

Contributed Paper

5:20

4pAA13. Sixteen years later—Skirball Center for the performing arts at NYU completes the installation of Electronic Architecture. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

The Skirball Center for the Arts at New York University was designed to be a multi-purpose venue with Electronic Architecture providing variable acoustics. Although the infrastructure (wiring, loudspeakers) was integrated during construction in 2000, the system was not completed and remained idle since that time. In 2016, New York University decided to complete the system installation. We will discuss the nature of the venue and its programming, the reasons that the system remained dormant for so long, and the changes that sparked interest in completing the system.

THURSDAY AFTERNOON, 7 DECEMBER 2017 SALON F/G/H, 1:00 P.M. TO 3:05 P.M.

Session 4pAB

Animal Bioacoustics: Neurophysiology of Echolocation

Dorian S. Houser, Chair
National Marine Mammal Foundation, 2240 Shelter Island Drive, San Diego, CA 92106

Chair's Introduction—1:00

Invited Papers

1:05

4pAB1. Biosonar gain control in odontocetes: Evoked-potential studies. Paul E. Nachtigall and Alexander Supin (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, nachtiga@hawaii.edu)

The biosonar of odontocetes processes echo signals with a wide range of echo levels. Several mechanisms that serve to compensate for the echo-level variation (biosonar gain control) have been revealed: (1) the adjustment of the emitted sonar pulse levels (the longer the distance to the target, the higher the level of the emitted pulse), (2) the release from forward masking of the echo by the preceding self-heard emitted pulse (the longer the distance to the target, the more release of the echo-response from masking), and (3) an active control of hearing sensitivity (the lower the echo level, the higher the sensitivity). Investigations using the auditory evoked potential technique have demonstrated that these processes manifest different aspects of the functioning of a common gain-control system. The control of the emitted sonar pulse level does not change the echo-to-emitted pulse ratio but makes forward masking able to function at various target distances. The active control of hearing sensitivity also adjusts forward masking to function at various target distances. These combined processes are capable of producing an effective reduction in the variation of the level of the response to the echo when the target strength and target distance vary within a wide range

1:25

4pAB2. Echo jitter delay discrimination in bottlenose dolphins (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), James J. Finneran (U.S. Navy Marine Mammal Program, San Diego, CA), Dorian S. Houser, and Ryan Jones (National Marine Mammal Foundation, San Diego, CA)

Experiments with bats show echo delay discrimination capabilities on sub-microsecond scales, which is an acuity that is much higher than that predicted by the auditory nervous system's ability to directly encode high-frequency phase information (i.e., through neural phase locking). To provide a cross-species comparison, echoic discrimination experiments were conducted with two bottlenose dolphins. Dolphins were trained to echolocate a "phantom" target and report a change from a target with a static delay of ~12 ms to one with alternating delay, or "jitter," imposed on the static delay. Performance was measured as a function of jitter delay. Both dolphins were able to reliably detect jitter down to ± 1 μs, although lower amounts of jitter were not detected. When the jittered echo was phase shifted by 180° relative to the static echo, performance was near 100% at all jitter delays, including ± 0 μs. Both dolphins utilized intermittent

patterns of click emissions, which were unusual for the relatively short ranges employed. Bottlenose dolphins do not appear to possess the sub-microsecond delay acuity capabilities observed in some bats. However, the current results suggest that dolphins can encode information resembling echo phase information in determining target range. [Funding from ONR.]

1:45

4pAB3. Narrowband auditory brainstem responses to "self-heard" and external clicks in the bottlenose dolphin. Dorian S. Houser, Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Carolyn E. Schlundt (Government IT Services, San Diego, CA)

Knowledge of the emitted click or chirp in echolocating dolphins and bats is believed to be critical to using echo delays to reveal target range and shape. In dolphins, the neural representation of the "self-heard" click should not equate to that of a click measured in the far-field due to sound propagation path differences and temporal dispersion within the cochlea. To investigate these differences, auditory brainstem responses (ABRs) to self-heard clicks masked with noise bursts having various high-pass cutoff frequencies were measured in two dolphins. A passive listening experiment was also conducted in which similarly masked external, spectrally "pink" clicks were used as stimuli. In both experiments, narrowband ABRs were derived using a high-pass subtractive noise technique. Latencies of the ABR to the external click demonstrated frequency-dependent latency shifts. Latencies of the ABR to the self-heard click were delayed relative to those of the passive listening experiment and similar across frequencies from ~28 to 113 kHz, suggesting that neural responses to the self-heard click are synchronous within the bandwidth of echolocation. Longer ABR latencies are potentially due to spectral differences between external and self-heard clicks, click-induced forward masking, and possibly, neural inhibition associated with click production.

2:05

4pAB4. Selectivity of bat midbrain neurons to stimulus elements embedded in natural echolocation sequences. Silvio Macias, Jinhong Luo (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Biology-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

We investigated response selectivity of single neurons in the inferior colliculus (IC) of the big brown bat, *Eptesicus fuscus*, to acoustic elements embedded in the natural temporal patterning of dynamic frequency modulated (FM) echolocation sequences. Acoustic stimuli consisted of species-specific sonar sequences, with natural dynamic spectro-temporal features, containing both bat-emitted FM pulses and echoes from a moving target. Roughly 50% of sampled IC neurons showed responses to a restricted subset of stimulus elements in the natural sonar sequence and selectivity to the time delay between pulses and echoes, the bat's cue for target distance. Importantly, response selectivity emerged only when stimulus elements were embedded in natural sequences and was absent when the same neurons were stimulated with isolated pulse-echo pairs presented randomly at 300 ms stimulus intervals. These data provide compelling evidence that natural and dynamic sound sequences are necessary to evoke selectivity to pulse-echo delay in midbrain neurons of an FM bat. Our findings suggest that the FM bat's fine temporal control of pulse interval during insect pursuit contributes to neuronal selectivity that can serve to sharpen acoustic imaging by sonar.

2:25

4pAB5. Dynamic representation of 3D auditory space in the midbrain of the free-flying echolocating bat. Ninad B. Kothari, Melville Wohlgemuth, and Cynthia F. Moss (Johns Hopkins Univ., 3400, N. Charles St., Ames Hall Rm. 124, Baltimore, MD 21218, ninadbkothari@jhu.edu)

Our natural world is three-dimensional. A fundamental requirement of spatial orientating behaviors in the natural environment is the representation of 3D sensory space. Despite the importance of 3D sensory coding of a natural scene to guide movement, most neurophysiological investigations of this problem have been limited to studies of restrained subjects, tested with 2D, artificial stimuli. Here we show for the first time that auditory neurons in the midbrain superior colliculus of the free-flying echolocating bat encode 3D egocentric sensory space, and that sonar-guided inspection of objects in the environment sharpens spatial tuning of single neurons. Combining wireless multichannel neural recordings from free-flying bats, synchronized with video and audio data, and an echo model that computes the flying animal's instantaneous, stimulus space, we demonstrate 3D echo-evoked receptive fields of single auditory midbrain neurons in animals orienting in a complex environment. We discovered that the bat's active sonar inspection of objects dramatically tightens range tuning of single neurons and shifts peak activity to represent closer distances. Our research demonstrates dynamic 3D space coding in a freely moving mammal engaged in a real-world navigation task.

2:45

4pAB6. Non-auditory, electrophysiological potentials preceding biosonar click production in bottlenose dolphins. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, Ryan Jones, Dorian S. Houser, Alyssa W. Accomando, and Sam H. Ridgway (National Marine Mammal Foundation, San Diego, CA)

The auditory brainstem response (ABR) to a dolphin's own emitted biosonar click may be measured by averaging epochs of the instantaneous electroencephalogram (EEG) that are time-locked to the emitted click. In this study, waves in the averaged EEG preceding the biosonar click-evoked ABR were measured using surface electrodes placed on the head in six configurations while dolphins performed an echolocation task. Simultaneously, clicks were measured using contact hydrophones on the melon and a hydrophone in the farfield. The results revealed an electrophysiological potential (the pre-auditory wave, PAW) preceding the production of each biosonar click. The largest PAW amplitudes occurred with the non-inverting electrode just posterior of the blowhole and right of the midline — the apparent side of biosonar click generation. Although the source of the PAW is unknown, the temporal and spatial properties rule out an auditory origin. The PAW may be a myogenic potential associated with click production; however, it is not known if muscles within the dolphin nasal system can be actuated at the rates reported for dolphin click production, or if sufficiently coordinated and fast motor endplates of nasal muscles exist to produce a PAW detectable with surface electrodes. [Work supported by ONR.]

Session 4pAO

Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics: Biological Effects on Seabed Geoacoustic Properties II

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, 10000 Burnet Road, Austin, TX 78758

Megan S. Ballard, Cochair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, 10000 Burnet Road, Austin, TX 78758

Kelly M. Dorgan, Cochair
Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528

Invited Papers

1:00

4pAO1. Impact of organic matter on the physical and dynamic properties of marine mud. Richard H. Bennett (SEAPROBE, Inc., 501 Pine St., Picayune, MS 39466, rhbenn_seaprobe@bellsouth.net), Matthew H. Hulbert (Res. Dynam., West Chester, PA), and Roger Meredith (Retired, Slidell, LA)

The physical and dynamic properties of marine mud are a function of the interaction of clay fabric, organic matter (OM), and seawater physico-chemistry, and, when present, free gas. OM is the focus of this presentation and is a determinant of several properties in mud deposits including (1) *free water* available versus total water content in clay fabric pore space as a result of (2) *OM seawater hydration,* (3) reduction of *permeability* by volumetric contribution of hydrated OM in pore space, (4) *OM density* reduction *by seawater hydration* versus dry OM density, (5) *dynamic behavior* by physico-chemical attachment of OM to clay particles in potential energy fields created by the clay fabric signatures (edge-to-face, edge-to-edge, and offset face-to-face) in seawater, (6) high percentages of OM (TOC >2%) result in a *high degree of compressibility* at considerably lower stress compared to mud deposits with <2%TOC. Measurement and characterization of OM, previously often neglected, is expected to provide new significant insights and understanding of marine mud physical and dynamic properties and contribute to more reliable research data for scientists and engineers. Handling and storage methods used for mud samples containing OM are critical for accurate results.

1:20

4pAO2. Biological effects and the Grain-Shearing model of wave propagation in unconsolidated sediments. Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

The Grain-Shearing (GS) model of wave propagation in unconsolidated sediments is embodied in two sets of dispersion relations, one for the compressional wave and the other for the shear wave. Besides frequency, these expressions for the wave speed and attenuation depend on the physical properties of the sediment, notably the bulk density and porosity. They also involve a material exponent, n, and grain shearing coefficients that are associated with the sliding of grains against one another, which is a non-linear, stick-slip type of process that is controlled by micro-asperities on the surface of the grains. Bioturbation is known to affect both the bulk density and porosity of sediments, while biogenic coatings of mineral grains modify the surface roughness, which alters the grain-shearing coefficients. The magnitude of the effects of bioturbation and biogenic coatings on the wave speeds and attenuations predicted by the GS theory will be discussed, based on available biological information in the literature. [Research supported by ONR.]

1:40

4pAO3. Microbiota and macrobiota alterations impact acoustic predictions of geotechnical sediment properties. Allen H. Reed (Marine Geosci. Div., Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529, allen.reed@nrlssc.navy.mil)

Computed tomography supplies a three-dimensional vision of sedimentary geometries that are fabricated by microscopic and macroscopic biota. The biota alter sediment structure and properties in such a way that the altered sediment structure impacts acoustic determinations of idealized seafloor properties, which are based upon grain size and associated, yet textbook, geoacoustic properties. The presence of these alterations affects propagation by scattering acoustic energy at density discontinuities and non-planar surfaces (aka ”seafloor roughness”) both of which have a likely impacts: dampening of signal strength and increases in reverberation. The result is that the information sought about the seafloor is obscured; sediment properties, composition, strength, and potential for deformation and mobility are ambiguous and poorly constrained from available acoustic information in areas where biological activity significantly alters the sediment structure. Here, the three-dimensional geometry of the microbiotic and macrobiotic impacts on sediments are presented in

images that display unique features, which are often discounted in sediment acoustic models, but which should be considered as important and highly relevant sources of uncertainty with respect to acoustic determinations of the seafloor geotechnical and geophysical properties.

2:00

4pAO4. Comparison between infauna abundance and seabed geoacoustic properties. Kevin M. Lee, Megan S. Ballard, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Will M. Ballentine, Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

To investigate the effects of infauna on seabed geoacoustic properties, a field experiment was conducted during May 2017 in Petit Bois Pass, near the mouth of Mobile Bay, Alabama. *In situ* measurements of seabed geoacoustic properties were collected using hydrophones to generate and receive compressional waves from 5 kHz to 100 kHz and bimorph transducers to measure shear waves from 300 Hz to 1.5 kHz. The measurement system was deployed multiple times at three distinct sites within the pass, and acoustic measurements were conducted at depths ranging from 5 cm to 20 cm into the sediment. For each deployment of the acoustic system, diver cores were collected. A subset of the cores were sieved on site to collect infauna, and the remaining cores were taken back to the laboratory where they were sectioned and analyzed for porosity and grain size distribution. Comparison between compressional and shear wave speed and attenuation, the sediment geotechnical properties, and the distribution and abundance of infauna will be presented. [Work supported by ONR and ARL:UT IR&D.]

2:20–2:35 Break

2:35

4pAO5. Core and resonance logger (CARL) measurements of fine-grained sediments containing infauna. Gabriel R. Venegas, Kevin M. Lee, Megan S. Ballard, Andrew R. McNeese, Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu), and Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL)

Benthic infauna perform important ecological functions such as water filtration, the recycling of organic matter, and forage for larger predators. However, how infauna affect bulk acoustical properties of the sediment is less studied and warrants further investigation. During a field experiment conducted in Petit Bois Pass, AL, multiple cores of fine-grained sediments containing infauna were collected by divers and analyzed using a core and resonance logger (CARL). CARL is a core-logger capable of performing both conventional pitch-catch measurements radially along the core as a function of depth and lower frequency resonance measurements. Resonance measurements were performed by exciting acoustic modes within the core and sensing them externally along the core's length. By utilizing both pitch-catch transducers as receivers, symmetric and asymmetric modes can be identified, and sound speed can be inferred as a function of frequency. The frequency ranges for the pitch-catch and resonance measurements were 100 kHz to 400 kHz and 15 kHz to 30 kHz, respectfully. Subsequently, the cores were sectioned and were either sieved for infauna or analyzed for porosity and grain size distribution. The measured sound speeds will be compared with biological and geological properties of each individual core. [Work supported by ONR.]

