Session 2aAA

Architectural Acoustics: Architectural Acoustics and Audio: Even Better Than the Real Thing

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. Analysis of the effect on the indoor acoustic comfort with the living surfaces in the design studio. Filiz B. Kocyigit and Nazli N. Yildirim (Interior Architecture & Environ. Design, Atilim Univ., incek, Ankara 06560, Turkey, filizbk@gmail.com)

It is clearly shown in studies that acoustic comfort in the interiors play an effective role on directly increasing the productivity of users. Despite having the similar physical characteristics as the classrooms, design studios differ in terms of function and user behaviors. The training processes in the design studios have complicated function, but this process is limited in the classrooms. Recent research shows that living surface contribute to acoustical control at different frequencies, during the design periods at the studios. In these studies, living surfaces that provide indoor acoustic comfort are investigated in the general context. The effects of selected plants' structural differences on the indoor sound control at different frequencies are analyzed. Three different plants such as grass, moss, and aloevera are selected and their behaviors controlling the sounds of high, medium, and low frequencies in the indoor space are investigated. The behavior of these plants for the control of sounds in different frequencies in the studios has been examined. It is aimed to increase the educational efficiency of the users by using the combination of the most effective plant species in the studios' surface in terms of indoor acoustic comfort within the scope of the present data.

8:30

2aAA2. Sound system Déjà Vu all over again!—Revisiting past design approaches in an acoustically challenging multi-purpose space. Deb Britton and Dustin Goen (K2, LLC, 5777 Central Ave., Ste. 225, Boulder, CO 80301, deb@k2audio.com)

In the late 90s, we were tasked with designing a sound reinforcement system that would provide great speech intelligibility, in a highly reverberant space, without modifying any of the architectural finishes. The sound system had to support different types of events, held in different locations within the space, and with varying audience sizes. In 2015, we were asked to revisit the original loudspeaker design, and upgrade the sound system in that same space. Flash forward a few more years, and the entire building is getting a renovation—and we are once again designing a completely new sound system. This paper presents a case study of the upgrades, and describes the challenges, the lessons learned, and the changes in approach from the original upgrade through to the most recent design.

8:55

2aAA3. Beyond the chamber, plate, or digital reverb. Electronic architecture and the acoustically flexible recording environment. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com) and Russ Berger (RBDG, Carrollton, TX)

For more than 20 years, Electronic architecture has been successfully integrated in a variety of venues for live performances. We will discuss how the inclusion of this technology in recording facilities has both altered recording methods, and expanded clientele at three recording facilities: The Dimenna Center for Classical Music is New York's premier facility for recording acoustic music. Recent projects include the soundtrack for the Coen Brothers' feature film *Hail, Caesar!;* the Grammy-nominated cast album for *The King and I* featuring Kelli O'Hara and Ken Watanabe; the soundtrack for James Schamus' *Indignation;* a Grammy Award-winning album by Room Full of Teeth; and an upcoming harmonia mundi release featuring Pablo Heras-Casado and Orchestra of St. Luke's. Sweetwater Music has recently expanded its recoding capabilities and upgraded the E-Coustic system with the latest hardware and software. Recent sessions include the Counting Crows, Adrian Belew, and Steve Curtis Chapman. Fo'Yo' Soul is Grammy award winning Kirk Franklin's new facility in Arlington Texas designed by Russ Berger. The studio (Uncle Jesse's Kitchen) features one of the first E-Studio product installations that allows acoustical conditions in the studio to be adjusted at the touch of a button, allowing soloists and ensembles to record in an environment tailored for comfort and creativity.

9:20

2aAA4. Audio rendering based on separation of direct and diffuse sound. Akio Ando (Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 3190 Gofuku, Toyama 930-8555, Japan, andio@eng.u-toyama.ac.jp)

Multichannel audio with many channels can capture the sound field with higher sense of reality. In the reproduction, however, a system with many loudspeakers can be setup only in the special room like a studio. Therefore, the audio rendering that converts the number of channels should be necessary in the reproduction. Many studies have been made on the audio rendering. These methods were based on the directional information of channels. On the other hand, there are two kinds of sense of reality: the reality by the object sound ("object reality") and that by the field sound ("field reality"). There are also two kinds of sound in the multichannel audio signal: the direct and diffuse sound. The direct sound brings the object reality, and the diffuse sound brings the field reality. Since the conventional rendering methods rely on the directional information, it may be difficult to handle the diffuse sound. In this paper, we address a rendering method based on the separation of direct and diffuse sound. Since the direct sound can be rendered by the conventional method, we clarify the requirements for the diffuse sound rendering from the viewpoint of sound reflection in the room.

9:45

2aAA5. Please turn it down!. Thomas J. Plsek (Brass/Liberal Arts, Berklee College of Music, MS 1140 Brass, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu)

Sustained levels of 95 to 100 dBA and beyond are not uncommon in rock and pop music genres, leading to daily noise doses that are 100% or more above the recommendations established by the National Institute for Occupational Safety and Health. There are three options of addressing this situation: leaving the environment, using hearing protection, and turning the offending level down. It is the last of these that I will explore. The first two listed have problems with implementation. Using Leq measurements of various ensemble performance situations, the question addressed is whether or not musicians can learn to control the level of their instruments efficiently enough, referencing NIOSH recommended noise exposures, to make a meaningful difference in exposure time without significantly sacrificing the quality of the music.

10:10–10:25 Break

10:25

2aAA6. Effects for emphasis, enhancement, expression, and exaggeration. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Sound recordings built through multitrack production face a competitive soundscape. Signal processing strategies have evolved with the production goal of increasing the audibility of specific signal traits — making the subtle more apparent, the performance more compelling, the recorded art more successful.

10:50

2aAA7. AtatTATürk's Sarcophagus Acoustic Properties. Filiz B. Kocyigit (Architecture, Atilim Univ., incek, Ankara 06560, Turkey, filizbk@gmail.com)

This research investigates the acoustical characteristics of Atatürk sarcophagus. The Hall of Honor, which included the Atatürk's sarcophagus, was elevated from the whole mass, as it seemed strongly to the outside architect. The first floor is surrounded by this hall. The monument was built on a platform raised by six meters with a staircase in the direction of the square, with the ground closed and a massive wall with small windows, which was turned over with stone columns. The east entrance of the monument is at the beginning of the Aslan Road. At the beginning of this road, the entrance is strengthened by two guards, four meters high stairs are raised. When the monument is placed on the hill, two strong axes perpendicular to each other are taken as principal. Results, at the research, include equivalent sound pressure levels (Leq) as a function of location, frequency, and time of day. Special and important days like Revolution days are also researched. The spectra are generally flat over the 63–2000 Hz octave bands, with higher sound levels at lower frequencies, and a gradual roll off above 2000 Hz. Many units exhibit little if any reduction of sound levels in the nighttime.

11:15

2aAA8. Nature as muse: How cave acoustics can help us be more imaginative with our reverb choices. Yuri Lysoivanov (Recording Arts, Flashpoint Chicago, 28 N. Clark St. #500, Chicago, IL 60602, yuri.lysoivanov@columbiacollege.edu)

The cave is an oft-overlooked option for the audio practitioner's toolbox. With the aid of the science team from the National Park Service, we recorded impulse responses and analyzed the properties of several acoustically distinctive spaces inside Mammoth Cave in Kentucky, the longest cave network in the world. An understanding of the unique characteristics of these areas can help inform more creative approaches in reverb selection and educate us to be more innovative in music, sound, and room design.