2:55

4pAO6. Effects of marine infauna on the acoustic properties of sediment. Will M. Ballentine, Kelly M. Dorgan (Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528, wballentine@disl.org), Kevin M. Lee, Megan S. Ballard, Andrew R. McNeese, Preston S. Wilson, and Gabriel R. Venegas (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Marine infauna alter the surrounding habitat in many ways. Compact mud burrows, tubes built from shell hash, and large subsurface galleries are a few examples of these alterations. Structural changes such as these can have varying effects on the geoacoustic properties of sediment. Here, we investigate how infauna may affect the sound speed and attenuation in sediments both in natural diverse communities of organisms, and in laboratory mesocosm experiments using controlled monocultures of species representing potentially important functional groups. Field studies were conducted at Petit Bois Pass off the coast of Dauphin Island, Alabama in May 2017 in which sediment cores were collected and brought back to the lab for acoustic measurements. These measurements can be directly compared with those taken *in situ* using a deployable field apparatus. For sediments with both natural and manipulated communities of infauna, sound speed and attenuation were measured at multiple depths and at high frequencies with wavelengths corresponding to the scales of expected impacts of individual organisms (100–400 kHz) to assess the effects of different infaunal functional groups.

3:15

4pAO7. Sediment characterization using normal-incidence echo sounding in a biologically active environment. Marcia J. Isakson, Michael Rukavina, and Johanna Owens (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

In collaboration with the Dauphin Island Sea Lab, normal-incidence acoustic measurements were taken in the Petit Bois Pass which is know to harbor benthic biology such as tube worms. The acoustic system used is easily deployed from a small craft and self-calibrating. The acoustic measurements were taken coincidentally with grab samples to characterize both the sediment type and the types of benthic biology. Data were also collected in an area void of benthic biology for comparison. These measurements were analyzed to assess the ability of normal-incidence measurements to determine sediment type in the presence of benthic biology and also to develop methods for characterizing benthic biology with acoustics. [Work sponsored by ONR, Ocean Acoustics.]

3:35

4pAO8. Low-frequency acoustic behavior of photosynthetically active seagrasses. Jay R. Johnson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Jean-Pierre Hermand (LISA Environ. HydroAcoust. Lab, Université libre de Bruxelles, Brussels, Brussels Capital, Belgium), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic remote sensing techniques are an important tool for mapping seagrass coverage. Three different photosynthesis-related processes can occur in seagrasses that affect the acoustic behavior. First, the air channels within the leaf pressurize with produced gas. Next, bubbles form on the leaf blades. Finally, some of these bubbles break off and enter the water column. A one-dimensional acoustic resonator technique was adapted to monitor the photosynthetic activity of two Mediterranean seagrasses, *Posidonia oceanica* and *Cymodocea nodosa*. Measurements of the low-frequency (1–8 kHz) effective sound speed of a mixture of seagrass leaf blades and artificial seawater were taken at regular intervals during periods of no direct light and exposure to photosynthetically active radiation. The acoustic response is compared to independent dissolved oxygen measurements and visual observation of bubble formation.

3:50

4pAO9. Variations in ultrasonic transmission behavior along seagrass leaf blades. Jay R. Johnson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Jean-Pierre Hermand (LISA Environ. HydroAcoust. Lab, Université libre de Bruxelles (ULB), Brussels, Brussels Capital, Belgium), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Seagrass is a complex multi-phase material, and an effective method for connecting acoustic propagation through seagrass meadows to internal or external characteristics of the seagrass would be beneficial for acoustic remote sensing applications. To investigate some of these connections, ultrasonic (1, 2.25, and 5 MHz) time-of-flight measurements through individual leaf blades of the endemic Mediterranean seagrasses *Posidonia oceanica* and *Cymodocea nodosa* are presented. Acoustic measurements were made at multiple points along the leaf blades and the sound speed and signal attenuation varied significantly within a single blade depending on measurement location. The measured acoustic variations are compared to external blade features such as discoloration, epiphyte coverage, and thickness. Microscopy images of blade cross-sections each taken at each acoustic measurement location are used to compare the void fraction to acoustic behavior. [Work supported by ONR, ONR Global.]

4:05

4pAO10. Acoustical characterization of a seagrass meadow in the Lower Laguna Madre. Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese, Jason D. Sagers (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Abdullah F. Rahman (School of Earth, Env., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX), Justin T. Dubin, and Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This talk presents preliminary results from an experiment conducted in the Lower Laguna Madre, Texas to characterize the physical and acoustical properties in a meadow of *Thalassia testudinum*. Concurrent measurements were collected using (1) acoustic probes, (2) side-scan and parametric sonar, (3) broadband propagation, and (4) sediment cores. The acoustic probes provided localized, direct measurements of sound propagation in the seagrass canopy as well as geoacoustic properties (compressional and shear wave speed and attenuation) of the seagrass-bearing sediment. The side-scan and parametric sonars were used to survey for seagrass abundance and sub-bottom layering. Broadband signals produced by a combustive sound source were recorded at several ranges by hydrophones and geophones and were used to infer geoacoustic properties of the seagrass and underlying sediment for rapid environmental assessment. The sediment cores were analyzed in the laboratory using both low-frequency resonator measurements and high frequency travel-time measurements to estimate compressional wave speed, after which they were sectioned and measurements of sediment grain size, porosity, and biomass were obtained. The combination of these data sets provides a unique characterization of the geoacoustic properties of a seagrass meadow. [Work sponsored by ARL:UT IR&D and ONR.]

4:20–4:40 Panel Discussion

Session 4pBA

Biomedical Acoustics: Imaging II

Marie M. Muller, Chair
MAE, NCSU, 911 Oval Drive, Raleigh, NC 27695

Contributed Papers

1:00

4pBA1. Application of laser ultrasound technique to evaluate wave velocity in bovine meniscus. Yoshitaka Sakata (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, duq0356@mail4.doshisha.ac.jp), Shunki Mori, Mami Kawase, and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

In vivo elasticity evaluation of meniscus and cartilage is necessary to understand the condition of the joint. Ishihara *et al*. have used the photoacoustic method with an endoscope.[1] Another possible technique is the laser ultrasound. Making use of the thermoelastic effect, we can stimulate a short longitudinal wave by irradiating the pulsed laser beam to the material. This technique enables noncontact measurements. As an initial study, we have measured the longitudinal wave velocities in a bovine meniscus sample using a pulsed laser (COHERENT Helios 1064-5-50). The meniscus samples were obtained from the knee joint of a bovine femur. The sample size was $4.00\times6.50\times6.00$ mm^3. We could excite short longitudinal waves around 30 MHz and measure wave velocities propagating in three directions (bone axis, posterior-anterior, and medial-lateral). The excited ultrasonic wave propagated through the meniscus sample and was received by a piezoelectric sensor. The velocity range was about 2950 m/s–3450 m/s, showing the highest velocity in the anterior-posterior direction. Velocities also depended on the water content and showed small heterogeneity. [1] M.Ishihara *et al*., Jpn. J. Appl. Phys. **42**, 556–558 (2003).

1:15

4pBA2. Time domain analysis of ultrasonic backscatter signals from cancellous bone. Jordan P. Ankersen, Phoebe C. Sharp, Joseph A. McPherson, Ann Viano, and Brent Hoffmeister (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, ankjp-18@rhodes.edu)

Ultrasonic backscatter techniques are being developed to detect changes in bone caused by osteoporosis. Most techniques analyze backscatter signals in the frequency domain by measuring quantities related to the power spectrum. Investigate the utility of two backscatter parameters determined from a time domain analysis of backscatter signals: the normalized backscatter amplitude ratio (nBAR) and the backscatter amplitude decay constant (BADC). A 3.5 MHz transducer was used to acquire backscatter signals from 54 specimens of bone prepared from 14 human femurs. nBAR was determined from the log of the ratio of the root mean square amplitude of two different portions of a backscatter signal. BADC was determined by measuring the exponential decay in the amplitude of a backscatter signal. nBAR and BADC both demonstrated highly significant ($p < 0.001$) linear correlations with bone density. However, the correlation coefficients were slightly stronger for nBAR ($0.79 \leq R \leq 0.89$) than for BADC ($0.67 \leq R \leq 0.73$). Parameters based on a time domain analysis of backscatter signals from bone may be sensitive to changes in bone caused by osteoporosis. Of the two parameters tested, nBAR demonstrated the strongest correlations with bone density. [**Funding**: NIH/NIAMS R15AR066900.]

1:30

4pBA3. Correlation of ultrasonic backscatter difference parameters with bone density in clinical ultrasound images. Abel Diaz, Ann Viano, Joseph A. McPherson (Dept. of Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, diaaa-18@rhodes.edu), Brian S. Garra (Div. of Imaging, Diagnostics and Software Reliability, US Food & Drug Administration, Silver Spring, MD), and Brent Hoffmeister (Dept. of Phys., Rhodes College, Memphis, TN)

The ultrasonic backscatter difference technique analyzes the power difference (in dB) between two gated regions of a backscatter signal and is being used to detect changes in bone caused by osteoporosis. The current study investigates correlations with bone density for these parameters determined from clinical ultrasound data. Ultrasonic backscatter images and signals were acquired from the hip and two vertebral bodies of human subjects using a 2.5 MHz phased-array transducer. Three backscatter parameters-normalized mean, slope, and intercept of the backscatter difference- were determined from the power difference between two gated regions of the signal. All three parameters were analyzed for 25 different gate choices. X-ray bone mineral density data for each subject were acquired for the three anatomical regions. Significant linear correlations ($p<0.05$) were found for all three ultrasonic parameters for at least one choice of gate parameters. R values ranged from to 0.38 to 0.63. For the first time, backscatter difference parameters measured *in vivo* have been shown to correlate with bone density. The correlation may depend on the gated regions of the backscatter signal chosen for analysis. [Funding: NIH/NIAMS R15AR066900.]

1:45

4pBA4. Modeling of the acoustic radiation force in elastography. Fabrice Prieur (Dept. of Informatics, Univ. of Oslo, P.O box 1080, Blindern, Oslo 0316, Norway, fabrice@ifi.uio.no) and Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Elastography is a non-invasive imaging technique that can assess *in vivo* tissue stiffness. Shear wave elastography imaging uses the acoustic radiation force (ARF) to produce shear waves. Numerical models rely on different formulations of the ARF. We present three existing formulations for the ARF: its full expression in the second-order approximation and two simplified formulations using a quasi-plane wave and an attenuated plane wave approximation. Analytical expressions for the ARF are derived for the special cases of a spherical concave source and for a quasi-Gaussian beam. They show discrepancies between the different formulations. For strongly divergent or highly focused beams the result from the second-order approximation differs from both simplified formulations. However the second-order and quasi-plane wave approximations create identical shear displacements. The k-Wave simulation package is used to compute the ARF and the ensuing transient displacement produced by an ultrasound probe. The second-order approximation and the quasi-plane wave approximation give different forces but identical displacements. The results using the plane wave approximation significantly differ. It is concluded that for highly focused transducers only the second-order approximation accurately estimates the ARF. For estimating shear displacements the second-order or quasi-plane wave approximation are equivalent and preferable to the plane wave approximation.

2:00

4pBA5. Combined subharmonic and ultraharmonic intravascular ultrasound imaging. Himanshu Shekhar (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Jeffrey S. Rowan, and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Abnormal proliferation of the vasa vasorum has been implicated in the rupture of atherosclerotic plaques. Imaging the vasa vasorum could help clinicians identify rupture-prone plaques and guide the choice of therapy. We hypothesized that subharmonic and ultraharmonic modes can be combined to improve the performance of contrast-enhanced intravascular ultrasound (IVUS) imaging. To test this hypothesis, vessel phantoms perfused with phospholipid-shelled ultrasound contrast agent (Targestar-P®, Targeson, Inc., San Diego, CA) were visualized using either subharmonic, ultraharmonic, or combined subharmonic and ultraharmonic modes. Flow channels, as small as 0.8 mm and 0.5 mm in diameter, were imaged using commercial peripheral and coronary imaging catheters at 12 MHz and 30 MHz transmit frequencies, respectively. Subharmonic and ultraharmonic imaging modes attained contrast-to-tissue ratios (CTRs) of 18 ± 2 dB and 20 ± 2 dB at 12 MHz transmit frequency, and 9 ± 2 dB and 13 ± 1 dB at 30 MHz transmit frequency, respectively. Combining subharmonic and ultraharmonic modes enhanced the CTRs to 33 ± 3 dB and 25 ± 2 dB at 12 MHz and 30 MHz transmit frequencies, respectively. These preliminary findings support the continued investigation of combined subharmonic and ultraharmonic IVUS for vasa vasorum imaging.

2:15–2:30 Break

2:30

4pBA6. A novel coding scheme and its application to ultrafast ultrasound imaging. Yang Zhang, Yuexin Guo, and Wei-Ning Lee (Elec. and Electron. Eng., The Univ. of Hong Kong, Rm. 807, Chow Yei Ching Bldg., Pokfulam Rd., Hong Kong, Hong Kong 0000, Hong Kong, zhangy@eee.hku.hk)

Ultrafast ultrasound imaging using plane or diverging waves, not focused beams, has enabled quantitative assessment of biological tissue elasticity, kinematics, and hemodynamics beyond anatomical information in the past decade. However, its sonographic signal-to-noise ratio (SNR) and penetration depth are limited by insufficient energy delivery under safety limits. We hereby propose a novel coding scheme and apply it to ultrafast ultrasound imaging to increase the SNR without compromising the spatial resolution and frame rate. In our coded ultrafast ultrasound imaging scheme, each transmit is a long pulse containing N ($N = 2^k$, k = 0, 1, 2,…) waves with short time intervals and polarity coefficients of +1 or −1, instead of the conventional short pulse with single wave. In reception, a linear decoding scheme comprised of addition, subtraction and delay operations is devised to recover N times higher intensity backscattered signals to gain SNR of 10·log10(N). Experimental results acquired by the Verasonics Vantage system from the calibration phantom and *in vivo* human back muscle show that the proposed method using two transmits of N = 32 waves achieves 13.0 ± 0.5 dB SNR improvement at 40 mm depth at 4000 frames/sec, leading to better image contrast and larger penetration depth.

2:45

4pBA7. Etiology of the color Doppler ultrasound twinkling artifact on *in situ* human kidney stones. Julianna C. Simon (Graduate Program in Acoust., Pennsylvania State Univ., University Park, PA and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu), Barbrina Dunmire, Bryan Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Jeffrey Thiel, Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and James R. Holm (Ctr. for Hyperbaric Medicine, Virginia Mason Medical Ctr., Seattle, WA)

Hyperbaric pressures of 0.3–10 MPa (absolute) have been shown to reduce the magnitude of the color Doppler ultrasound twinkling artifact on *ex vivo* human kidney stones, supporting the hypothesis that surface crevice microbubbles cause twinkling. For the first time, we investigate the etiology of twinkling on *in situ* human kidney stones. Eight human subjects with kidney stones known to twinkle were imaged with a Philips/ATL P4-2 transducer and Verasonics ultrasound system for 45 minutes while inside a hyperbaric chamber. Subjects breathed ambient air while the pressure in the chamber cycled to a maximum pressure of 0.4 MPa, with a scheduled decompression stop at 0.16 MPa where subjects breathed pure oxygen. Preliminary results show no change in the twinkling amplitude at 0.4 MPa compared to the baseline measurements before pressurization; however, a statistically significant increase in the twinkling amplitude was observed when subjects breathed pure oxygen at 0.16 MPa ($p = 0.046$). The increase in the twinkling amplitude upon changing respiratory gas composition further supports the crevice microbubble hypothesis of twinkling. Higher pressures than explored in this study may be needed to reduce twinkling on *in situ* human kidney stones. [Work supported by the National Space Biomedical Research Institute through NASA NCC 9-58 and NIH grant DK043881.]

3:00

4pBA8. Recovery of the complete data set from ultrasound sequences with arbitrary transmit delays. Nick Bottenus (Biomedical Eng., Duke Univ., 1427 FCIEMAS, Box 90281, Durham, NC 27708, nbb5@duke.edu)

Ultrasound beamforming relies on particular models of propagation to convert samples of the backscattered field through time into spatial samples of an image. The most common model used is straight-line propagation of a focused wave, assuming a narrow steered and focused beam that propagates radially along a selected direction. This model describes the main propagating pulse but fails to capture the true spatial extent of the wave. The reconstructed image suffers from defocusing, reduced signal-to-noise ratio (SNR), and contrast loss. A method is proposed to model transmission as the superposition of individual transmit elements on the transducer array and to efficiently recover the "complete data set," individual element transmit and receive responses, from an arbitrary scan sequence. Standard diverging wave focusing is applied to the complete data set to produce an image independent of the applied transmit focusing. For a sequence with a fixed transmit focus, the result is a high-SNR, two-way focused image. The proposed method also improves upon other synthetic transmit aperture methods such as plane wave imaging in that it captures the full diffracting wave field rather than relying on approximate field models, maximizing SNR at depth. Phantom and *in vivo* images are used to demonstrate improvement.

3:15

4pBA9. Study of effects of ultrasound stimulation on platelet-like-particle mediated fibrin-network deformation. Aditya A. Joshi (Mech. & Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Eng. Bldg. 3, Raleigh, NC 27695, aajoshi4@ncsu.edu), Nandi Seema, Ashley Brown (UNC-NCSU Joint Dept. of Biomedical Eng., North Carolina State Univ., Raleigh, NC), and Marie M. Muller (Mech. & Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

An important function of platelets is their ability to bind to fibrin. Over time, platelets contract the fibrin network to induce clot collapse through a process known as clot retraction. Synthetic platelet-like particles (PLPs) created from ultralow-crosslinked (ULC) microgels by conjugating ULCs to a fibrin-specific antibody are capable of mimicking this ability of natural platelets to induce fibrin network collapse. However, the rate of clot retraction is low compared to natural platelets. We demonstrate that deformability of a tissue-mimicking phantom containing ULCs increased in presence of ultrasound stimulation. We observe that the deformability of the ULC was optimal for 1MHz stimulation and a 0.025 mg/mL microgel concentration. PLP-laden, ULC-laden, and microgel-free fibrin clots were created and exposed to ultrasound stimulation for a period of 72 hours. The second set of clots was also created and monitored for 72 hours without ultrasound exposure. Clots were imaged via CryoSEM at 24 and 72 hours after polymerization to determine the effects of the PLPs and ultrasound stimulation on fibrin-network collapse. CryoSEM analysis of clots demonstrated increased density and decreased porosity in the fibrin network structure in presence of ultrasound, indicating microscopic clot collapse. These results suggest the potential of combining PLPs and ultrasound stimulation to alter fibrin clot properties and could be used in the future to enhance wound healing outcomes.

3:30

4pBA10. Validation of a wide-angle parabolic model for shallow-focus ultrasound transducers. Joshua Soneson and Yunbo Liu (Appl. Mech., FDA, 10903 New Hampshire Ave., Silver Spring, MD 20993, joshua.soneson@fda.hhs.gov)

Recently, a novel numerical method was developed for a wide-angle parabolic equation which accommodates steep gradients and discontinuities in the pressure distribution of the source boundary condition (Soneson, IEEE Trans Ultrason., Ferroelectr., Freq. Control **64**, pp. 679–687, 2017). The method allows rapid computation of acoustic fields with improved diffraction modeling capability over the standard parabolic approximation of the Helmholtz equation with no additional computational overhead, and is free from oscillatory artifacts which previously precluded the use of wide-angle models with discontinuous sources. In this work the wide-angle model, using the expression of a converging spherical wave (with a sharp cutoff at the edge of the active area) as the source boundary condition, is validated against measurements obtained using a 200 μm membrane hydrophone. The ultrasound field was produced by a single-element transducer with f-number 0.7 at 1.5 MHz, which was selected to test the wide angle model's account of diffraction physics well outside the regime of the standard parabolic model. Close agreement between the measured and computed fields is shown along the central axis and in the focal plane. This analysis represents a critical component of the rigorous validation of the model, which will come with a forthcoming uncertainty analysis.

3:45

4pBA11. Mapping the electrical activity of the heart *in vivo* using ultrafast acoustoelectric imaging. Beatrice Berthon, Philippe Mateo, Nathalie Ialy-Radio, Mickael Tanter, Mathieu Pernot, and Jean Provost (Institut Langevin, ESPCI Paris, PSL Res. Univ., CNRS 7587, INSERM U979, 17 rue moreau, Paris 75012, France, jean.provost@espci.fr)

The heart is driven by an electrical activation that propagates in cells and triggers their concerted contraction. When this electrical activation pattern is altered, it can lead to debilitating diseases such as arrhythmia or heart failure. Yet, directly imaging the electrical activity of the heart remains difficult to achieve. We have recently introduced Ultrafast Acoustoelectric Imaging (UAI) that combines ultrasound plane wave emissions and the acoustoelectric effect, i.e., the modulation of electrical impedance by ultrasound waves, to map electrical current densities in real-time with 1-mm and 5-ms spatial and temporal resolutions, respectively. Herein, we present its application to direct mapping of the cardiac activation in isolated live rat hearts and in pig hearts *in vivo*. UAI was performed in isolated rat hearts (n = 4) and in open-chest pigs (n = 1) using a 5-MHz linear array ultrasound probe fitted to a 256 channel- Vantage ultrasound system (Verasonics, Inc., WA). Two electrodes positioned on the heart and fed to the Vantage system were used to detect electrical impedance variations caused by ultrasound plane waves, which were then backprojected to reconstruct images of electrical current densities. UAI images depicted the propagation of the electrical activation of the cardiac tissue as validated by the electrocardiogram. These first *in-vivo* results suggest that UAI may allow the emergence of new biomarkers for the diagnosis and pre-clinical study of cardiac activation diseases such as arrhythmias.

Session 4pED

Education in Acoustics, Physical Acoustics, Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics: Synthetic Aperture Sonar for Youngsters

Murray S. Korman, Cochair
Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402

Gregory E. Coxson, Cochair
U.S. Naval Academy, Electrical and Computer Engineering Dept., Mail Stop 14 B, 105 Maryland Street, Annapolis, MD 21402

Chair's Introduction—4:00

Invited Paper

4:05

4pED1. Classroom demonstration of synthetic aperture and array concepts. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu)

Although often time consuming to develop and present, classroom demonstrations are an effective tool used in physics education. This talk walks through a demonstration of the basic concepts of array processing and synthetic aperture sonar. An emphasis has been placed on limiting equipment and setup time required to implement this demonstration, while allowing the presenter break the discussion into several conceptual topic areas. Primary topics include pulse compression and temporal/range resolution, array aperture footprint and azimuthal/range resolution, influence of transmitter and receiver beampatterns, and the importance of temporal coherence and measurement platform motion. Hardware require for this demonstration can be made as simple as a speaker and a single electret microphone used with a common PC soundcard.

Contributed Papers

4:25

4pED2. The Coffee-Can radar at the naval academy. Gregory E. Coxson and Erich Keyes (Elec. and Comput. Eng. Dept., U.S. Naval Acad., Elec. and Comput. Eng. Dept. U.S. Naval Acad., M.S. 14 B, 105 Maryland St., Annapolis, MD 21402, coxson@usna.edu)

The Coffee-Can radar was designed by Professor Gregory Charvat as a simple low-cost radar system for small teams of students to build and test during an intersession course at MIT. The name derives from the use of coffee cans as transmit and receive antennas. Since the success of Charvat's course, the coffee-can radar has been used at a number of schools for hands-on experiences with radar. It is one of the small radar systems used in courses in Principles of Radar at the United States Naval Academy, where laboratory and project-based learning are necessary components of the educational experience. This paper will describe how the radar operates, and will explain the role this radar plays in the Naval Academy's Principles of Radar and EW course. Attendees will receive a demonstration of the radar's three modes: ranging, range-rate estimation, and synthetic aperture imaging; and will have the opportunity to operate the radar. Additional small radar systems used in the Principles of Radar course will be available as well.

4:45

4pED3. Demonstration of synthetic aperture sonar or radar using shallow water waves in a ripple tank with small cylindrical targets. Kathryn P. Kirkwood and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m193342@usna.edu)

In an ideal stripmap synthetic aperture radar SAR or synthetic aperture sonar SAS experiment, a collocated transmitter and receiver array, respectively, generate a linear frequency modulated LFM pulse chirp that (1) reflects off of point-like targets that are placed along a smooth surface in the x,y plane and (2) receives backscattered echoes off the targets. The array (located a height h from the surface) may be towed (SAS) or flown (SAR), or in some airborne acoustic applications placed on a vehicle or track. In the stop and hop approximation waves are transmitted and then echoes vs. times are received and recorded at a point location along the track. At discrete locations along the track a set of N echoes are recorded which are pulse compressed using a correlation process. Using a time correlation backprojection algorithm the echoes are used to predict an image of the targets, namely the two-dimension reflectance f(x,y). Using a ripple tank, point-like spherical pulses are generated at locations guided by a linear track. The echoes from several slender cylindrical rods are received by a linear capacitance-to-voltage converter needle like probe [McGoldrick, Rev. Sci. Inst. **42**, 359–361(1971)] and backprojection is performed to predict the target locations.

Session 4pNS

Noise, ASA Committee on Standards, and Structural Acoustics and Vibration: Wind Turbine Noise

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

Kenneth Kaliski, Cochair
RSG Inc., 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
Hellweg Acoustics, 13 Pine Tree Road, Wellesley, MA 02482

Chair's Introduction—1:00

Invited Papers

1:05

4pNS1. Measurements of underwater sound radiated from an offshore wind turbine. James H Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett Bay Campus URI, Narragansett, RI 02882, miller@uri.edu), Ying-Tsong Lin, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Kathleen J. Vigness-Raposa, Jennifer Giard (Marine Acoust., Inc., MIddletown, RI), and Tim Mason (Subacoustech Environ. Ltd., Southampton, United Kingdom)

Underwater acoustic and geophysical systems were deployed to monitor the operation of the Block Island Wind Farm (BIWF). The BIWF consists of five GE Haliade 150-6MW wind turbines each with 150 m diameter blades. The five wind turbines were laid out about 1 km apart in a southwest-to-northeast arc. Each turbine is equipped with a direct drive permanent magnet generator, with no gearbox coupled to the generator. These turbines are variable speed and have independent pitch control by blade. The equipment used to monitor the BIWF operation consisted of a towed array of eight hydrophones, two VLAs with four hydrophones each and a fixed sensor package for measuring particle velocity. This sensor package consists of a three-axis geophone on the seabed and a tetrahedral array of four hydrophones at 1 m from the bottom. Additionally, an acoustic vector sensor was deployed in mid-water. During operations in December 2016, an acoustic signal was detected by the tetrahedral array of hydrophones at a position 50 meters west of the southwestern-most turbine. The frequency of this signal was approximately 72 Hz and the rms sound pressure level was about 100 dB re 1 micropascal. [Work supported by Bureau of Ocean Energy Management (BOEM).]

1:25

4pNS2. Sound level impact studies for wind energy in NY State: A whole new world. Robert O'Neal (Epsilon Assoc., Inc., 3 Mill & Main Pl Ste 250, Maynard, MA 01754-256, roneal@epsilonassociates.com)

In 2012, New York State adopted regulations for any energy facility over 25 megawatts in electrical capacity ("Article 10"). Since that time at least 15 wind energy projects have entered permitting, with many of them deep into the required sound level studies prescribed by Article 10. The author is working on seven of these projects and will cover the key steps in the process ranging from the initial public involvement through submittal of an application. The sound studies for Article 10 have become one of the most thorough and complex in the industry. Key elements of the technical studies include multi-season existing condition measurement programs, sound level modeling, meteorological data analysis, literature reviews, regulatory comparisons, and evaluation of a multitude of additional sound-related criteria such as hearing impairment and speech interference. The depth of the sound studies appear to be expanding beyond that contemplated in the Article 10 regulations.

1:45

4pNS3. Public acceptance of wind energy: Impact of sound levels. Thomas R. Haac (RSG, Inc., 55 RailRd. Row, White River Junction, VT 05001, ryan.haac@rsginc.com), Matt Landis (RSG, Inc., Burlington, VT), Kenneth Kaliski (RSG, Inc., White River Junction, VT), Ben Hoen, Joesph Rand (Lawrence Berkley National Labs, San Francisco, CA), Jeremy Firestone (Univ. of Delaware, Newark, DE), Johannes Pohl, Gundula Huebner (Institut für Psychologie der Martin-Luther-Universität Halle-Wittenberg AG Gesundheits- und Umweltpsychologie, Halle (Saale), Germany), and Debi Elliot (Portland State Univ., Portland, OR)

Lawrence Berkeley National Laboratory led a survey of 1,729 individuals located within 8 km of utility scale wind turbines in the United States. The survey included respondents around both large and small wind projects throughout the country. The survey focused on social acceptance, procedural and distributional justice, landscape and sound perceptions and annoyance, and compensation. A total of 15 of the wind projects were modeled to estimate the sound levels at each respondent's home. Modeled metrics included background

sound levels, maximum one-hour sound levels, percentage of time the respondent is downwind of a turbine, and a long-term sound level estimate using the local wind project capacity factor. Statistical analyses were conducted to estimate the acoustical drivers (sound level and sound level difference above background) toward the propensity for annoyance, and how these were affected by non-acoustic factors (e.g., compensation, prior attitude toward the project, visibility, etc.).