Contributed Paper

11:40

2aAA9. Requirement of acoustical engineering discipline in building designs. Zohreh Razavi (Norman Disney & Young, 1166 Alberni St., Vancouver, BC V6E 3Z3, Canada, z.razavi@ndy.com)

While assurance of professional design and commitment for field review from all engineering disciplines and an architect is required, acoustical review is not mandated by legislations or By-laws. Although sound control review requirement is mandated by most By-Laws, this is at an architect's discretion to decide whether retaining an acoustical engineer for a sound control review for a project is required or not. Architects, even those focused on sustainable and human-centered design have often overlooked acoustical quality as a design goal. The launch of programs like the "WELL Building Standard" which includes several features related to acoustics, and addition of the "acoustic performance credit to LEED v4" are signs that the green building industry is starting to promote acoustic performance an integral component of sustainable design. Though architects are more aware of the importance of acoustics, achieving good acoustic performance is still overlooked. Once acoustical compliance letter is mandated during building permit, the importance of acoustics in building designs will be recognized.

TUESDAY MORNING, 6 NOVEMBER 2018 SHAUGHNESSY (FE), 9:00 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics: Topics in Animal Bioacoustics: Hearing

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

Contributed Papers

9:00

2aAB1. Impact of stimulus bandwidth on the estimate of the upper-frequency limit of hearing in toothed whales. Dorian S. Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), Sean P. Coffinger (Univ. of California, San Diego, CA), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (U.S. Navy Marine Mammal Program, San Diego, CA), and Robert F. Burkard (Univ. at Buffalo, Buffalo, NY)

No consensus on the stimuli used in toothed whale hearing tests using evoked potential methods currently exists. However, stimulus bandwidth should directly affect determination of the upper-frequency limit (UFL) of hearing because of the steep reduction in hearing sensitivity at frequencies immediately below the UFL—the broader the stimulus bandwidth, the greater the amount of spectral "splatter" into lower frequencies where hearing is more sensitive. To test this hypothesis, variability in auditory evoked potential thresholds to tone-pips (with variable bandwidth) and amplitude-modulated tones was determined in dolphins near their UFL of hearing and ½ to 1 octave below where hearing is more sensitive. Subjects included both normal and hearing-impaired dolphins. At frequencies where hearing was sensitive and the audiogram was relatively flat, negligible changes in the hearing threshold with changing stimulus bandwidth were observed. Conversely, thresholds near the UFL of hearing declined with increasing stimulus bandwidth, regardless of whether dolphins had normal hearing or hearing impairment, and resulted in broadened hearing bandwidth estimates. Thus, it is recommended that stimulus bandwidth be reported in hearing tests where the UFL is tested since stimulus bandwidth affects thresholds through spectral splatter into lower frequency bands where hearing is more sensitive.

9:15

2aAB2. Forward masking recovery in bottlenose dolphins (*Tursiops truncatus*): Auditory brainstem responses to paired-click stimuli in high-pass masking noise. Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), Sean P. Coffinger (Psych. Dept., Univ. of California - San Diego, San Diego, CA), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Robert F. Burkard (Dept. of Rehabilitation Sci., Univ. at Buffalo, Buffalo, NY)

Adaptation is a fundamental process in the auditory system and underlies the automatic gain control system of echolocating bottlenose dolphins. As pulse-echo delay increases, auditory brainstem response (ABR) amplitudes to the echoes progressively increase. This study examined adaptation across cochlear frequency regions using a paired-click (i.e., forward masking) paradigm and high-pass masking noise. Bottlenose dolphins passively listened to paired click stimuli (20–160 kHz "pink" spectra). A "conditioning" click was followed by a "probe" click with equal amplitude and a time delay ranging from 125 to 750 μs. ABRs to click pairs were obtained with and without high-pass masking noise that precluded the basal turn of the cochlea from responding to the click stimuli. The ABR evoked by a single click (temporally aligned with the first click of the paired-click condition) was subtracted from the click-pair ABR to visualize the response evoked by the probe click. Probe ABR amplitudes recovered linearly with increasing delay relative to the conditioning click, and were approximately 70% of full response amplitude at 750-μs delay. Paired-click interval ABR amplitude recovery functions were similar for the unmasked and high-pass masked conditions. [Work sponsored by ONR.]

9:30

2aAB3. Responses of the auditory system of odontocetes to level changes of long lasting acoustical stimuli. Evgeniya Sysueva, Dmitry Nechaev, Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com), and Vladimir Popov (Inst. of Ecology and Evolution, Moscow, Russian Federation)

The effects of level changes of long lasting sound stimuli (tone pip trains) on evoked potentials (the rate following response, RFR) were investigated in a beluga whale (*Delphinapterus leucas*). The stimuli were of 64 kHz carrier frequency at levels from 80 to 140 dB re 1 μPa. During stimulation, the stimulus level either was kept constant (the steady-state stimulation) or was changed up/down by 20 or 40 dB every 1000 ms. After transition from a lower to upper stimulus level (increase), the response amplitude increased quickly and then decayed slowly. After transition from an upper to lower stimulus level (decrease), the response amplitude fell quickly and later recovered slowly. In the both cases, during the 1000 ms stimulus, the response amplitude almost reached that of the steady-state stimulus of the same level. The auditory system of the beluga (and, hypothetically, other odontocetes) may be characterized as quickly flexible and capable of quick adjustment of its responses to the current auditory scene. [This study was supported the Russian Science Foundation (Project No. 17-74-20107) awarded to E.V.S.]

9:45

2aAB4. Directional hearing sensitivity for 2–30 kHz sounds in the bottlenose dolphin (*Tursiops truncatus*). Alyssa W. Accomando, Jason Mulsow, Brian K. Branstetter (Biologic and BioAcoust. Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., #200, San Diego, CA 92106, alyssa.accomando@nmmpfoundation.org), James J. Finneran, and Keith Jenkins (US Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA)

Bottlenose dolphins (*Tursiops truncatus*) depend on sounds with frequencies <30 kHz for social communication and foraging, but little information on the directional dependence of hearing thresholds for these sounds exists. This study measured behavioral hearing thresholds for 2, 10, 20, and 30-kHz tones projected from eight different angular positions around the dolphin in both the horizontal and vertical planes to determine whether the receiving beam at these frequencies was directional. Omni-directional hearing was hypothesized for sounds below 30 kHz, but this hypothesis was rejected. Results from two bottlenose dolphins demonstrated a positive relationship between directivity and the frequency of the test tone, with asymmetric beam patterns. Directional hearing sensitivity declined most dramatically between 10 kHz and 2 kHz. The results suggest that dolphins' directional hearing is more pronounced for lower frequencies than previously predicted.

10:00

2aAB5. Passive echo detection and discrimination in bottlenose dolphins. Sean P. Coffinger (Psych., Univ. of California - San Diego, 10131 Caminito Zar, San Diego, CA 92126, sean.coffinger@gmail.com), Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Jason Mulsow (National Marine Mammal Foundation, San Diego, CA)

Echolocation in bottlenose dolphins (*Tursiops truncatus*) is an active process involving outgoing signal transmission and echo reception. Transmitted echolocation "clicks" interact with a target to produce echoes that contain representative information about the target. Echolocation allows dolphins to detect, discriminate, and identify complex targets using only their hearing. However, it is unclear whether dolphins capitalize on an internal representation of echolocation clicks for matched filter processing or whether the same information (and performance) can be obtained through passive echo reception (i.e., without a corresponding ensonifying click). Here, we investigate the passive echo discrimination ability of a bottlenose dolphin by embedding a target echo in a stream of distractor echoes with similar spectral and time-domain characteristics. Initial testing has demonstrated excellent echo-discrimination ability in the light of multiple distractor echoes of similar amplitude. Additional testing is underway with expanded distractor sets and echo-ordering complexity. The paradigm specifically investigates the plausibility of perceptual identification without the potential for matched filter processing. If representational identity is attainable via a passive process and not degraded relative to active echolocation, then the function of the echolocation click might not serve in matched filter processing, though it would remain important for ensonification and in determining target location.

10:15–10:30 Break

10:30

2aAB6. Dependence of bottlenose dolphin (T*ursiops truncatus*) auditory brainstem responses on noise burst rise time, amplitude, and envelope shape. Ryan A. Jones (Biologic and BioAcoust. Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, ryan.jones@nmmpfoundation.org), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), Jason Mulsow (Biologic and BioAcoust. Res., National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Dept. of Rehabilitation Sci., Univ. at Buffalo, Buffalo, NY)

A series of experiments examined the dependence of dolphin auditory brainstem response (ABR) peak amplitudes and latencies by stimulus level, rise times, and envelope shape. Stimuli were spectrally "pink" noise bursts with frequency content from 10 to 160 kHz, rise times that varied from 32 μs to 4 ms, and plateau sound pressure levels from 102 to 138 dB re 1 μPa. Cosine and cosine-cubed rise (and fall) envelopes were tested, and results were compared to those obtained for dolphins using noise bursts with linear rise envelopes [Finneran *et al*., J. Acoust. Soc. Am. **143**, 2914–2921 (2018)]. Results for the cosine and cosine-cubed envelopes showed that ABR peak amplitudes were a function of envelope sound pressure at the end of a fixed integration window of approximately 260 μs. This is congruent with the previous findings for linear envelopes. The ABR peak latencies for the cosine and cosine-cubed envelopes were dependent on the second derivative of the envelope pressure function, as opposed to the first derivative of the pressure function for linear envelopes. These data are consistent with single-unit and nearfield response data for terrestrial mammals. [Work supported by Navy Living Marine Resources and SSC Pacific Naval Innovative Science and Engineering Programs.]

10:45

2aAB7. The production and reception of underwater sound in Hawaiian monk seals (*Neomonachus schauinslandi*). Jillian Sills, Kirby Parnell, and Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California Santa Cruz, 115 McAllister Way, Santa Cruz, CA 95060, coll@ucsc.edu)

The endangered Hawaiian monk seal is a primitive phocid (true) seal endemic to the tropical Hawaiian Islands. At present, there is a lack of substantive bioacoustic information available for this species, with no formal descriptions of underwater vocalizations and limited data concerning underwater hearing. To address these knowledge gaps, we are working to better understand species-typical auditory capabilities and sound production by thoroughly evaluating a single individual living in human care. A mature male monk seal was trained to perform an auditory go/no-go signal detection task in water. Detection thresholds were measured for narrowband tones across the frequency range of hearing to generate a full underwater audiogram. Additionally, an acoustic recorder was placed in this monk seal's living enclosure for a full year, enabling characterization of his underwater repertoire and seasonal trends in vocal behavior. This study presents the first examination of underwater vocalizations in Hawaiian monk seals, provides insight into the perceptual abilities of this species and the evolution of underwater hearing within the phocid lineage, and enables improved assessments of noise effects on these vulnerable seals. [Work supported by Navy's Living Marine Resources Program.]

11:00

2aAB8. Bottlenose dolphin (*Tursiops truncatus*) vocal modifications in response to spectrally pink background noise. Maria Zapetis (Psych., Univ. of Southern MS, 118 College Dr., #5025, Hattiesburg, MS 39406, maria.zapetis@usm.edu), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), Carolyn E. Schlundt (Government IT Services / US Navy Marine Mammal Program, San Diego, CA), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), and Heidi Lyn (Psych., Univ. of Southern MS, Gulfport, MS)

The increase of oceanic shipping is a global predicament. The resulting proliferation of underwater noise levels is a serious concern for marine mammal welfare, as it has the potential to interfere with the communicative signals of bottlenose dolphins (*Tursiops truncatus*). The Lombard effect and other noise-induced vocal modifications may be employed to compensate for reduced signal-to-noise ratios. This study aimed to determine which vocal modifications dolphins use during experimentally controlled background noise conditions. Three dolphins (ages 30–52) participated in behavioral hearing tests using an adaptive up-down staircase, go/no-go procedure with 15 or 40 kHz tones. Tones decreased by 3 dB increments if a dolphin responded to the tone with a conditioned whistle, and increased by 3 dB if they did not. Dolphins performed this task during ambient noise (control) conditions, as well as three elevated bandpass noise (experimental) conditions: 0.6–5 kHz (115 dB re 1 μPa) and 0.6–10 kHz (115 and 125 dB re 1 μPa). The acoustic parameters of the dolphins' response whistles and victory squeals, such as duration, frequency, amplitude, and response latency, were analyzed and compared between control and noise conditions. These data provide a complement to field studies of odontocete noise-induced vocal modifications in the wild.

11:15

2aAB9. Stealthy moths avoid bats with acoustic camouflage. Thomas R. Neil, Zhiyuan Shen (Life Sci., Univ. of Bristol, Life Sci. Bldg., 24 Tyndall Ave., Bristol BS8 1TQ, United Kingdom, t.r.neil@bristol.ac.uk), Bruce W. Drinkwater (Dept. of Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), Daniel Robert, and Marc W. Holderied (Life Sci., Univ. of Bristol, Bristol, United Kingdom)

Intense predation pressure from echolocating bats has led to the evolution of a host of anti-bat defences in nocturnal moths. Some have evolved ears to detect the ultrasonic biosonar of bats, yet there are many moths that are completely deaf. To enhance their survival chances, deaf moths must instead rely on passive defences. Here, we show that furry morphological specializations give moth bodies and wing joints acoustic stealth by reducing their echoes from bat calls. Using acoustic tomography, echo strength was quantified in the spatial and frequency domains of two deaf moth species that are subject to bat predation and two butterfly species that are not. Thoracic fur determines acoustic camouflage of moths but not butterflies. Thoracic fur provides substantial acoustic stealth at all ecologically relevant ultrasonic frequencies, with fur removal increasing a moth's detection risk by as much as 38%. The thorax fur of moths acts as a lightweight porous sound absorber, facilitating acoustic camouflage and offering a significant survival advantage against bats.

11:30

2aAB10. The auditory attributes of Golden Eagles: Do Golden (*Aquila chrysaetos*) and Bald Eagles (*Haliaeetus leucocephalus*) share the same auditory space? Edward J. Walsh, Peggy B. Nelson, Julia Ponder, Christopher Milliren, Christopher Feist, Jeff Marr, Patrick Redig, and JoAnn McGee (Univ. of Minnesota, S39 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, ewalsh@umn.edu)

According to the U.S. Fish and Wildlife Service (2018), fatalities associated with wind turbine collisions have been reported for more than 200 bird species. Furthermore, based on statistical models of industry growth it has been suggested that as many as 1.4 million bird fatalities/year could be realized if the Department of Energy (DOE) wind energy goals are achieved; i.e., wind energy supplying 20% of total U.S. energy needs by 2030. Although passerine bird fatalities are most commonly reported, raptors that hunt by day, including bald and golden eagles, are the second most frequent casualties of turbine collisions. To address this concern, deterrence protocols designed to discourage eagles from encroaching into wind energy facility air spaces and thereby constrain the degree of risk to which birds are exposed are under investigation. As part of an effort to guide development of acoustic deterrence protocols, we report that the responsive frequency range of golden eagles is similar to that reported for bald eagles; upper and lower frequency limits of hearing are approximately 6.0 and 0.3 kHz, respectively. Suprathreshold response profiles measured in golden eagles exhibit standard features that will be compared with those of bald eagles. [Work supported by DOE grant #DE-EE0007881.]

11:45

2aAB11. Effects of numbers of repetitions, repetition rate, frequency separation, and frequency range on auditory streaming in budgerigars (*Melopsittacus undulatus*). Huaizhen Cai, Laurel A. Screven, and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, 206 Park Hall, Buffalo, NY 14228, huaizhen@buffalo.edu)

Auditory streaming has been widely investigated behaviorally and physiologically in animals. The paradigm used in European starlings (*Sturnus vulgaris*) by MacDougall-Shackleton and colleagues (1998, *J. Acoust. Soc. Amer.*) was used here in budgerigars (*Melopsittacus undulatus*) to measure the effects of different numbers of repetitions of HLH- (high and low tones) triplets, repetition rates, tone frequencies, and frequency separations between the two tones on auditory streaming. Similar to humans, budgerigars subjectively experienced the auditory streaming phenomenon; more repetitions of HLH- triplets, faster repetition rates, and larger frequency separations enhanced the streaming perception. Further, these results were consistent across the two frequency ranges used in this study. When increasing the numbers of HLH- repetitions, it took longer to establish the streaming perception in budgerigars, similar to the buildup phenomenon in humans. These results indicate, for the first time using a behavioral paradigm, that budgerigars experience auditory streaming in a manner similar to humans.