2:05

4pNS4. Human postural sway results in response to audible and infrasound emissions from wind turbines. Peggy B. Nelson, Andrew Byrne, Matthew Waggenspack (Dept. of Speech-Language-Hearing Sci., Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu), Michael Sullivan (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Christopher Feist, and William Herb (St. Anthony Falls Lab., Univ. of Minnesota, Minneapolis, MN)

Fifty healthy adult subjects ages 21–73 years attended to audible and infrasound signals generated from a wind turbine, recorded at 600 m and re-created in the laboratory. Stimuli consisted of modulated and unmodulated audible sound at 50 dB SPL, as well as recorded and peak-enhanced infrasound at an overall level of approximately 85 dB SPL (peaks up to 95 dB SPL). Participants were tested for their postural stability, detection, and ratings of audible and infrasound emissions from turbines. They also completed pre- and post-testing surveys for symptoms of imbalance. All subjects were blind to the knowledge that the stimuli were recorded from wind turbines. No significant adverse effects from healthy adults have been noted to date. Some individuals reliably indicated detection of infrasound signals. A few participants indicated mild symptoms (a rating of 1 on a scale from 0 to 4) following the test. Testing of patients who report vertigo will begin soon. Detailed results and implications will be discussed.

2:25

4pNS5. Subjective assessment of wind turbine noise—The stereo approach. Steven E. Cooper and Chris Chan (The Acoust. Group, 22 Fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

The conduct of stereo measurements for both playback in high-quality headphones and in a hemi-anechoic room has been undertaken for a number of wind farms and other low-frequency noise sources as an expansion on the material previously presented at the Boston ASA meeting. The results of the additional monitoring, evaluation of modulation index, and subjective analysis of this procedure is discussed and identifies the benefits of monitoring noise complaints and assessments of wind farm noise in stereo.

2:45

4pNS6. Measurements of infrasound blade pass frequencies in the far field. Andy Metelka (Sound and Vib. Solutions Canada, Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Infrasonic BPFs from wind turbines do appear as far as 120 km away from the nearest wind turbine at relatively low pressure levels. Previous papers clearly identify BPFs inside homes showing that they are rotational components of wind turbines. Both seismic ground vibration simultaneous to infrasound pressure are measured in the extreme far field in order to understand transfer paths. Air borne measurements clearly dominate in the far field also indicating audible amplitude modulation has no contribution to BPFs at these distances.

3:05–3:20 Break

3:20

4pNS7. Infrasound blade pass frequency transmissibility measurements inside homes near wind turbines. Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Penetration of wind turbine blade pass frequencies are compared at different homes and various rooms show how levels change with wind direction and wind speed. Rooms facing turbines exhibited higher penetration compared to basement rooms, indicating air borne transfer paths vs. ground borne vibration. Construction characteristics of different homes are also compared using transmissibility calculations. Older homes exhibited higher transmissibility while newer airtight homes, under certain conditions, had lower transmissibility.

3:40

4pNS8. Acoustic compliance with permit conditions—What does it mean? Steven E. Cooper (The Acoust. Group, 22 fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au) and les huson (L Huson & Assoc., Woodend, VIC, Australia)

The use of a dBA L90/L95 limit for operational wind farms, that rely upon a regression analysis of wind and noise that is averaged over time, does not relate to complaints of disturbance. The relevance of the wind farm noise contribution versus the true L90 background level versus wind speed, the percentage of time above the nominal permit condition, and the impact of special audible characteristics is discussed.

Contributed Papers

4:00

4pNS9. Preliminary estimates of acoustic intensity vorticity associated with a turbine blade rate. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu), Peter H. Dahl (Mech. Eng. Dept. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

We report on measurements recently made at the Wild Horse wind farm, near Ellensburg, WA, as part of short experiment to evaluate a device sensitive to acoustic particle velocity. Measurements were made at range of approximately 60 m, directly in front of (facing) a turbine, at a height 1.5 m above ground using a sound level meter (SLM) and vector sensor positioned within a few meters of each other. The SLM recorded a steady A-weighted SPL of 58 dB and C-weighted SPL between 70 dB and 84 dB, noting some low-frequency variability due to wind noise. The vector sensor measurement consists of 4 coherently-recorded signals, one from an omnidirectional microphone and three from a tri-axial accelerometer embedded in a lightweight 10-cm diameter sphere, which helps immunize this acoustic particle motion measurement from wind noise. Here we focus on combining velocity and pressure measurements to form acoustic vector intensity. Real (active) and imaginary (reactive) components of this field display temporal properties corresponding to the position of the 3-turbine blades. Time-varying vorticity in the intensity vector demonstrate existence of a signature in wind-turbine noise related to the blade-passing rate (1 Hz) that is not registered with conventional sound level measurements.

4:15

4pNS10. Wind turbines—A Cape Bridgewater residents' experience. Melissa Ware (PO Box 5355, Geelong, Victoria 3215, Australia, wmlyss@yahoo.com.au) and Steven E. Cooper (The Acoust. Group, Lilyfield, NSW, Australia)

In 2008, 29 2 MW wind turbines were built on neighboring properties, 850 m from our solid limestone home. Having a bilateral senso-neuro hearing loss may result in a different experience. The experiences of living near a wind farm, the consequences, and involvement in the Cape Bridgewater Study are discussed.

4:30

4pNS11. Revisiting the South Australian Environment Protection Authority 2013 Waterloo study using the Schomer principle. Mary L. Morris (PO Box 188, Eudunda, SA 5374, Australia, morrisdg@outlook.com) and Steven E. Cooper (The Acoust. Group, Lilyfield, NSW, Australia)

In 2013, the South Australian Environment Protection Authority conducted a 10 week study in relation to noise complaints at the Waterloo wind farm in the Mid North of South Australia. The study involved the use of diaries by 28 households located around the wind farm and noise loggers at 6 dwellings. In analyzing the residents' diaries, the Environment Protection Authority focussed on "noise" events and disregarded comments or observations by the residents which related to other forms of disturbance such as sensation. In the light of the Schomer principle presented in the 2017 ASA meeting in Boston, the 2013 Waterloo study data has been revisited to include an examination of the Power Output and WAV files associated with periods where residents have reported high levels of disturbance.

4:45

4pNS12. Politics, regulatory, and acoustic failures. Natalie J. Squire (Acoustar, 18 Macs Rd., Buninyong, VIC 3357, Australia, wnsquire@bigpond.com.au)

There have been three senate inquiries in recent years, all of which have called for regulatory reform. Noise related nuisances and associated amenity detriments are very often attributed to the operation of nearby wind turbines. In many circumstances, disturbances are reported despite the reports of acoustic experts concluding that the wind farm is operating in compliance with the noise limits conditioned in relevant planning approvals. The current practise whereby wind farm operators directly engage acousticians requires review. This is particularly so in the state of Victoria where under-resourced municipal councils, usually without the technical capacity to understand complex acoustic issues, are now responsible for wind farm permit regulation and enforcement—and a state where the EPA plays no formal role in the assessment of noise or mitigation of complaints of noise pollution emitted from a wind farm. The societal benefits of independent acoustic reporting and the importance of peer review is discussed in relation to the Waubra and Cape Bridgewater wind farms.

5:00

4pNS13. Startle reflex and sensitisation—How are these biological phenomena relevant to wind turbine noise exposure? Sarah E. Laurie (Waubra Foundation, PO Box 7112, Banyule, Melbourne, VIC 3084, Australia, sarah@waubrafoundation.org.au), Steven E. Cooper (The Acoust. Group, Lilyfield, NSW, Australia), and Robert Thorne (Acoustar, Brisbane, QLD, Australia)

Recent laboratory and field research has identified strong amplitude modulation as a trigger for sleep disturbance via acute physiological stress events. Reported, observed, and objectively recorded sudden increases in heart rate as part of a "flight fight response" during both day and night time noise exposure suggests that direct stimulation of the sympathetic nervous system via the startle reflex response may be involved. Mammalian field research has demonstrated that repeated elicitation of the acoustic startle reflex leads to observed sensitisation. Sensitization is also observed in Individuals chronically exposed to amplitude modulated industrial noise from sources including wind turbines. Relevant existing scientific literature, and examples of these events will be discussed.

5:15

4pNS14. Noise prediction of axial fan duct using a lattice Boltzmann approach. Kentaro Hayashi (Res. & Innovation Ctr., Mitsubishi Heavy Industries,Ltd., 5-717-1, Fukahori-machi,Nagasaki, Nagasaki 850-0392, Japan, kentaro1_hayashi@mhi.co.jp)

Large axial fans in fire power plants are noise source of the plants, they are required to reduce the noise level. Fan noise propagates in a fan duct, and since it is emitted to the exterior, silencers are used as countermeasure in general. It is important for noise reduction to predict noise generation and noise propagation in fan ducts. The Lattice-Boltzmann Method (LBM) based approach was applied for the unsteady simulation of axial fan and its duct to predict fan noise generation and sound propagation simultaneously. The LBM has minute numerical viscosity, and is suitable for the analysis of the sound propagation and the flow field. In this research, aeroacoustic analysis by using applicable analysis model in the design phase has been conducted and validated compared to experimental results. The results shows good agreement with experiment within 3dB at overall value and predicted spectra are also compared with experimental results.

Session 4pPA

Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Acoustics of Detecting Gravitational Waves using LIGO

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

Kenneth E. Gilbert, Cochair
Physics/NCPA, University of Mississippi, P.O. box 35, 1703 Hunter Road, Thaxton, MS 38871

Chair's Introduction—2:30

Invited Papers

2:35

4pPA1. Gravitational wave sources and data analysis. Benjamin Owen for the LIGO Scientific Collaboration and the Virgo Collaboration (Phys. and Astronomy, Texas Tech Univ., Box 41051, Lubbock, TX 79409-1051, benjamin.j.owen@ttu.edu)

What we look for affects how we look for it. I summarize current and predicted sources of gravitational waves, how the physics affects the signals, and how the signal morphologies affect analysis techniques. Expecting long and short signals, well modeled signals and unknown signals, means that gravitational waves use a variety of techniques familiar from acoustics, from matched filtering to wavelets and cross-correlation of data streams. These tell us today about black holes, and one day will tell us about neutron stars, cosmic explosions, and perhaps even stranger things.

3:05

4pPA2. A technical overview of the LIGO detectors. Adam Mullavey (LIGO Livingston Observatory, 19100 LIGO Ln., Livingston, LA 70754, amullavey@ligo-la.caltech.edu)

The LIGO detectors are a technological marvel, the culmination of ~50 years of research and development, that have opened up a new window on the universe by making the first detections of gravitational waves. The LIGO detectors can measure strain caused by gravitational wave perturbations down to 10^{-23} (at their most sensitive frequency band). In this talk, I will give an overview of the techniques that have allowed us to reach such incredible precision, and where possible relate said techniques to the field of acoustics.

3:35

4pPA3. Gravitational-wave detectors: Upgrades and new facilities. Katherine Dooley (Phys. and Astronomy, Univ. of Mississippi, PO Box 1848; 108 Lewis Hall, University, MS 38677, kldooley@olemiss.edu)

The LIGO observatories have detected gravitational waves, but even so there's much that can be done to further improve their sensitivity. I will describe plans to upgrade the existing Advanced LIGO detectors as well as design studies for new detectors in brand new facilities.

4:05–4:35 Panel Discussion

Session 4pSC

Speech Communication: Speech Perception and Word Recognition (Poster Session)

Mark VanDam, Chair
Speech & Hearing Sciences, Washington State University, PO BOX 1495, Spokane, WA 99202

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

4pSC1. Vowel intelligibility in clear speech produced by electrolaryngeal speakers. Steven R. Cox, Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., Hy Weinberg Ctr. 136, Garden City, NY 11530, scox@adelphi.edu), and Philip C. Doyle (Health and Rehabilitation Sci., Western Univ., London, ON, Canada)

This study assessed vowel intelligibility when electrolaryngeal (EL) speakers were instructed to use clear speech (CS). Eighteen consonant-vowel-consonant words containing /i/, /ɪ/, /ɛ/, and /æ/ were spoken by 10 laryngectomees in habitual speech (HS) and CS conditions. A total of 4,320 words across both speech conditions were recorded and then transcribed by 12, naïve listeners. Results indicate that vowel intelligibility was 85.4% (*range* = 77.8% to 90.6%) when EL speakers used HS compared to 82.7% (*range* = 75.0% to 92.8%) in CS. This finding suggests that CS does not facilitate improved vowel intelligibility for EL speakers. Future research will seek explanations for the lack of a CS benefit by examining the acoustic changes that occur when EL speakers use CS.

4pSC2. Phonological and auditory context effects in the perception of synthetic liquid-plus-stop clusters. Terrance M. Nearey and Benjamin V. Tucker (Linguist, Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 0A2, Canada, tnearey@ualberta.ca)

In experiments with English synthetic stops in the /VCCV/ disyllables /arga, alga, arda, alda/, listeners give more /-da/ responses after /ar-/ syllables [see, e.g., A. Lotto and K. Kluender, Percept. Psychophys. **60**, 602–619 (1998)]. The importance of general auditory contrast compared to more specific phonetic mechanisms is a key issue in the literature. In three experiments that used factorially crossed F3 continua for both the liquid /r-l/ and for the stop /g-d/, we recorded responses to all four disyllables (4 alternative forced choice). Experiments 1 and 2 both showed that categorizing the liquid as /r/ biases stop responses toward /d/. However, Experiment 1, which involved a 4-step extreme-/r/ to extreme-/l/ continuum, showed effects consistent with auditory contrast of the F3 values across the stop. Experiment 2 focused on the ambiguous region of the /l-r/ continuum [sampled in 7 steps]. While a clear effect of the [-rd-] phonological bias remained, there was little reliable evidence of contrast effects in F3 across the stop gap. Here, we report analyses of a new larger (n > 60) Experiment 3 with more stimuli (70 compared to less than 50 each) that subsumes the ranges of the previous two experiments. We evaluate the hypothesis (among others) that the auditory-contrast-like effect is confined largely to stimuli with relatively low F3 at the offset of the /VC-/ syllable.