TUESDAY MORNING, 6 NOVEMBER 2018 ESQUIMALT (VCC), 8:00 A.M. TO 11:35 A.M.

Session 2aAO

Acoustical Oceanography, Animal Bioacoustics, Underwater Acoustics, and Signal Processing in Acoustics: Machine Learning and Data Science Approaches in Ocean Acoustics I

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Shima Abadi, Cochair
University of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011

Invited Papers

8:00

2aAO1. Data-driven discovery of dynamics for control. Steven Brunton (Univ. of Washington, 3026 NE 85th St., Seattle, WA 98115, sbrunton@uw.edu)

The ability to discover physical laws and governing equations from data is one of humankind's greatest intellectual achievements. A quantitative understanding of dynamic constraints and balances in nature has facilitated rapid development of knowledge and enabled advanced technology, including aircraft, combustion engines, satellites, and electrical power. There are many more critical data-driven problems, such as understanding cognition from neural recordings, inferring patterns in climate, determining stability of financial markets, predicting and suppressing the spread of disease, and controlling turbulence for greener transportation and energy. With abundant data and elusive laws, data-driven discovery of dynamics will continue to play an increasingly important role in these efforts. This work develops a general framework to discover the governing equations underlying a dynamical system simply from data measurements, leveraging advances in sparsity-promoting techniques and machine learning. The resulting models are parsimonious, balancing model complexity with descriptive ability while avoiding overfitting. The only assumption about the structure of the model is that there are only a few important terms that govern the dynamics, so that the equations are sparse in the space of possible functions. This perspective, combining dynamical systems with machine learning and sparse sensing, is explored with the overarching goal of real-time closed-loop feedback control of complex systems.

8:20

2aAO2. Machine learning applied to broadband sound propagation on the New England Shelf. David P. Knobles (KSA LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept. and the Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mohsen Badiey (Elec. Eng., University of Delaware, Newark, DE)

Addressed is to what degree can variability of an ocean waveguide, such as that associated with the seabed, be inferred by estimating a mapping $y = f(x,\theta)$ for input/category pairs (x,y) and parameterization θ. Feedforward neural network algorithms are employed to learn the optimal parameterization θ approximating f. The mapping includes a chain of functions with hidden layers or functions not directly related to the input. Supervised learning is applied to broadband propagation in a shallow water environment called the New England *Mudpatch* whose acoustic characterization has been studied continuously since 2015. In 2017 about 400 MK-64 Signal Underwater Sound (SUS) and Combustive Sound Sources (CSS) were deployed in a 30×11 km^2 area with horizontal variability of the seabed for the purpose of inferring the statistics of the seabed characterization from the observed statistics of the acoustic field. The goal is to connect θ to physical source and waveguide parameter values. [Work supported by ONR.]

8:40

2aAO3. Estimation of the acoustic environment through machine learning techniques. Oscar A. Viquez, Erin M. Fischell, and Henrik Schmidt (Massachusetts Institute of Technol., 77 Massachusetts Ave., Bldg. 5-204, Cambridge, MA 02139, oviquezr@mit.edu)

The material composition of the bottom of shallow waterways can have significant effects on the corresponding acoustic environment, which autonomous underwater vehicles (AUVs) rely upon for sensing, navigation, and communication. Techniques that require an adequate environmental model are often used onboard AUVs to interpret the sensor data, but such techniques are often sensitive to even small deviations between the model and reality. A proposed approach to reduce this deviation is to use data from local soil and bathymetry surveys to generate environmental model approximations that may be loaded onto the vehicle in advance. During the early stage of deployment, the vehicle uses a K-nearest-neighbor classification approach to compare field calibration measurements with the various models, and select the most suitable solutions for use during the remainder of the active mission. Acoustic field simulations based on the environmental models are produced using normal mode theory as well as wavenumber integration, then compared with field array data. The techniques developed here could be used to facilitate the use of environment-sensitive approaches for detection and tracking during autonomous operations. [Work supported by the Office of Naval Research.]

9:00

2aAO4. Ocean acoustic range estimation in noisy environments using convolutional networks. Emma Reeves Ozanich, Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu), Akshaya Purohit (Elec. and Comput. Eng., Univ. of California San Diego, La Jolla, CA), and Haiqiang Niu (Chinese Acad. of Sci., Inst. of Acoust., Beijing, China)

Machine learning has been able to accurately estimate ocean acoustic source range in real data using passive ship signals (Niu *et al.,* JASA 142, 1176–1188 (2017); Niu *et al.,* JASA 142, EL455–460 (2017)). In this paper, we train a convolution neural network (CNN) to learn range-dependent features directly from the multi-frequency complex pressure field. The CNN network uses 2D inputs to take advantage of spatial relationships between frequency and depth. KRAKEN is used to simulate pressure magnitude and phase received on a vertical array at multiple frequencies, ranges, and signal-to-noise ratios (SNR), generated by adding white Gaussian noise. We show that performance at low SNR can be improved by increasing the size of the training data. The CNN-LSTM (long short-term memory) network, which incorporates sequential learning, is compared to the CNN at varying SNR. The models are tested on real data from the SBCEx17 experiment.

Contributed Papers

9:20

2aAO5. Classification of multiple source depths in a time-varying ocean environment using a convolutional neural network (CNN). Hee-Chun Song (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

Machine learning (ML) is the idea that there are generic algorithms that can tell us something interesting about a set of data without us having to write any custom code specific to the problem. Instead of writing code, we feed data to the generic algorithm and it builds its own logic based on the data. In this paper, ML is applied to classification of multiple (4) sources at various depths in a shallow-water waveguide where the input data are time-varying channel impulse responses (CIRs) received by a vertical array (4 hydrophones) from each of the sources over a period of 22 hours (528 examples). Specifically, the first 16 hours of CIRs are used for training a 4-layered CNN (convolutional neural net) to exploit the structure of CIRs in the two-dimensional time and depth domain, similar to image processing. Our hypothesis is that the dataset collected over a semi-diurnal tidal period (>12 hours) could capture the internal representations or feature vector needed for classification. The trained model is then tested on the next 6 hours of CIRs and achieves 98% accuracy, indicating the potential of ML in underwater applications.

9:35

2aAO6. Estimating the probability density function of transmission loss in an uncertain ocean using machine learning. Brandon M. Lee and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu)

Predicted values of transmission loss (TL) in ocean environments are sensitive to environmental uncertainties. The resulting predicted-TL uncertainty can be quantified via the probability density function (PDF) of TL. Monte Carlo methods can determine the PDF of TL but typically require thousands of field calculations, making them inappropriate for real-time applications. Thus, a variety of alternative techniques based on polynomial chaos, field shifting, modal propagation in ocean waveguides, and spatial variations of TL near the point(s) of interest have been proposed. This presentation describes an innovative approach to estimating the PDF of TL based on nominal TL, ocean environmental parameters, and machine learning. This approach has two main challenges. First, appropriate representations must be found for *ground-truth* PDFs of TL generated from Monte Carlo calculations so that a neural network can be constructed to predict each parameter of the estimated PDF of TL. Four such representations are considered here. And second, a framework must be developed to generate training data for the neural networks. A proposed framework that predicts candidate environments' training utility without computing any PDFs of TL is described. The performance of this approach is analyzed and compared to that of prior techniques. [Sponsored by ONR.]

9:50–10:05 Break

10:05

2aAO7. Using machine learning in ocean noise analysis during marine seismic reflection surveys. Shima Abadi (Eng. and Mathematics, Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, abadi@uw.edu)

Marine seismic reflection surveys use repetitive broadband sound signals to image the structure of the seafloor. High energy sound signals generated by airguns are recorded by single or multiple long horizontal hydrophone arrays towed behind the vessel. Marine seismic reflection surveys raise concern over their effects on marine animals. In particular, there is concern about (1) the impact of the airgun sound power level on marine mammals and (2) the increase in the background noise level caused by the reverberation between shots which may mask marine mammals' vocalizations and degrade the performance of the passive acoustic monitoring systems. Quantifying these effects is a challenging task due to the complexity of local geology, seafloor topography, and uncertainty in water properties. In this study, a machine learning approach is used to predict the airgun sound power level and the reverberation level off the ocean floor. The data utilized in this study are from the COAST (Cascadia Open-Access Seismic Transects) seismic reflection survey conducted with the R/V Marcus Langseth in July 2012. This experiment spanned a wide range of water depths from the continental shelf to deep water. Data was recorded by an 8-km horizontal array with 636 hydrophones sampled at 500 Hz.

10:20

2aAO8. Underwater acoustic target recognition using graph convolutional neural networks. Razi Sabara and Sergio Jesus (Univ. of Algarve, Gambelas Campus, Centro de Investigacao Tecnologica do Algarve, Faro 8005-139, Portugal, razi.sabara@gmail.com)

Motivated by recent progress in signal processing on graphs and convolutional neural networks, we have developed an underwater acoustic target recognition system based on graph convolutional neural networks. We evaluate our framework by application to various real-world datasets and validate its effectiveness. Our experiments demonstrate that the proposed approach achieves high classification accuracy.

10:35

2aAO9. Exploring matrix and tensor factorization for discovering latent structures in large echosounder datasets. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu) and Valentina Staneva (eSci. Inst., Univ. of Washington, Seattle, WA)

Moored autonomous echosounders are increasingly popular as an integrated component in ocean observing systems for measuring biological response to environmental changes. Different from ship-based observations, these large echo datasets often lack concurrent ground truth information from net or optical samples and reliable calibration, both of which are required for conventional analysis routines. However, even though accurate inference of organism abundance is not possible in such scenarios, rich spatio-temporal structures exist in the data and may be exploited to capture variation of migration patterns of different animal groups in the water column. Here, we explore the use of unsupervised matrix and tensor factorization approaches to analyze the long-term echo data sets from the Ocean Observatories Initiative (OOI) network. The data stretch in multiple dimensions, including time, space, and frequency. We restructure the echo data to explore latent structures at different temporal scales using different factorization formulations. Outputs from different methods are compared among one another and with those from conventional echo analysis routines. The results show the importance of augmenting generic factorization formulations with temporal and spatial continuity constraints for biologically meaningful analysis of large echosounder datasets.

10:50

2aAO10. Navigating noise when comparing satellite and acoustic remote sensing data. Carrie C. Wall, Kristopher Karnauskas, Maxwell B. Joseph, Joseph McGlinchy, and Brian R. Johnson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu)

Using a novel combination of NASA satellite products and acoustic remote sensing measurements, we aim to examine the link between the surface expression and vertical structure of ocean productivity and biomass in the California Current System. Water column sonar data collected by the National Marine Fisheries Service for fisheries management have been archived at the NOAA National Centers for Environmental Information. These data are used to evaluate the variability of the marine biogeography. To identify patterns of acoustic reflectance of marine organisms, the data must first be free of noise and seafloor acoustic returns. However, removing the variety of noise present has proved to be a challenge due to need for

manual tuning, a resource prohibitive step, and the size and complexity of the acoustic data. Collaboration with the University of Colorado's Earth Lab has led to the development of machine learning models to automatically remove noise from this large dataset. An analysis of the fully automated, semi-manual, and machine learning quality control processes will be presented. The cleaned acoustic data are then compared to the interannual variability of the distribution of surface chlorophyll concentration and temperature from satellite ocean color measurements.

11:05

2aAO11. Data assimilation for oceanographic and acoustic forecasting. EeShan C. Bhatt and Henrik Schmidt (Mech. Eng., Massachusetts Inst. of Technol., Rm. 5-223, 77 Massachusetts Ave., Cambridge, MA 02139, eesh@mit.edu)

Data assimilation relates observed physical data with dynamic modeling to provide a joint estimate of a field of interest that is bounded by input error. Here, a modular data assimilation framework is introduced for acoustic and sound speed field estimation with a discussion on streamlining separate and varied oceanographic data sources into previously established four-dimensional acoustic modeling. This methodology is applied to acoustic and physical data from an experiment in the Santa Barbara Channel, where passing cargo ships served as sources of opportunity, and a sensitivity analysis for acoustic transmission due to changing sound speed fields is presented. [Work supported by ONR under the Information in Ambient Noise MURI.]

11:20

2aAO12. Source localization using a compact tetrahedral array. 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), Aditi Tripathy (Ocean Eng., Univ. of Rhode Island, Kingston, RI), Makio Tazawa (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jennifer Amaral, Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Ying-Tsong Lin, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

We localized sound sources collected on a compact tetrahedral hydrophone array in a continental shelf environment south of Block Island, Rhode Island. The tetrahedral array of phones, 0.5 m on a side, was deployed to monitor the underwater sound of construction and operation of the first offshore wind farm in the United States. Signals from shipping and marine mammals, including fin whales, humpback whales, and right whales, were detected on the array. Directions of arrival (DOAs) for a number of signals were computed using a time difference of arrival technique. Given the DOAs, ranges were estimated using supervised machine learning techniques outlined by Niu *et al.* (JASA, 2017). The approach was tested using simulated data from Kraken assuming environmental information consistent with this continental shelf environment. Performance on signals from individual ships and marine mammals is presented. Ship localizations are compared to Automated Identification System (AIS) fixes. An error analysis is also presented. [Work supported by the Office of Naval Research and the Bureau of Ocean Energy Management.]

TUESDAY MORNING, 6 NOVEMBER 2018 COLWOOD 1/2 (VCC), 8:00 A.M. TO 11:50 A.M.

Session 2aBAa

Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications I

Guillaume Haiat, Cochair
Multiscale Modeling and Simulation Laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Pierre Belanger, Cochair
Mechanical Engineering, Ecole de technologie superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K/, Canada

Chair's Introduction—8:00

Invited Papers

8:05

2aBAa1. Picosecond opto-acoustics for the remote ultrasonography of single cells. Bertrand Audoin (Université Bordeaux, 341 cours de la Libération, Talence 33405, France, bertrand.audoin@u-bordeaux.fr)

Picosecond ultrasonics (PU) is a technique that offers the possibility to generate and detect acoustic waves remotely with an unequalled frequency bandwidth. After decades of developments and applications for researches in solid-state physics, a vast application field was recently demonstrated for PU in biology and medicine. The technique allows imaging single cells with a lateral resolution limited by optics and with the mechanical properties as the contrast mechanism. The opto-acoustic images can reveal the structure of the cell nucleus, the fine details of the actin network and of the adhesion pattern. In addition to imaging capabilities, quantitative analysis of the interaction of the GHz acoustic waves with the complex cell medium provides information on the cell nanostructure. For instances, the standard deviation of impedance data we measured for single nuclei revealed differences between different cell types arising from the multiplicity of local chromatin conformations within the nucleus. Moreover, the distribution of the cell-substrate interface stiffness

we identified allowed us to separate the contribution of passive and active adhesion processes. The technique allows thus gaining new insights into cell mechano transduction.