4pSC3. Individual differences in cue weights are correlated across contrasts. Meghan Clayards (McGill Univ., 1085 Ave. Dr. Penfield, Montreal, QC H3A 1A7, Canada, meghan.clayards@mcgill.ca)

When listeners make judgments about phonological contrasts, they integrate different acoustic dimensions putting more weight on some than others. Individuals differ in how they weight different cues. Schultz, Francis & Llanos (2012) found that weights for VOT and f0 as cues to initial stop voicing in English are weakly positively correlated across individuals. We ask whether this is specific to VOT and f0 or a more general property of cue weighting across individuals. Secondly, across contrasts do the same listeners have stronger cue weights? 43 listeners performed a 2AFC task for four sets of minimal pairs that each varied orthogonally in two dimensions. All heard *bet-bat* and *Luce-lose* (vowel spectral quality vs. duration) and *bog-dog* (burst spectrum vs. formant transitions). 24 participants also heard *sock-shock* (sibilant spectrum vs. formant transitions) and the other 19 heard *dear-tear* (VOT vs. f0). Cue weights were fit with random slopes in a logistic regression for each minimal pair. Weights were positively correlated across individuals both within and across contrasts for *bet-bat, Luce-lose*, and *sock-shock*. *Bog-dog* and *dear-tear* had less consistent results. Overall this indicates that some individuals are better able to extract and use acoustic-phonetic information across different acoustic dimensions.

4pSC4. Intelligibility of sinewave consonants. James Hillenbrand and Michael J. Clark (Western Michigan Univ., 1903 W Michigan Ave., Kalamazoo, MI 49008, james.hillenbrand@wmich.edu)

A good deal of experimental work has assessed the intelligibility of sinewave speech (SWS), synthesized by mixing sinusoids that follow the formants of natural utterances. While SWS is clearly intelligible at some level, most SWS work has been conducted using sentences, whose intelligibility is affected by many factors in addition to those related to recognition at the phonetic level. Earlier work [Hillenbrand *et al.*, *J. Acoust. Soc. Am*, **129**, 3991–4000] measuring the intelligibility of SWS vowels in isolated syllables reported an identification rate of 55%, far above chance but ~40 percentage points lower that of the original signals. The present work tested the intelligibility of SWS versions of 23 consonant types in CV and VCV syllables with three vowel types ([ɑ i u]) spoken by one man and one woman. SW signals were generated from unedited envelope peaks rather than formants. Intelligibility averaged across 59 listeners was 59%, with large variability across both listeners (sd = 9.9) and, especially, consonant type (sd = 23.4). Recognition at the feature level was 65.5% for place, 78.4% for

manner, 90.0% for voicing. Voicing results are especially striking given that SWS is fully aperiodic, and the fact that the representation of spectral shape in SWS is necessarily coarse.

4pSC5. Computational modeling of human isolated auditory word recognition using DIANA. Filip Nenadic (Linguist, Univ. of Alberta, Edmonton, AB, Canada), Louis ten Bosch (Radboud Univ., Nijmegen, Netherlands), and Benjamin V. Tucker (Linguist, Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

In recent years, computational modeling has proved to be an essential tool for investigating cognitive processes underlying speech perception (see, e.g., Scharenborg & Boves, 2010). Here we address the question of how an end-to-end computational model that uses the acoustic signal as input simulates behavioral responses of actual participants. We used the Massive Auditory Lexical Decision (MALD) database recordings comprising of 26,800 isolated words produced by a single male native speaker of English. MALD response data came from 232 native speakers of English, with each participant responding to a subset of recorded words in an auditory lexical decision experiment (Tucker *et al*., submitted). We applied DIANA, a recently developed end-to-end computational model of word perception (Ten Bosch *et al*., 2013; Ten Bosch *et al*., 2015) to model the MALD response latency data. DIANA is a model that takes in the acoustic signal as input, activates internal word representations without assuming prelexical categorical decision, and outputs estimated response latencies and lexicality judgements. We report the results of the participant-to-model comparison, and discuss the simulated between-word competition as a function of time in the DIANA model.

4pSC6. Asymmetric discrimination of phonetically incongruent audio-visual vowels. Matthew Masapollo (Dept. Cognit., Linguistic & Psychol. Sci., Brown Univ., 190 Thayer St., Providence, RI 02912, matthew_masapollo@brown.edu), Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada), James Morgan, Lauren Franklin (Dept. Cognit., Linguistic & Psychol. Sci., Brown Univ., Providence, RI), and Lucie Ménard (Dept. of Linguist, Univ. of Montreal at PQ, Montreal, QC, Canada)

Masapollo, Polka, and Ménard (2017) recently reported a robust directional asymmetry in visual vowel perception: perceivers discriminated a change from an English /u/ viseme to a French /u/ viseme significantly better than a change in the reverse direction. This asymmetry parallels a frequent pattern in auditory vowel perception that points to a universal bias favoring "focal" vowels, whose distal articulatory gestures result in the convergence of adjacent formant frequencies. In the present study, we investigated how the integration of acoustic and visual speech cues influences these directional effects in bimodal vowel perception. In AX discrimination tests in which acoustic and visual cues were phonetically-incongruent, subjects showed an asymmetry, comparable to that found with unimodal stimuli, when there was an acoustic change in focalization, with visual focalization held constant. These findings indicate that acoustic cues are capable of inducing an asymmetry even when there is no cross-modal correspondence between the acoustic and visual channels. We are currently examining whether a similar asymmetry also emerges when there is a visual change in focalization, with acoustic focalization held constant. By comparing different types of audio-visual incongruence, we will be able to examine whether one sensory channel has more perceptual potency than the other.

4pSC7. Individual differences in perceptual adaptation to phonetic categories: Categorization gradiency and cognitive abilities. Donghyun Kim, Meghan Clayards (Linguist, McGill Univ., 1085 Dr. Penfield, Rm. 111, Montreal, QC H3A 1A7, Canada, heydonghyun@gmail.com), and Eun Jong Kong (Dept. of English, Korea Aerosp. Univ., Goyang-si, Gyeonggi-do, South Korea)

We examine whether listeners flexibly adapt to unfamiliar speech patterns such as those encountered in foreign-accented English vowels. In these cases, the relative informativity of acoustic dimensions (spectral quality vs. duration) can be changed such that the most informative dimension (spectral quality) is no longer informative, but the role of the secondary cue (duration) is enhanced. We further test whether listeners' adaptive strategies are related to individual differences in utilizations of secondary cues (measured by categorization gradiency) and cognitive abilities. Native English listeners (N = 36) listened to continuum of vowels /ɛ/ and /æ/ (as in *head* and *had*) varying spectral and duration values to complete a perceptual adaptation task, a visual analogue scaling (VAS) task, and were given cognitive ability tasks examining executive function capacities. Results showed that listeners mostly used spectral quality to signal vowel category at baseline, but rapidly adapted by up-weighting reliance on duration when spectral quality was no longer informative. The VAS task showed substantial individual differences in categorization gradiency with more gradient listeners using a secondary cue more, but gradiency was not linked to degree of adaptation. Finally, results of cognitive ability tasks revealed that individual differences in inhibitory control, but not the other cognitive abilities, correlated with the amount of perceptual adaptation.

4pSC8. Working memory capacity and lexical knowledge in perceptual restoration of interrupted speech. Naveen K. Nagaraj (Audiol. and Speech Pathol., Univ. of Arkansas Medical Sci., 2801 S. University, Ste. 600, Little Rock, AR 72204, nknagaraj@uams.edu) and Beula Magimairaj (Commun. Sci. and Disord., Univ. of Central Arkansas, Conway, AR)

Role of working memory capacity (WMC) and lexical knowledge in perceptual restoration (PR) of missing speech was investigated using the interrupted speech perception (ISP) paradigm. 75 young normal hearing listeners' speech identification was measured using low-context sentences interrupted by silence and three noise at 1.5 Hz. Noise conditions created by manipulating the spectro-temporal content of filler noise were as follows: (1) low frequency (LF) speech shaped noise (SSN), (2) temporal fine structure filled (TFSf) noise consisting of LF TFS from the missing speech, and (3) temporal envelope filled (TEf) noise consisting of LF TE extracted from the missing speech. WMC was measured using verbal reading span and visuospatial symmetry span. Lexical knowledge was assessed using standard vocabulary and meaning from context tests. We hypothesized that during noise filled ISP conditions WM mechanism is crucial for retrieving and integrating relevant information from long-term memory. Results showed that ISP was better for SSNf than other conditions tested. Both lexical knowledge and verbal WMC explained unique variance in SSNf, but were unrelated to silent gated condition. It was only, lexical knowledge that uniquely predicted PR for TFSf and TEf conditions. Findings in general suggest that PR of filled interrupted speech depends crucially on individuals' lexical knowledge.

4pSC9. Delayed effects of speech and non-speech stimuli on sibilant categorization. Eleanor Chodroff (Linguist, Northwestern Univ., 2016 Sheridan Rd., Chicago, IL 60208, eleanor.chodroff@northwestern.edu) and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD)

Adaptation to the speech of a novel talker can involve at least two types of mechanism: phonetic learning (e.g., Samuel & Kraljic, 2009) and spectral contrast (e.g., Lotto & Kluender, 1998). While phonetic learning persists over long time periods, auditory effects such as spectral contrast have been demonstrated to occur for sounds that are temporally adjacent or separated by brief delays (e.g., 1.3s; Holt, 2005). The present study examined whether phonetic and auditory mechanisms can be distinguished by introducing a substantially longer delay between exposure and test. Five exposure conditions were examined across participants: exposure to syllables beginning with a relatively high or low COG [z], exposure to white noise matched in LTAS to the high or low COG [z]-initial syllables, and a baseline (no exposure) condition. Following exposure, participants performed a one-back image repetition detection task for 15 minutes, and then categorized members of the same 10-point [ʃ]-[s] continuum over 6 blocks. In comparison to baseline, exposure to speech had a significant and expected influence on sibilant categorization (e.g., the [ʃ]-[s] boundary shifted toward [s] following high COG [z] exposure). Speech and non-speech exposures were compared to determine whether spectral contrast is a viable explanation for long-lasting sibilant adaptation.

4pSC10. Effect of musical experience on learning to understand vocoded speech. Kieran E. Laursen (Dept. of Psychol., Lawrence Univ., 711 E. Boldt Way, Appleton, WI 54911, kieran.e.laursen@lawrence.edu), Iain C. Williams (Psychology, Univ. of North Carolina, Wilmington, NC), Tahnee Marquardt (Oxford Mindfulness Ctr., Univ. of Oxford, Oxford, United Kingdom), Sara L. Prostko, and Terry L. Gottfried (Psychology, Lawrence Univ., Appleton, WI)

This study explores whether musicians have an advantage over non-musicians in processing and comprehending 8-channel vocoded speech, spectrally degraded speech that imitates cochlear implant output (see Loebach, Bent, & Pisoni, 2008). Musicians and non-musicians completed a pre-test in which they were asked to transcribe a number of vocoded sentences and words. Afterwards, they completed training on either vocoded or natural stimuli. After training, participants completed post-tests during which they transcribed vocoded speech, including items from the original test and from new speakers and 4-channel vocoded stimuli. In preliminary studies, musicians showed no advantage over non-musicians, and contrary to expectations, participants in natural-speech training condition improved more from pre- to post-test vocoded word recognition than participants in the vocoded training condition. Current studies are examining the relation of rhythmic and melodic perception (Musical Ear Test, Wallentin *et al*., 2010) to performance on the vocoded speech learning; the vocoded and natural speech training tasks are also compared to a control training task (testing sine-wave tone detection). Results of these tests will be used to evaluate the extent to which musical ability and training may be related to perception of degraded speech.

4pSC11. The role of familiarity in audiovisual speech perception. Chao-Yang Lee, Margaret Harrison (Commun. Sci. and Disord., Ohio Univ., Grover W225, Athens, OH 45701, leec1@ohio.edu), and Seth Wiener (Modern Lang., Carnegie Mellon Univ., Pittsburgh, PA)

Speech perception involves processing a spoken message's linguistic content and information about the talker's voice carrying the message. Speech perception is also affected by visual information, e.g., the McGurk effect. Familiarity with a talker's voice facilitates auditory speech perception (Nygaard & Pisoni, 1998), but it is not clear whether visual familiarity with a talker's face affects audiovisual speech perception (Walker, Bruce, & O'Malley, 1995). In this study we investigated how visual familiarity with a talker affects the perception of the English sibilants /s/ and /ʃ/, which involve visible lip-rounding contrasts. Participants identified syllable-initial sibilants from stimuli that were audio-only, visual-only, audiovisual-congruent, or audiovisual-incongruent (e.g., audio "save" paired with visual "shave"). We also examined whether visual familiarity affects the occurrence of the McGurk effect. Participants identified syllable-initial stops from syllables that were audiovisual-congruent or incongruent (e.g., audio /bɑ/ paired with visual /gɑ/). The results indicated that participants familiar with the talker identified sibilants faster in all conditions and more accurately in the visual-only condition; no accuracy difference was found between the two groups in the other conditions or in the number of McGurk responses. These results are discussed in the context of processing intersensory information.

4pSC12. Examining the effect of first formant transition on perception of dynamic spectral change in the second formant. Amie Roten and Michelle R. Molis (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland HCS, 3710 SW US Veterans Hospital Rd., P5-NCRAR, Portland, OR 97239, amie.roten@va.gov)

Speech perception studies using methods such as formant-flattening and the silent-center paradigm have demonstrated that vowel-inherent spectral change (VISC) in speech-like stimuli adds significant information aiding in accurate phoneme identification. For listeners to reliably utilize this cue, formant extent thresholds must be smaller than transitions occurring in natural syllables. Although natural speech always involves movement of multiple formants over time, limited studies have determined formant transition thresholds in the context of competing formant movement. The purpose of this study was to begin to examine these interactions by determining the effect of F1-region transitions on perception of change in F2 (ΔF2). Listeners were presented 120 ms, two-formant stimuli approximating a vocoid F1/F2 configuration using a 4-interval, 2AFC, two-down/one-up adaptive paradigm. The first formant pivoted around a 500 Hz center frequency, and the second around 1500Hz. Thresholds for detection of both upward and downward ΔF2 were determined in the context of a flat F1, and an F1 transitioning linearly up or down by 1 ERB. Differences in performance were observed depending on F1 context, as well as direction of ΔF2, which may have bearing on speech perception ability. [Work supported by NIH/NIDCD.]