8:25

2aBAa2. Cell quake elastography. Guy Cloutier, Pol Grasland-Mongrain (Univ. of Montreal Hospital Res. Ctr., 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada, guy.cloutier@umontreal.ca), Ali Zorgani (Univ. of Lyon, Bron, France), Shoma Nakagawa, Simon Bernard, Lia Gomes Paim, Greg FitzHarris (Univ. of Montreal Hospital Res. Ctr., Montreal, QC, Canada), and Stefan Catheline (Univ. of Lyon, Lyon, France)

The ability to measure the elasticity of a cell provides information about its anatomy, function, and pathological state. Many techniques have been proposed to measure mechanical properties of single cells but most need a model of the cell characteristics and the fixation of the cell on a substrate. Moreover, current measurements take seconds to hours to perform, during which biological processes can modify the cell elasticity. Here, we have developed an alternative technique by applying shear wave elastography to the micrometer scale. Elastic waves were mechanically induced in live mammalian oocytes using a vibrating actuator. These audible frequency waves were observed optically at 205,000 frames per second and tracked with an optical flow algorithm previously developed in the context of ultrasound elastography. Whole-cell elasticity was then reconstructed using an elastography method inspired by the seismology field. Using this approach we showed that the elasticity of mouse oocytes is decreased when the oocyte cytoskeleton is disrupted with cytochalasin B. The technique is fast (less than 1 ms for data acquisition), precise (spatial resolution of a few micrometers), and able to map internal cell structures. The proposed method may represent a tractable option for interrogating biomechanical properties of diverse cell types.

8:45

2aBAa3. Acoustic propagation in the complex sclera for understanding the biomechanical changes associated with myopia. Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riverresearch.org), Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York City, NY), Sally A. McFadden (Vision Sci., Hunter Medical Res. Inst. and School of Psych., Faculty of Sci., Univ. of Newcastle, Newcastle, NSW, Australia), and Quan V. Hoang (Singapore Eye Res. Inst., Singapore National Eye Ctr., DUKE-NUS, Singapore, Singapore)

Myopia affects up to 2.3 billion people and, although minimal levels of myopia are considered a minor inconvenience, high myopia is associated with sight-threatening pathology in 70% of patients. The anatomic changes occurring in the posterior sclera play an important role in myopia progression. Therefore, we investigated the biomechanical properties of the sclera using a well-established myopia model in guinea pigs. We employed two ultrasound-based approaches to better understand and quantify the microstructural changes occurring in the posterior sclera associated with high-myopia development. The first approach applied quantitative-ultrasound (QUS) methods to intact *ex-vivo* eyeballs of myopic and control eyes using an 80-MHz ultrasound transducer. The second approach used a scanning-acoustic-microscopy (SAM) system operating at 250 MHz to form two dimensional maps of acoustic properties of thin sections of the sclera with 7-μm resolution. We tested the hypothesis that the QUS and SAM-based properties are altered in myopia and can provide new contrast mechanisms to quantify the progression and severity of the disease as well as determine where the posterior sclera is most affected. Ultimately these methods will provide novel knowledge about the microstructure of the myopic sclera that can improve monitoring and managing high-myopia patients.

Contributed Paper

9:05

2aBAa4. Analytical solution for elliptic shear standing wave pattern in a bounded transversely isotropic viscoelastic material. Martina Guidetti and Thomas J. Royston (BioEng., Univ. of Illinois at Chicago, 851 South Morgan St., MC 063, Chicago, IL 60607, troyston@uic.edu)

Dynamic elastography methods—based on photonic, ultrasonic, or magnetic resonance imaging—are being developed for quantitatively mapping the shear viscoelastic properties of biological tissues, which are often altered by disease and injury. These diagnostic imaging methods involve analysis of shear wave motion in order to estimate or reconstruct the tissue's shear viscoelastic properties. Most reconstruction methods to date have assumed isotropic tissue properties. But, application to tissues like skeletal muscle and brain white matter with aligned fibrous structure resulting in local transverse isotropic mechanical properties would benefit from analysis that takes into consideration anisotropy. A theoretical approach is developed for the elliptic shear wave pattern observed in transverse isotropic materials subjected to axisymmetric excitation normal to the fiber axis. This approach, utilizing Mathieu functions, is enabled via a transformation to an elliptic coordinate system with isotropic properties and a ratio of minor and major axes matching the ratio of shear wavelengths perpendicular and parallel to the plane of isotropy in the transverse isotropic material. The approach is validated via finite element analysis cases studies. This strategy of coordinate transformation to equivalent isotropic systems could aid in analysis of other anisotropic tissue structures. (Grant support: NIH AR071162.)

Invited Paper

9:20

2aBAa5. Mechanical biomarkers by torsional shear ultrasound for medical diagnosis. Guillermo Rus, Juan M. Melchor, Inas Faris, Antonio Callejas, Miguel Riveiro, Francisca Molina, and Jorge Torres (Structural Mech., Univ. of Granada, Politecnico de Fuentenueva, Granada 18071, Spain, grus@ugr.es)

The WHO estimates that 15 million babies yearly (1 in 10) will be born preterm. Worldwide, complications of preterm births have supplanted pneumonia as the primary cause of child mortality [1]. The biology of cervical ripening that leads to birth is poorly understood, and there is no clinical tool to quantitatively evaluate the cervical biomechanical state, which in words of Feltovich [2]"… likely contributes to the reason the singleton spontaneous preterm birth rate has not changed appreciably in more than 100 years." Towards this problem, we work on enabling new sensor technologies sentient to soft tissue biomechanics, to endow a new class of biomarkers that quantify the mechanical functionality of the cervix, and indeed any soft tissue, ranging pathologies from tumors, atherosclerosis, liver fibrosis to osteoarticular syndromes. Ultrasonic characterization of soft tissue has been developed as a clinical diagnostic tool [3] and evolved through different technologies including our torsional wave principle [4]. Our recent advances covering (a) torsional waves (shear elastic waves that propagate in quasifluids radially and in depth in a curled geometry), (b) sensors (based on a novel arrangement of concentric sandwiches of elements), (c) propagation models and (d) patient testing, are allowing to quantify the mechanical functionality beyond linear parameters: dispersive and nonlinear. [1] WHO, 2012; [2] Feltovich. *AJOG*207(2012):345–354 [3] Barr. *Ultras Quart*28(2012):13–20; [4] Melchor. *Ultrasonics*, 54(2014):1950–1962.

Contributed Papers

9:40

2aBAa6. Shear wave elastography with combined high spatial and temporal frequency excitations. Matthew W. Urban and Piotr Kijanka (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Shear wave elastography is a method for quantitatively measuring the mechanical properties of soft tissues. Shear wave attenuation is high in soft tissues and increases with frequency. In a recent work, high resolution and accuracy can be reached at higher temporal frequencies. To combat the high shear wave attenuation and to obtain higher accuracy and lower variability, we propose to combine acoustic radiation force excitations that have multiple push beams and high temporal frequencies. Multiple push beams were generated with multiple focused beams and multiple intersecting unfocused beams. Measurements made with tonebursts of ultrasound with length 200 μs repeated at 500 and 1000 Hz were compared against a single toneburst. Measurements were made in commercial homogeneous and heterogeneous phantoms. Wave velocity images were reconstructed from the acquired data. The accuracy and variance were evaluated in the homogeneous phantoms. The contrast and accuracy were evaluated in the heterogeneous phantoms. In both types of phantoms, the magnitude at selected frequencies will be compared among the different excitations. The frequency bandwidth for each excitation will be compared. This work highlights that the combination of multiple push beams and excitations with high temporal frequencies can be combined for improved imaging in shear wave elastography.

9:55–10:10 Break

10:10

2aBAa7. Effect of acoustic impedance mismatch between skin and bone on transcranial ultrasound transmission. Shreyank Gupta (Mech. Eng., École de technologie supérieure, 1100 Notre-Dame St. W, Montreal, QC H3C1K3, Canada, shreyank.gupta.1@ens.etsmtl.ca), Guillaume Haïat (CNRS, Laboratoire Modélisation et Simulation Multiéchelle UMR CNRS 8208, Creteil, France), Catherine Laporte (Elec. Eng., École de technologie supérieure, Montreal, QC, Canada), and Pierre Belanger (Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

A major problem with transcranial Doppler (TCD) ultrasound is the poor transmission of ultrasound through the skull bone causing image quality degradation. The reasons for the poor image quality are (1) acoustic impedance mismatch along the wave propagation path and (2) bone frequency-dependent attenuation. Transmission loss due to acoustic impedance mismatch is typically ignored in the literature. Therefore, the objective of this paper is to study the effect of acoustic impedance mismatch on the transmitted energy as a function of frequency. To achieve this, the wave propagation was modelled analytically from the ultrasonic transducer into the brain. The model calculates frequency-dependent transmission coefficient for a given skin and bone thickness combination. The model incorporates both attenuation and acoustic impedance mismatch effects. The model was validated experimentally by comparing with measurements using a bone phantom plate mimicking the acoustic properties of the skull. The average error on the skin thickness between the model and the experiment was less than 6% for a constant bone thickness. Further analysis of the simulation suggests that an optimized excitation frequency can be chosen based on skin and bone thicknesses that would improve ultrasound transmission.

10:25

2aBAa8. Ultrasonic evaluation of anisotropic structure of swine skull. Nagomi Murashima (Life and Medical Sci., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe-shi, Kyoto 610-0321, Japan, ctub1035@mail4.doshisha.ac.jp), Leslie V. Bustamante Diaz (Sci. and Eng., Doshisha Univ., Kyoto, Japan), and Mami Matsukawa (Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

For the ultrasound treatment in brain, such as high intensity focused ultrasound (HIFU), understanding of the complicated ultrasonic propagation in the skull bone is important, because of the anisotropic and heterogeneous character. In this study, we experimentally investigated the ultrasonic wave propagation in the skull sample, considering the anisotropic structure. A cylindrical bone sample was fabricated along the thickness direction of the right occipital bone of a swine (diameter: 11.5 mm, height: 3.0 mm). The ultrasound velocity in the MHz range was measured in the radial direction by rotating the sample (0 degree: sagittal-axis) and changing the measurement position. The structure of the sample was also evaluated by the X-ray micro CT. There are three orthogonal trabecular alignments in bone sample: main, intermediate-shafts (IS) and minor. In the outer layer of the skull, IS were mostly found along the frontal-axis. Ultrasound velocity became higher as the wave propagation direction approached to IS and values were in the range from 2110 to 2790 m/s. In the dipole, ultrasound velocity became higher as the wave propagation direction approached to main-axis and values were in the range from 2080 to 2600 m/s.

10:40

2aBAa9. Validation of single-element ultrasound targeting in the brain for human neuromodulation. Joseph Blackmore (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, joseph.blackmore@wadh.ox.ac.uk), Verena Braun, Christopher Butler (Nuffield Dept. of Clinical NeuroSci., Univ. of Oxford, Oxford, United Kingdom), and Robin Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Non-invasive brain stimulation techniques allow for excitation or inhibition of neural activity via externally applied stimuli through the skull. Recently, ultrasound has been shown to stimulate neurons with better spatial localization than existing electrical stimulation modalities and has subsequently been employed in several human neuromodulation studies. While increased focality provides the potential for modulation of specific brain targets, the overlying skull bone can distort the ultrasound beam leading to shifting and fragmentation of the focus. Consequently, care must be taken to ensure the ultrasound focus is coincident with the intended brain target to confirm the induced effects are due to direct ultrasound activation. Here, we present numerical simulations using k-Wave and experimental measurements demonstrating focusing to different areas of the visual cortex using ex-vivo human skulls with single-element transducers. We show that targeting of many areas can be achieved without the use of lenses or arrays. The sensitivity of transducer angle and location on the resultant pressure field within the skull is assessed along with comparisons between MRI and CT datasets and elastic and fluid-based simulations. Finally, we also explore correction methods for targeting areas of the brain through more complex skull topologies, for example, through the petrous ridge.

Invited Papers

10:55

2aBAa10. Simulation of ultrasound propagation through human skull: Experimental validation and application to treatment planning. Frederic Padilla (Focused Ultrasound Foundation, 151 Cours Albert Thomas, Lyon 69390, France, frederic.padilla@inserm.fr), Raphaël Loyet (LabTAU INSERM U1032, Lyon, France), David Moore (Focused Ultrasound Foundation, Charlottesville, VA), Achour Ouaked (LabTAU INSERM U1032, Lyon, France), John Snell, Matt Eames (Focused Ultrasound Foundation, Charlottesville, VA), SYLVAIN Chatillon (LIST, CEA, Gif sur yvette, France), and Cyril Lafon (LabTAU INSERM U1032, Lyon, France)

Transcranial focused ultrasound (FUS) is a non-invasive therapeutic modality that can be used to treat essential tremor via local thermal ablation of a small spot in the thalamus. Acoustic energy is focused through the skull using a multi-element transducer. Per-element phase delay is derived using individual patient's skull CT to compensate for skull heterogeneity, but MRI thermometry is still required for precise targeting and localization of the focal spot. There exists an opportunity for improved and accurate numerical simulations through skull to improve focal spot positioning and treatment, to allow treatment pre-planning, and to permit parametric studies that will identify important acoustic and thermal properties of human skull bones and tissues. Here, we report on a novel 3D numeric simulation framework, based on the CIVA Healthcare simulation platform, to simulate propagation through skulls and the resulting heating. This simulation platform has been developed to pilot various fast algorithms and to simulate the heating in tissues induced by FUS by solving Pennes' BioHeat Transfer Equation. Simulations of propagation through human skulls with the Insightec 650 kHz clinical system, with 1024 elements, were performed and validated by comparison to experimental acoustic signals acquired with a hydrophone and to MRI thermometry data.

11:15

2aBAa11. Implementation of a dynamic ray tracing method in CIVA HealthCare HIFU simulation platform—Application to the propagation in inhomogeneous and heterogeneous tissues. Sylvain Chatillon (LIST, CEA, Institut CEA LIST CEA Saclay, Bât. Digiteo - 565, Gif sur yvette 91191, France, sylvain.chatillon@cea.fr), Raphaël Loyet, Françoise Chavrier (LabTau, INSERM, Lyon, France), Ayache Bouakaz (U 930, INSERM, Tours, France), Jean-Yves Chapelon (LabTau, INSERM, Lyon, France), Stéphane Leberre, and Pierre Calmon (LIST, CEA, Gif sur yvette, France)

For several years, CEA-LIST has been developing in partnership with INSERM, a HIFU simulation platform, CIVA Healthcare. It provides specific tools for the development and optimization of probes and therapeutic protocols in order to target clinical problematic of specific organs. This platform allows to easily simulate 3D pressure field induced by HIFU, in linear and non-linear regimes, and the corresponding thermal dose in tissues and phantoms. In this communication, we propose a particular focus on the dynamic ray method for linear acoustic propagation. This model enables fast simulations of HIFU propagation in heterogeneous configurations, including soft and hard tissues, made of isotropic and anisotropic medium, with homogeneous or inhomogeneous elastic properties. The contribution of a source point located on the emitting surface is obtained considering the evolution of a pencil of rays centered on the geometrical path linking this source and the observation point. This models takes into account the propagations of bulk waves in each medium and the various reflections and refractions, with and without mode conversion, at each interface. Examples of computation of pressure fields through hard tissues (bones, skull) and in heated soft tissues will be presented. [Work supported by French Nation Research Agency (ANR SATURN -15-CE19-0016).]

Contributed Paper

11:35

2aBAa12. A ray acoustics-based simulation for predicting trans-vertebral ultrasound propagation: Simulation accuracy. Rui Xu and Meaghan A. O'Reilly (Sunnybrook Res. Inst., 2075 Bayview Ave., Apt. 200, Toronto, ON M4N 3M5, Canada, xurui.xu@mail.utoronto.ca)

The blood-spinal cord barrier (BSCB) prevents drug delivery to spinal cord parenchyma. Focused ultrasound can temporarily increase BSCB permeability in small animal models. The human vertebral column distorts ultrasound foci, preventing clinical translation of this technique. Beamforming must be used to correct focal distortion and may be performed using ray acoustics and patient-specific preoperative CT scans. We evaluate the simulation accuracy of ray acoustics, applied to trans-vertebral ultrasound propagation, through comparison with experiment. Trans-vertebral ultrasound propagation was measured for a spherically focused transducer geometrically focused to the centre of individual thoracic vertebral foramen. Simulation pressures were compared to experiment pressures. Simulation error in voxel pressure was evaluated using root-mean-square error, and was similar to error in a water-only case. Average simulation error across all measurements and simulations in maximum pressure location and weighted >50% focal volume location were 2.3 mm and 1.5 mm, respectively. Simulation error is small relative to the dimensions of the transducer focus (4.9 mm full width half maximum), the spinal cord (8–10 mm diameter), and vertebral canal diameter (15–20 mm diameter), suggesting that ray acoustics may be sufficiently accurate for ultrasound beamforming to the vertebral foramen.