4pSC13. Study on interactions between voicing production and perception using auditory feedback paradigm. Shunsuke Tamura, Miduki Mori, Kazuhito Ito, Nobuyuki Hirose, and Shuji Mori (Kyushu Univ., 744 Motooka, Nishi-ku, Fukuoka, Japan, Rm. 827, 8th Fl., West Zone II Bldg., Kyushu University Ito Campus, Fukuoka 819-0395, Japan, tamuras@cog.inf.kyushu-u.ac.jp)

A previous study reported that perturbed auditory feedback affected voicing production [Mitsuya, MacDonald, and Munhall (2014). J. Acoust. Soc. Am., **135**, 2986–2994]. In this study, we investigated whether perturbed auditory feedback would also affect voicing perception. Eighteen native Japanese speakers participated in the experiment. Half of the participants performed an auditory feedback task in which a syllable sound /da/ was presented simultaneously with the participant's utterance of /ta/. The other participants did a passive listening task in which participants heard a syllable sound /da/ without the utterance. Before and after each task, participants performed a speech production task of /da/-/ta/ and a speech identification task of /da/-/ta/ continuum stimuli varying in voice-onset time (VOT). Results showed that perturbed auditory feedback lengthened the VOT of /ta/ production, whereas passive listening did not affect voicing production. Regarding voicing perception, passive listening shortened the VOT boundary of /da/-/ta/, which may be due to selective adaptation. On the other hand, perturbed auditory feedback did not vary the boundary. One interpretation of these results is that the effects of voicing production modulation on voicing perception can be cancelled out by selective adaptation, which may have occurred by listening to a syllable sound /da/ during auditory feedback task.

4pSC14. Segmenting words from bilingual speech: Evidence from 8- and 10-month-olds. Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montréal, QC H3A 1G1, Canada, adriel.orena@mail.mcgill.ca) and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Previous studies show that young monolingual infants use language-specific cues to segment words from multiword utterances in their native language. However, little is known about how infants deal with the segmentation challenge in bilingual environments. Here, we examined the word segmentation abilities of young infants in a mixed dual-language task. Infants were familiarized with an English-French passage containing a target word in each language, and were then tested on their recognition of those target words. Results confirm that 8-month-old monolingual infants show language-specific patterns in word segmentation: English- and French-monolingual infants segmented in their native language, but not in the other unfamiliar language. As a group, 8- and 10-month-old bilingual infants showed positive evidence of segmentation in both of their native languages. However, closer inspection of the data suggests that bilingual infants are only able to segment in the language in which they receive more exposure. Taken together, these results suggest a dose-response relationship between speech input and word segmentation: that is, more input in a language gives infants more opportunities to learn about how word boundaries are denoted in that language. This study also addresses the possible roles of attention and code-mixing on the development of word segmentation abilities.

4pSC15. Effect of speaker's age on perceivers' ability to predict rounding from AV speech. Melissa Redford (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu) and Jeffrey Kallay (Univ. of Oregon, Eugene, Ohio)

We have developed a noninvasive method that synthesizes the complex activity of speaking into a single measure to investigate the scope of anticipatory behavior as this relates to units of execution (Redford, Bogdanov, Vatikiotis-Bateson, 2016). The method leverages work in audio-visual speech perception to identify degree of anticipatory coarticulation at specific temporal locations, manipulated using a gating paradigm. The aim here was to test whether the method was robust to group differences in speakers' age. Twenty-four college-aged adults judged the presence or absence of rounding in minimal pair sentences produced by 3 adults and 3 five-year-old children based on their AV-recorded speech, gated at different distances from the target rounded/unrounded vowel. Perceivers' ability to correctly detect an upcoming rounded vowel varied as a function of the prevocalic consonant's coarticulatory resistance and as a function of distance from target. There was no main effect of age group on perceivers' accuracy. Age group did however interact with the other factors in that rounding was detected earlier in children's speech compared to adults' speech for one of three prevocalic consonants. The results confirm the promise of our method for studying developmental changes in linguistic chunks planned for execution. [Work supported by NICHD # R01HD087452.]

4pSC16. Do older and younger adult listeners show a "clear speech benefit" for speech produced by older adult talkers? Valerie Hazan and Outi Tuomainen (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

This study investigates whether a "clear speech benefit" is obtained for speech produced by older (OA) talkers and younger adult (YA) controls in a clear speaking style when heard in babble noise. The speech materials were recorded while OA and YA talkers read BKB sentences to a YA partner who repeated the sentence while hearing normally (NORM) or with a simulated hearing loss (HLS). The HLS condition naturally induced clear speech adaptations. 128 BKB sentences from 4 YA and 4 OA talkers (NORM, HLS) matched on a range of metrics were used in an adaptive listening test tracking the signal-to-noise ratio corresponding to 67% intelligibility. Listeners were 71 native British English listeners: 24 YA (M = 25.2 yrs), 27 OA-NH with normal hearing (M= 71.8), 20 OA-HL with presbyacusis (M= 73.7). Speech perception in noise was hardest for OA listeners, especially from OA-HL. SNR thresholds were significantly lower for YA than for OA voices. The clear speech benefit for HLS speech was only significant for YA voices for all listener groups. In summary, OA talkers were less intelligible than YA talkers regardless of listener age; clear speech adaptations by OA talkers did not result in enhanced intelligibility for OA or YA listeners.

4pSC17. Inhibitory and lexical frequency effects in younger and older adults' spoken word recognition. Sarah Colby (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., Montreal, QC H3A 1G1, Canada, sarah.colby@mail.mcgill.ca), Victoria Poulton, and Meghan Clayards (Linguist, McGill Univ., Montreal, QC, Canada)

Older adults are known to have more difficulty recognizing words with dense phonological neighbourhoods (Sommers & Danielson, 1999), suggesting an increased role of inhibition in older adults' spoken word recognition. Revill and Spieler (2012) found that older adults are particularly susceptible to frequency effects, and will look more to high frequency items compared to younger adults. We aim to replicate and extend the findings of Revill and Spieler (2012) by investigating the role of inhibition along with frequency for resolving lexical competition in both older and younger adults. Older (n = 16) and younger (n = 18) adults completed a visual word paradigm eyetracking task that used high and low frequency targets paired with competitors of opposing frequency, and a Simon task as a measure of inhibition. We find that older adults with poorer inhibition are more distracted by competitors than those with better inhibition and younger adults. This effect is larger for high frequency competitors compared to low. These results have implications for the changing role of inhibition in resolving lexical competition across the adult lifespan and support the idea that decreased inhibition in older adults contributes to increased lexical competition and stronger frequency effects in word recognition.

4pSC18. Speech recognition and word learning in 24-month-olds: The roles of non-native speech and familiar words. Cynthia P. Blanco and Sandra R. Waxman (Psychology, Northwestern Univ., 2029 Sheridan Rd., Evanston, IL 60208, cynthiapblanco@gmail.com)

After 18 months of age, infants' lexical representations are sufficiently flexible to recognize acoustically unfamiliar productions as variants of familiar words (Best *et al*., 2009; Mulak *et al*., 2013). For novel words, 24-month-olds still have trouble generalizing from native to non-native pronunciations, although exposure to the accent improves recognition (Schmale *et al*., 2011, 2012; White & Aslin, 2011). In the present study, we tested how quickly 24-month-olds could use exposure and familiar words to learn the meanings of novel words, on-line, when listening to Spanish-accented speech. In the exposure phase, twenty-four-month-olds heard a novel word embedded in a dialogue that either contained linguistic cues to the referent's animacy (*The vep is eating*) or was uninformative (*The vep is right here*). Infants then saw two pictures at test, one animate and one inanimate, and were asked to find the *vep*. Looking time to the two potential referents here was compared to performance with native-accented speech (Ferguson *et al*., under review). Infants hearing Spanish-accented speech struggled to learn the referents of novel words, but differences also emerged between the accent conditions for familiar target words. Infants are slower to access the meanings of familiar words when hearing non-native speech.

4pSC19. Does the developing lexicon constrain infants' discrimination of English vowels? Megha Sundara (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, megha.sundara@humnet.ucla.edu)

Infants initially discriminate native and non-native contrasts. With perceptual reorganization within the first year of life, discrimination of non-native contrasts reduces while the discrimination of native contrasts improves. Several researchers have proposed distributional learning as a domain-general mechanism by which infants' acquire phonetic categories (e.g., Saffran *et al*., 1999; Maye *et al*., 2002). Recent proposals, however, argue for an interactive mechanism where learning words concurrently, supplements distributional learning of phonetic categories (e.g., Swingley, 2009; Feldman *et al*., 2013). Not only is this interactive mechanism available during the first year of life, its computational implementations outperform ones based on distributional learning alone (Feldman *et al*., 2013a; Feldman *et al*., 2013b). We evaluated predictions of the interactive and distributional learning models against infant discrimination data. Data from 4-, and 8-month-olds, learning (a) only English (b) only Spanish or (c) Spanish and English, tested on English (i) /e- ɛ/ (previously published in Sundara & Scutellaro, 2011), (ii) /i- ɪ/ and (iii) /e- ɪ/, were compared. All stimuli were produced by multiple female talkers from the Hillenbrand corpus; infants were tested using a visual fixation procedure with a habituation criterion of 50%. Our results are consistent with infants using a distributional not interactive learning mechanism.

4pSC20. Representations of speech signals recorded through a dynamic periphery inspired by horseshoe bat biosonar. Alexander Hsu (T. J. Watson Res. Ctr., IBM, Yorktown Heights, NY), Anupam Kumar Gupta, Rolf Müller (Virginia Tech, Blacksburg, VA), Xiaodong Cui, Kartik Audhkhasi, and Jin-Ping Han (T. J. Watson Res. Ctr., IBM, 1101 Kitchawan Rd., Yorktown Heights, NY 10598, hanjp@us.ibm.com)

Horseshoe bats have to navigate through complex environments such as dense forests and structure-rich vegetation relying on input from their highly sophisticated biosonar systems. One of the key components of these bats' ability to obtain high-quality acoustic information is to alter the shape of their outer ears rapidly. In prior work, the authors have shown that by mimicking the horseshoe bat rapid ear movements, a bat-inspired robotic dynamic periphery for recording speech signals could enhance speech recognition for limited dataset and also provide estimates for the speaker's direction along with speech recognition. In our current study, we continued to investigate how speech datasets processed by the dynamic periphery may be enhanced compared to a reference by extracting acoustical features

through Mel frequency cepstral coefficient (MFCC) transform, Lyon's cochlear bandpass filters, and a neural spike representation, respectively. This study aims to characterize the detailed acoustical differences and quantify the improved speaker intelligence with noise robustness through the dynamic periphery. The ultimate goal of this research is to identify a signal representation that is well suited to capitalize on the time-variant properties of the biomimetic recording periphery and make the dynamic information-bearing features accessible for the classification stages.

4pSC21. Filtered and unfiltered sentences produce different spectral context effects in vowel categorization. Christian Stilp and Ashley Assgari (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Speech perception is heavily influenced by surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects (SCEs)* that bias categorization of later sounds. For example, when context sounds have a low F1 bias, listeners report more high F1 responses to the target vowel, and vice versa. SCEs have been demonstrated using a variety of approaches, but most often the context was a single sentence filtered two ways (e.g., low F1 bias, high F1 bias) to introduce spectral properties that biased speech categorization. This maximizes acoustic control over stimulus materials, but vastly understates the acoustic variability of speech. Here, vowel categorization was examined following context sentences that naturally possessed desired spectral properties without any filtering. Sentences with inherent low-F1 or high-F1 peaks in their long-term spectra were presented before a target vowel (/ɪ/-/ɛ/). Filtered sentences with equivalent spectral peaks were included as controls. Across several experiments, as spectral peak magnitudes in filtered and unfiltered contexts increased, SCE magnitudes increased linearly. However, unfiltered sentences produced smaller and more variable SCEs than filtered sentences. Results raise important questions about how closely laboratory studies of speech perception model processes in everyday listening.

4pSC22. Discrete ratings and dimensional judgments of emotional speech: A preliminary look at gender differences. Shae D. Morgan and Rebecca Labowe (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Two models of emotion identification and discrimination are not often compared and may show differing trends in how listeners rate emotional stimuli. The discrete or "basic" emotion model posits that emotions are categorical, and that listeners develop auditory prototypes of "basic" emotions (e.g., anger) based on their acoustic profile and past experiences. More complex emotions (e.g., frustration) fall under a "basic" category. The dimensional model examines emotions along continua of different emotional dimensions, such as activation/arousal and pleasantness/valence. The present study introduces the Morgan Emotional Speech Set and examines listener judgments of the stimuli in the corpus. The database consists of 2160 emotional speech sentences (90 sentences x 4 emotions x 6 talkers) produced by three male and three female actors for use in future studies. Each sentence was rated by 10 listeners (5 male, 5 female), who assigned an emotion category to each sentence and also rated each sentence by its activation (low to high) and pleasantness (very unpleasant to very pleasant). Discrete ratings will be compared with dimensional judgments made by listeners to examine intended and perceived emotional content in the database. A preliminary look at gender differences in the ratings and judgments will also be discussed, as male and female listeners may rate emotions differently for same- or different-sex talkers.

4pSC23. F0 as a cue to prosodic structure in voice onset time categorization. Jeremy Steffman (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jsteffman@g.ucla.edu)

This study investigates the effect of F0-based cues to English prosodic structure on the categorization of voice onset time (VOT) continua. Human perceptual systems are capable of extracting relevant phonetic and phonological information in spite of pervasive variation in the speech signal, with prosodic structure being one source of variation. For example, in English VOT in voiceless stops is robustly longer when intonational phrase (IP) initial, versus IP-medial. Kim & Cho (2013) found listeners compensate for perceived IP boundaries, requiring longer VOT to categorize sounds as voiceless when IP-initial. However, because the target sound was in an utterance-medial IP, compensation could have arisen due to the lengthened duration of the preceding IP-final syllable, as in speech rate normalization, instead of due to prosodic structure itself. Mitterer *et al*. (2016) found categorization shifts solely on the basis of duration, by rendering F0 contours prosodically ambiguous, yet it remains unknown what effect F0 has on categorization when it unambiguously cues prosodic structure. This study tests the hypothesis that compensation is triggered by more than duration normalization, by varying F0 cues independently from duration. Results will be discussed in terms of language-general auditory processing and top-down language-specific influences on segment categorization.

4pSC24. Perceived anger in clear and conversational speech: Effect of aging and hearing loss. Shae D. Morgan and Ashton Crain (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Previous research suggests that young normal-hearing and older hearing-impaired adult listeners judge clear speech as sounding angry more often than conversational speech. Interestingly, older hearing-impaired listeners were less likely than young normal-hearing listeners to judge sentences as angry in both speaking styles. It was unknown, however, whether this difference in ratings of emotion were driven by the age or hearing status differences between the two groups. A secondary investigation showed that young adult listeners with a simulated hearing loss that matched the older hearing-impaired group rated emotions nearly identically to the young normal-hearing group, suggesting no effect of hearing loss on ratings of emotion. The simulated hearing loss failed to account for other auditory factors or psychological processes associated with aging that may have account for the group differences. The present study carried out the same emotional rating task using clear and conversational speech sentences (as used in the previous studies) on a group of older adults with clinically normal hearing to determine whether differences in anger (and other emotion) perception are driven by age differences, hearing loss, or other factors.

Session 4pSPa

Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity I

Kay L. Gemba, Cochair
MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Room 446, La Jolla, CA 92037

Jit Sarkar, Cochair
Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Drive, Mail Code 0238, La Jolla, CA 92093-0238

Chair's Introduction—1:00

Invited Papers

1:05

4pSPa1. Exploiting natural surface noise and ships as sources of opportunity for passive bottom reflection-loss estimation. Lanfranco Muzi and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu)

The marine ambient-noise field, including natural and anthropogenic sources, can be exploited to estimate the bottom reflection coefficient and the associated loss, which is an important contributor to the total transmission loss in shallow water scenarios. The physical nature of the natural surface-noise field is such that the bottom reflection coefficient can be estimated passively by beamforming on a vertical line array, if there are no localized sources near the array and the noise level generated by waves and wind is adequate. Recently, an algorithm for processing the noise generated by a single ship moving over a wide range of steering angles has been shown to extend the technique to this source of opportunity, even when the natural noise level is too low to produce any measurable loss. In this study, simulation and data from recent experimental campaigns (from arrays moored to the bottom or mounted on an autonomous underwater vehicle) are used to illustrate the technique, and investigate in particular both the effect on the performance of low natural noise levels, and the possibility of using more than one ship as source of opportunity.

1:25

4pSPa2. Headwaves in ocean acoustic interferometry. Martin Siderius (ECE Dept., Portland State Univ., P.O. Box 751, Portland, OR 97207, siderius@pdx.edu), Jie Li (Marine Physical Lab., Univ. of California, Scripps Inst. of Oceanogr., San Diego, CA), and Peter Gerstoft (Marine Physical Lab., Univ. of California, Scripps Inst. of Oceanogr., La Jolla, CA)

In ocean acoustic interferometry, signals measured on two or more receivers are cross-correlated to produces an estimate of the Green's function between these receivers. A "virtual refracted" wave is an early arrival in the time-domain Green's function estimate. This virtual refracted wave is a phenomenon that has been widely reported on in the seismic interferometry literature. This early arrival of energy is also referred to as spurious energy or non-physical arrivals. Although this can be interpreted as a virtual wave it can be a result of the physically propagating headwave. The headwave travels in the seabed and re-radiates into the water column and therefore has important information content such as the seabed sound speed. In seismic interferometry, active sources and horizontal arrays are used but the virtual refracted headwave phenomenon is also observable on vertical arrays and with passive measurements of ocean noise. The signal processing used is a generalization of passive fathometer processing which applies beamforming (including adaptive methods) to the array data. Modeling and experimental data will be presented to show the headwave can be observed and used to estimate the seabed sound speed. The generalized passive fathometer signal processing is compared to the seismic interferometry processing.

1:45

4pSPa3. Extracting tomographic arrival time information from ships of opportunity in the Santa Barbara channel using blind deconvolution. Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, J. Erskine Love Bldg., Rm. 131, Atlanta, GA 30332-0405, ncd001@lvc.edu), Kay L. Gemba, Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

This paper summarizes the ongoing investigations surrounding the use of a ray-based blind deconvolution algorithm to recover arrival time information from sources of opportunity, such as shipping vessels, recorded on vertical line arrays (VLAs) in ocean waveguides. The deconvolution is primarily performed by using an estimate of the unknown source phase, obtained through wideband beamforming, to essentially match filter the VLA recordings and recover the channel impulse response (CIR). This paper will focus on results from an experiment performed in 2016 in the Santa Barbara shipping channel (water depth ~550 m). Four VLAs, with both short (~15 m) and long (~56 m) apertures, were deployed between the north and south bound shipping lanes and continuously collected acoustic data during one week. With the ultimate goal of passive acoustic tomography in mind, this paper aims to discuss (1) the robustness of the algorithm to extract differential arrival times along VLA elements using ships as sources of opportunity, (2) the achievable accuracy of blind arrival time measurements in comparison to the time-of-flight precision required for tomographic inversions, and (3) the ideal parameters (e.g., frequency bandwidth, snapshot duration, beamforming methodology...) for which to perform this ray-based blind deconvolution method in SBC-like ocean environments.

2:05

4pSPa4. Differential arrival time accuracy using ships of opportunity. Kay L. Gemba, Jit Sarkar, Bruce Cornuelle (MPL/SIO, UCSD, Univ. of California, San Diego, La Jolla, CA 92093-0238, gemba@ucsd.edu), Nicholas C. Durofchalk, Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), William S. Hodgkiss, and William A. Kuperman (MPL/SIO, UCSD, La Jolla, CA)

Previously, we demonstrated using the Noise-09 data set that it is possible to estimate the relative channel impulse responses (CIR) over a 1.5 km transect independently at 3 vertical line arrays (VLAs). This vessel provided sufficient SNR and bandwidth for the uncertainty of the direct arrival time to reach 5 microseconds. The time evolution of the CIRs as the ship passed by was subsequently used to determine VLA hydrophone positions relative to a reference hydrophone with accuracy on the order of centimeters. An experiment was performed in the Santa Barbara Channel using four VLAs placed between the sea lanes of in- and outgoing shipping traffic. Ship tracks were obtained from the Automatic Identification System (AIS). We extend the previous processing to this data set and discuss the uncertainties of relative multi-path arrival-times and differential travel times between VLAs.

2:25–2:40 Break

2:40

4pSPa5. Active vs passive moving source tomography: Comparing results from the Santa Barbara Channel Experiment (SBCEx16) on sources of opportunity. Jit Sarkar, Bruce Cornuelle, Kay L. Gemba, William A. Kuperman, William S. Hodgkiss (Scripps Inst. of Oceanogr., UC, San Diego, 9500 Gilman Dr., Mail Code 0238, La Jolla, CA 92093-0238, bsarkar@ucsd.edu), Karim G. Sabra (College of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Jeffery D. Tippmann, and Christopher M. Verlinden (Scripps Inst. of Oceanogr., UC, San Diego, La Jolla, CA)

An experiment was performed in the Santa Barbara Channel (SBCEx16) using four vertical line arrays (VLAs) of hydrophones placed midway between the in- and out-going shipping lanes of a maritime highway. The goal of this experiment was to study whether these sources of opportunity can be used for passive tomographic purposes. In addition to the continuous passive observations, active acoustic source-tows were conducted. The environment was monitored continuously throughout the course of the experiment using thermistor strings, and ship tracks were obtained from the Automatic Identification System (AIS). The results from both types of tomography are presented and compared.

3:00

4pSPa6. Using observations of commercial shipping from horizontal arrays deployed close to a shipping lane to map seabed properties. Paul Hursky (Sonar-synesthetics, 4274 Pilon Point, San Diego, CA 92130, paul.hursky@gmail.com)

Several horizontal arrays were deployed on the seabed off the coast of Florida, outside Port Everglades. This is a busy site for cruise and container ships, commercial shipping obligated to transmit Automatic Identification System (AIS) information, indicating own lat-longs, as well as other parameters of their identity and motion. Acoustic data from these arrays and AIS transmissions were recorded for several days, capturing many passing ships that could then be used as sources at known locations for inversion of seabed properties. One of the arrays had geophones sensing three orthogonal vector components of particle velocity, as well as omni-directional pressure. This area is known to have a distressingly range-dependent seabed, making it difficult to successfully perform matched field processing, for example. We will present our processing of this data, including adaptive beamforming of the geophone data and methods that exploit beam migration patterns (in angle and time difference of arrival) to help isolate multipath arrivals, which are mapped via back-propagating rays to specific grazing angles and locations on the sea floor to provide measurements of bottom loss. The complete set of such observations from ship tracks at multiple frequencies, are incorporated into an inversion for seabed properties.

3:20

4pSPa7. Using vessel noise from a single hydrophone to estimate environmental properties. Graham A. Warner (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, graham.warner@jasco.com), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada)

This paper estimates seabed and water-column properties of a shallow-water site in the Chukchi Sea using vessel noise recorded on a single ocean-bottom hydrophone. A shallow-hazards seismic survey vessel transited with a fixed heading, passing within 200 m of the hydrophone. Sound pressure levels as a function of frequency and range are inverted using a trans-dimensional (trans-D) Bayesian approach to estimate range-independent environmental properties (sound-speed profile, water depth, and seabed geoacoustic properties). The trans-D inversion allows the data to determine the most appropriate environmental model parameterization in terms of the number of sound-speed profile nodes and subbottom layers. The inversion also estimates the vessel source levels, source depth, hydrophone height above the seabed, a range-correction factor, and error statistics, and provides uncertainty estimates for all model parameters and parameterizations. The sound-speed profile is found to be in good agreement with a measured profile, and the upper sediment-layer sound speed agrees with estimates from separate inversions of airgun and bowhead whale modal dispersion data. This method could be applied to monitor sound-speed profile changes over long time scales, e.g., using Automatic Identification System position data for vessels in a shipping lane.

3:40

4pSPa8. Echolocating dolphins as opportunistic signal sources for observing fine structure in the impulse responses of acoustic scatterers. Eric L. Ferguson, Stefan B. Williams (Australian Ctr. for Field Robotics, School of Aerosp. Mech. and Mechatronic Eng., University of Sydney, NSW, Australia), Craig T. Jin (Computing and Audio Res. Lab., School of Elec. and Information Eng., Sydney, NSW, Australia), and Brian G. Ferguson (Dept. of Defence, Defence Sci. and Technol., PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

Echolocating dolphins emit wideband acoustic signals as sequences of short duration pulses or *clicks*. Often, the waveforms of the clicks (especially during a buzz phase) have simple shapes that resemble a Ricker wavelet or a Gaussian modulated sinusoidal pulse. The main energy lobe of the pulse is of short duration (about 20 microseconds) with an inverted monopulse shape indicating a rarefaction (negative pressure pulse). An experimental bistatic active sonar system is configured in which a dolphin's biological sonar transmissions are coupled with a wide aperture receiving array. The acoustic transmissions of free-ranging Indo-Pacific bottlenose (*Tursiops aduncus*) dolphins are sampled every 4 microseconds at three collinear hydrophones having an interelement spacing of 14 m and located 1 m above the sea floor in shallow water of depth 20 m. Insonification of the seafloor and sea surface boundaries by the short-duration pulses reveals fine structure in the observed forward scattering impulse responses of the boundaries. Features (or highlights) in the forward scattering impulse responses are measured to within a fraction of a microsecond leading to precise localization of the scatterer using the modified wavefront curvature passive ranging method. Also, it will be shown that a similar result is observed for a scatterer in the water column that was moving rapidly with respect to the echolocating dolphin.

Session 4pSPb

Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity II

Kay L. Gemba, Cochair
MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Room 446, La Jolla, CA 92037

Jit Sarkar, Cochair
Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Drive, Mail Code 0238, La Jolla, CA 92093-0238

Invited Papers

4:05

4pSPb1. Cascade of blind deconvolution and array invariant for robust source-range estimation. Hee-Chun Song, Chomgun Cho (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), gihoon byun, and Jea Soo kim (Korea Maritime and Ocean Univ., Busan, N/A, South Korea)

The array invariant proposed for robust source localization in shallow water is based on the dispersion characteristics in ideal waveguides. It involves conventional plane-wave beamforming using a vertical array, exploiting multiple arrivals separated in beam angle and travel time, i.e., beam-time migration. The approach typically requires either a short pulse emitted by a source or the Green's function that can be estimated from a probe signal to resolve distinct multipath arrivals. In this letter, the array invariant method is extended to unknown source waveforms by extracting the Green's function via blind deconvolution. The cascade of blind deconvolution and array invariant for robust source-range estimation is demonstrated using a 16-element, 56-m long vertical array at various ranges (1.5–3.5 km) from a towed source broadcasting broadband communication waveforms (0.5–2 kHz) in approximately 100-m deep shallow water.

4:25

4pSPb2. Acoustic interrogation of objects using signals from rotorcrafts. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

Rotorcraft generate acoustic waveforms that are repeatable, loud, and contains a large number of harmonics. They are a good source for generating step frequency-like waveforms that propagate over long distances and can easily be moved to interrogate new locations. One potential application of this source is a monostatic synthetic aperture sodar (SAS). A simple simulation was performed using back projection to generate SAS imagery. The movement of the helicopter generates the synthetic aperture, the signals generated by the main and tail rotors produce the waveform, and microphones mounted on the helicopter receive the reflected sound waves. The measured acoustic signals needs to be motion-compensated and focused, similar to the processing in synthetic aperture radar. A signal bandwidth of 170 Hz will generate an acoustic image with a downrange resolution (c/2B) of approximately 1 meter, and a 100 meter aperture will generate an acoustic image with a crossrange resolution (λ/d) of approximately 2 degrees for a center frequency of 100 Hz. Issues such as turbulence, saturation from self-noise, waveform estimation, acoustic reflectance of targets, range ambiguity, and wind need to be included in future simulations to obtain more realistic results.

4:45

4pSPb3. Using nonlinear time warping to estimate North Pacific right whale calling depths and propagation environment in the Bering Sea. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Julien Bonnel, Margaux Thieury (ENSTA Bretagne, Brest cedex 9, France), Aileen Fagan (U.S. Coast Guard Acad., New London, CT), Christopher M. Verlinden (SIO, UCSD, La Jolla, CA), Dana Wright, Jessica Crance, and Catherine L. Berchok (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., Seattle, WA)

Calling depth distributions and ranges are estimated for two types of calls produced by critically endangered eastern North Pacific right whales (NPRW) in the Bering Sea, using passive acoustic data collected with bottom-mounted single-hydrophone recorders in 50–90 m water depths. Nonlinear time resampling of 12 NPRW "upcalls" and 20 broadband "gunshots" typically isolated 3 to 4 individual mode arrivals below 200 Hz. Matched-mode processing (MMP) methods were used to remove the unknown source phase and amplitude structure, but incoherently averaging MMP ambiguity surfaces across frequency yielded many local minima when inverting for sediment properties. Instead, the ambiguity surfaces were plotted as a function of range and frequency, which revealed the type and degree of environmental mismatch present in initial waveguide models. This qualitative approach revealed the existence of large sound speed gradients in the sediment, along with downward-refracting sound speed profiles during the summer months. Gunshot sounds were generally produced at a few meters depth, while upcall depths clustered between 10 and 25 m, consistent with previously published bioacoustic tagging results from North Atlantic right whales.

Contributed Papers

5:05

4pSPb4. Array invariant-based localization using ships of opportunity. Gihoon Byun, Jea Soo kim (Ocean Sci. and Technology-Korea Maritime and Ocean Univ., Korea Maritime Univ., Dongsam 2-dong, Yeongdo-gu, Busan 606-791, South Korea, gihoonbyun77@gmail.com), Chomgun Cho, Hee-Chun Song (Scripps Inst. of Oceanogr., Univ. of California, La Jolla, CA), and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

The array invariant (AI) proposed for robust source-range estimation with minimal knowledge of the environment in shallow water is based on the dispersion characteristics in ideal waveguides. This approach involves plane wave beamforming, utilizing coherent multiple arrivals separated in beam angle and travel time, referred to as "beam-time migration." To resolve multipath arrivals in beam-time domain, AI requires either an impulsive source or Green's function typically estimated from a known probe signal. For unknown source waveforms, it is possible to estimate the Green's function using a ray-based blind deconvolution (RBD) which also utilizes simple conventional beamforming. Recently, the cascade of RBD and AI has been demonstrated for a towed source at 50-m depth broadcasting communication waveforms [*J. Acoust. Soc. Am.* **141**, 3270–3273 (2017)]. Rather than the towed source, this study focuses on the feasibility of tracking a ship radiating random and anisotropic noise. The combination of RBD and AI is demonstrated to localize a ship of opportunity (200–900 Hz) along a track at ranges of 1.8–3.4 km and a 16-element, 56-m long vertical array in approximately 100-m deep shallow water.

5:20

4pSPb5. Blind deconvolution of sources of opportunity in ocean waveguides using bilinear channel models. Ning Tian, Justin Romberg, and Karim G. Sabra (Georgia Inst. of Technol., 75 5th St. NW, Atlanta, GA 30308, ningtian@gatech.edu)

We develop a general algorithmic framework for acoustic imaging using sources of opportunity. While these sources (e.g., ships at known locations) emit high-energy signals, we do not in general have knowledge of the precise waveforms (signatures) of these signals. Consequently, both Channel Impulse Responses (CIRs) and unknown source signals need to be simultaneously estimated from only the recorded signals on a receiver array using blind deconvolution, which is generally an ill-posed problem without any a priori information or additional assumptions about the underlying structure of the CIRs. By exploiting the typical ray-like arrival-time structure of the CIRs between a surface source and the elements of a vertical line array (VLA) in ocean waveguides, we apply a principle component analysis technique to build a bilinear parametric model linking the amplitudes and arrival-times of the CIRs across all channels for a variety of admissible ocean environments. The bilinear channel representation further reduces the dimension of the channel parametric model compared to linear models. We then develop a truncated power interaction deconvolution algorithm by applying the bilinear channel model to the traditional subspace deconvolution method. Numerical and experimental results will demonstrate the robustness of this blind deconvolution methodology.

5:35

4pSPb6. Experimental demonstration of passive acoustic tracking using a library of nearby sources of opportunity. Christopher M. Verlinden, Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu), Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), William S. Hodgkiss, and William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Acoustic sources can be tracked in the marine environment using a method that is similar to traditional matched field processing but differs in that it uses a library of data-derived replicas in place of modeled replicas. In order to account for differing source spectra between library and target vessels, cross-correlation functions are compared instead of comparing acoustic signals directly. Measured replicas are extrapolated to fully populate a search grid using waveguide invariant theory. The method has been demonstrated experimentally for localizing surface contacts in a shallow water (150 m) environment. Further, we present experimental validation of the method in deeper water and discuss the extension of the technique to subsurface contacts

5:50

4pSPb7. Localization of circularly towed sources using machine learning methods. Emma Reeves (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu), Haiqiang Niu (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Localization of a source towed in a circular geometry is accomplished using machine learning methods, wherein the relationship between received pressure and source range is learned directly from the acoustic data. Feed-forward neural networks (FNN), sup- port vector machines (SVM), and random forests (RF) have accurately estimated the range of a source towed in a linear geometry in shallow water (arXiv: 1701.08431v3). Here, we apply FNN, SVM, and RF to a circular source tow using data from the SCEx17 experiment to examine the effect of varying source-receiver geometries. The Sample Covariance Matrix (SCM) is constructed, vectorized and used as the machine learning input. For FNN and RF, these input vectors are combined to form a d x N matrix and an additional preprocessing step is applied to improve classification results. The input matrix is projected onto a d x k basis formed from the top k eigenvectors of its scatter matrix. This compact representation results in improved computation time and performance for FNN and RF compared with using the SCM inputs.

Session 4pUW

Underwater Acoustics: Arctic Acoustics

Matthew Dzieciuch, Cochair
SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225

Jason D. Sagers, Cochair
Environmental Science Laboratory, Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

1:00

4pUW1. The 2016–2017 deep-water Canada Basin Acoustic Propagation Experiment (CANAPE): A preliminary report. Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), Andrey Y. Proshutinsky, Richard A. Krishfield (Dept. of Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA), Jonathan D. Nash (Dept. of Earth, Ocean & Atmos. Sci., Oregon State Univ., Corvallis, OR), and John N. Kemp (Appl. Ocean Phys. & Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Arctic Ocean is undergoing dramatic changes in both ice cover and ocean structure. The Canada Basin Acoustic Propagation Experiment (CANAPE), which includes both deep and shallow water components, was designed to understand the effects of changing Arctic conditions on low-frequency propagation and ambient noise. The deep-water component, which is reported on here, includes a yearlong experiment in the Canada Basin during 2016–2017, preceded by a short Pilot Study during July–August 2015. During 2016-2017, a Distributed Vertical Line Array (DVLA) receiver with 60 Hydrophone Modules was moored within a six-element acoustic transceiver array with a 150-km radius. Environmental measurements on the DVLA include 28 Sea-Bird MicroCATs and upward- and downward-looking Acoustic Doppler Current Profilers (ADCPs) located below the hydrophone array. The acoustic transceivers had sources at 175-m depth and 15 Hydrophone Modules located above the sources. Environmental measurements on the transceiver moorings include ice-profiling sonars, upward-looking ADCPs on the subsurface floats, and 10 temperature sensors located below the acoustic transceivers. The one-year deployment provides measurements at least partially in open water during summer, in the marginal ice zone (MIZ) as it transitions across the array during the spring and autumn, and under complete ice cover during winter.

1:15

4pUW2. Acoustic propagation from the Canadian Basin to the Chukchi Shelf. Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Ying-Tsong Lin (AOPE Dept MS 11, Woods Hole Oceanogr. Inst., Woods Hole, MA), Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), and Sean Pecknold (Defence Res. and Development Canada-Atlantic, Dartmouth, NS, Canada)

The Pacific Arctic Region, encompassing the Bering, Chukchi, Western Beaufort, and Eastern Siberian shelves and seas, has experienced decadal changes in atmospheric conditions, seasonal sea-ice coverage, and seawater temperature. In the summer of 2016, the Canada Basin Acoustic Propagation Experiment (CANAPE) was conducted to understand the changing soundscape and to explore the use of acoustic signals as a remote sensing tool in the modern Arctic. During the experiment, low-frequency signals from five tomographic sources located in the Canada Basin were recorded by a short vertical line array of hydrophones deployed from a research vessel. The recordings were made at seven stations located in the Canada Basin, on the continental rise, and on the Chukchi Shelf. The propagation distances ranged from 50 km to 500 km, and the propagation conditions changed from ducted by the Beaufort Lens in the basin to upward refracting on the continental shelf. Multiple measurements of the sound speed profile were acquired at each station to characterize the temporal and spatial variability of the sound speed field. This talk examines the range-dependent measurements and explains the observed variability in the received signals through propagation modeling. [Work sponsored by ONR.]

1:30

4pUW3. Underwater sound propagation variability over the Chukchi Sea continental slope. Ying-Tsong Lin, Weifeng G. Zhang, Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, Woods Hole, MA 02543, ytlin@whoi.edu), Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Sean Pecknold (Defence Res. and Development Canada-Atlantic, Dartmouth, NS, Canada)

In the Canada Basin and Chukchi Sea regions, a vertical sound duct can be formed between the Pacific Summer Water Layer on the top and the Atlantic Water Layer on the bottom, providing an acoustic pathway connecting the deep basin and the shallow shelf over the Chukchi Sea continental slope. Previous studies have shown that the shelfbreak circulation (specifically upwelling), the sub-mesoscale eddies spun off the shelfbreak jet, and the ice coverage are the three major causes of the temporal and spatial variability of the Pacific Summer Water Layer in the region. In this paper, numerical simulations utilizing the Parabolic-Equation (PE) method are conducted to investigate the sound propagation variability over the Chukchi Sea shelfbreak and slope, along with an idealized ocean circulation model providing the fundamental basis of water-column fluctuations. The sound pressure sensitivity kernel derived from the acoustic propagator in the PE method is also used to provide physical insights into the sound propagation variability. The acoustic data over the northern Chukchi Sea shelfbreak collected during the 2016 shallow water cruise of the Canada Basin Acoustic Propagation Experiment will be examined and compared with the numerical simulations. [Work supported by the Office of Naval Research.]

1:45

4pUW4. Oceanographic measurements on Chukchi shelf break during fall 2016. Mohsen Badiey, Andreas Muenchow, Justin Eickmeier, and Lin Wan (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Newark, DE 19716, badiey@udel.edu)

During the Canada Basin Acoustic Propagation Experiment (CANAPE) in 2016, two extended shipboard oceanographic measurements were conducted simultaneously with the acoustic propagation from deep water to the Chukchi shelf-break region. These shipboard measurements were aimed at understanding the oceanographic variability including along the shelf eddy formation and upwelling around the shelf break region. We utilize the measured oceanographic data to construct the environmental input for acoustic models. While there was no ice formation or coverage during the observation period, the effects of upwelling and the eddy formation on acoustic propagation were present. This paper demonstrates the results measured in the experiment in the context of temporal and spatial variability of the water column in the Chukchi shelf region. [Work supported by ONR Ocean Acoustics].

2:00

4pUW5. Acoustic characterization of the new Arctic using mobile acoustic sources. John E. Joseph, D. Benjamin Reeder, and Liam J. Doyle (Oceanography, Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 315A, Monterey, CA 93943, jejoseph@nps.edu)

The Naval Postgraduate School (NPS) participated in Ice Exercise 2016 (ICEX-16), a multi-national naval exercise conducted in the Beaufort Sea during March 2016. Operating at the remote Ice Camp SARGO, NPS deployed several conductivity, temperature and depth (CTD) sensors to capture oceanographic variability to 500 m depth while performing a series of propagation tests. Four mobile mid-frequency sources transmitted signals for approximately 10 hours each to a pair of vertical line array receivers positioned in the field to investigate depth, range, angular and specular characteristics of acoustic propagation and their correlation to variability in oceanographic structure and under-ice conditions. CTD data indicated significant variability in sound speed at 50 m depth where cold, fresh mixed-layer water interfaces with contrasting warm, saline Pacific Summer Water (PSW) that lays immediately below it. The data also show a persistent and stable subsurface sound channel existed as a result of the PSW with peak temperature at 80 m situated above colder Pacific Winter Water (PWW), resulting in a sound channel axis near 140 m depth. Both features have important implications on sonar performance in the Arctic. Modeled and measured transmission loss are compared to quantify the effects.

2:15

4pUW6. Limitations of a Gaussian Beam Model for low-frequency acoustic propagation under ice. Sean Pecknold (DRDC Atlantic, PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca), Diana F. McCammon (Maritime Way Sci., Waterville, NS, Canada), and Dale D. Ellis (Maritime Way Sci., Dartmouth, NS, Canada)

Ray-based propagation models are often used to simulate underwater acoustic communications signals. For longer-range low frequency acoustic communications, particularly under ice cover, it is necessary to determine if a ray-based model can give accurate results. An elastic ice reflection algorithm was added to the Bellhop Gaussian beam model, and the results for mid- and low-frequency propagation were compared to results obtained using a normal modes model adapted for an elastic ice reflection coefficient, and to the OASES wavenumber integration model. The effects of simplified boundary conditions, frequency, and ice thickness on amplitude and phase matching are examined.

2:30

4pUW7. Modeling annual variations in high-frequency scattering from sea ice. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu)

As sea ice evolves naturally through its different growth and decay structures, so too does it manifest important changes in physical properties affecting its acoustic response. Variations in the porosity, permeability, and roughness of the ice lead to changes in scattered and reflected acoustic energy as well as the degree of attenuation. Knowledge of the long-term variations of high-frequency under-ice scattering over annual formation and melting cycles is essentially non-existent as only a limited number of measurements have been performed. Owing to this lack of data, fundamental questions remain as to the dominant mechanisms influencing the under-ice scattering process at any given time. In this work, we will show modeling results for high-frequency acoustic scattering from different types of sea ice as it forms and melts, highlighting the relative influence of various ice parameters on mean scattered levels and the dependence of these levels on frequency and grazing angle. Results yield insight into controlling factors and lead to recommendations for planned long-term measurements from moored instruments.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

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

Committees meeting on Tuesday, 5 December

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Studio 7
Acoustical Oceanography	7:30 p.m.	Salon A/B/C
Animal Bioacoustics	7:30 p.m.	Salon F/G/H
Architectural Acoustics	7:30 p.m.	Studio 9
Musical Acoustics	7:30 p.m.	Studio 4
Physical Acoustics	7:30 p.m.	Balcony L
Psychological and Physiological Acoustics	7:30 p.m.	Balcony M
Structural Acoustics and Vibration	8:00 p.m.	Studio 7

Committees meeting on Wednesday, 6 December

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Balcony M
Signal Processing in Acoustics	7:30 p.m.	Salon D

Committees meeting on Thursday, 7 December

Committee	Start Time	Room
Noise	7:30 p.m.	Studio 2
Speech Communication	7:30 p.m.	Salon A/B/C
Underwater Acoustics	7:30 p.m.	Salon F/G/H