TUESDAY MORNING, 6 NOVEMBER 2018 SIDNEY (VCC), 9:15 A.M. TO 11:55 A.M.

Session 2aBAb

Biomedical Acoustics and Physical Acoustics: Targeted Drug Delivery–Acoustic Radiation Force

John S. Allen, Cochair
Mechanical Engineering, University of Hawaii -Manoa, 2540 Dole Street, Holmes Hall 302, Honolulu, HI 96822

Alfred C. Yu, Cochair
University of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada

Chair's Introduction—9:15

Invited Papers

9:20

2aBAb1. Time course of the vasoactive response for sonoreperfusion therapy using long tone burst ultrasound. François Yu (Radiology, Université de Montréal, 900 rue Saint Denis, R11-422, Montréal, QC H2X0A9, Canada, francois.yu@umontreal.ca), Xucai Chen, Jianhui Zhu, Flordeliza Villanueva, and John Pacella (Vascular Medicine Inst., Univ. of Pittsburgh, Pittsburgh, PA)

We have previously shown that sonoreperfusion **(SRP)** using ultrasound **(US)** and microbubble **(MB)** can restore perfusion following microvascular obstruction and is mediated by nitric oxide vasodilation. Here, we compared the use of short (**SP**-10 cycles) and long (**LP**-5000 cycles) pulses for SRP and investigated the vasoactive response time-course and the downstream signaling 4 h post-treatment. In a rat hindlimb, we applied LP or SP (1 MHz,1.5 MPa) with the same number of acoustic cycles for 2 min during MB infusion. Both pulses affected the perfusion rate but with distinct temporal responses. For SP, the flow rate peaked (12.3±3.4 dB/s) at 15 min but returned to baseline **(BL)**(3.2±0.8 dB/s) after 60 min. Conversely, for LP, the flow rate also initially dropped but started to increase at 6 min and peaked (13.0±2.1 dB/s) at 15 min and was still high (8.3±2.0 dB/s) after 4 h compared to BL and SP. This sustained increase in flow rate, which was different from reactive hyperemia, was concordant with 2.5-fold increases in phosphorylated and total eNOS expression (Western blot) compared to untreated tissue, and consistent with a 2-fold increase in eNOS transcription (qPCR). These data support that long pulse SRP therapy increased in muscle perfusion for at least 4 h consistently with increased eNOS transcription and expression.

9:40

2aBAb2. Feasibility of using ultrasound with microbubbles to purify cell lines for immunotherapy application. Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington; Phys. Faculty, Moscow State Univ., Seattle, Washington), Lev Ostrovsky (Dept. of Appl. Mathematics, University of Colorado; Inst. of Appl. Phys., Russian Acad. of Sci., Boulder, CO), Andrew Brayman, John Kucewicz, Brian MacConaghy, and Dino De Raad (Univ. of Washington, Seattle, WA)

Cell-based immunotherapies exploit cell surface antigens to identify and purify cell lines. Fluorescence-based sorters require large sample volumes and are too costly for small labs. Magnetic sorters require enzymatic digestion to remove magnetic particles. We propose to label cells with antibody-conjugated microbubbles (MBs) and selectively sort them using ultrasound. After sorting, the MBs can be removed by a small overpressure. TargeStar-SA microbubbles (MBs) were conjugated to leukemia cells expressing CD7 antigens. Conjugated cell suspensions were placed in a flow with erythrocytes (which lack CD7) and imaged under magnification. Cell motion was quantified with or without ultrasound insonation. The acoustic radiation force (ARF) acting on a cell-MB pair was modeled by assuming the driving force is associated with the MB, and the viscous drag is due to the larger cell. A separate observation of cell-MB rotation caused by the ARF was explained by considering the torque on the cell, where adherent MBs act as point forces. Under insonation, tagged cells were observed to rotate to align with the ARF, and to move in the direction of the ARF. If a tagged cell was adherent to the coverslip, it only rotated. The model for rotation-only behavior fit well with the data. Under flow, tagged leukemia cells were deflected by ultrasound, while erythrocytes were unaffected. These initial studies suggest a new way for isolating and sorting cells. [Funded by LSDF #3292512, RBBR 17-02-00261, NIH P01 DK43881, and NSBRI via NASA NCC 9-58.]

10:00

2aBAb3. Cytomechanical perturbations induced by pulsed ultrasound and acoustic cavitation. Alfred C. Yu (Univ. of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada, alfred.yu@uwaterloo.ca) and Jennifer M. Wan (Univ. of Hong Kong, Pokfulam, Hong Kong)

The therapeutic applicability of ultrasound is perhaps well demonstrated with the advent of high-intensity focused ultrasound (HIFU) that works by rapidly heating tissues to induce ablation. Ultrasound also holds tremendous therapeutic potential at low intensities that are near or below the clinical diagnosis limit (720 mW/cm^2 spatial-peak time-averaged intensity, as set forth by FDA), presumably acting through a mechanical interaction pathway. To establish the therapeutic potential of low-intensity ultrasound, it is fundamentally important to characterize its biophysical interactions with living cells. This talk shall present a series of non-thermal wave-matter interactions of low-intensity pulsed ultrasound. The biophysical interactions between low-intensity ultrasound and living cells will particularly be demonstrated through direct observations acquired using live optical and confocal microscopy tools. We will also examine the scenario where microbubbles are introduced as agents to induce acoustic cavitation. Temporary membrane perforation may be readily achieved in this case (often referred to as sonoporation), and it has been tipped as an emerging paradigm for drug/gene delivery. Epitomizing observations on this process will be shown during the presentation.

10:20–10:35 Break

10:35

2aBAb4. Acoustic fields and forces in drug delivery applications. John S. Allen (Mech. Eng., University of Hawaii -Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

For ultrasound drug delivery applications, the ability to manipulate and move both the carriers and their payloads provides a way of increasing over efficacy. Both primary and secondary acoustic radiation forces have been used though the optimal acoustic forcing parameters and delivery system are subject to on-going investigations. The primary radiation or Bjerknes force occurs from inhomogeneous propagation of the acoustic field and has been used to direct targeted ultrasound contrast agents or particles in vessels towards endothelial cells or pathological targets at the vessel wall. Secondary Bjerknes forces arise from multiple scattering effects between neighboring particles. For corresponding attractive force regimes, greater particle congregation is possible. Overview of theoretical background and predictions of the acoustic radiation forces is given. The formulation of particle translation by primary radiation in the long wavelength limit is discussed with respect to unsteady drag. Pair wise interaction between particles moving in tandem is compared with formulations for along the lines of center. Nonstationary forcing with respect to amplitude modulation or frequency (chirp) alters the attraction and repulsion regimes compared to continuous forcing. Cellular transport for payloads may be enhanced using more optimally tuned pulse sequences.

10:55

2aBAb5. Cloaked microbubbles and acoustic radiation force for ultrasound molecular imaging. Mark Borden (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., Campus box 427, Boulder, CO 80309, mark.borden@colorado.edu), Shashank Sirsi (Mech. Eng., Univ. of Colorado, Dallas, TX), Jason Streeter, and Paul Dayton (Biomedical Eng., UNC/NCSU, Chapel Hill, NC)

In designing targeted contrast agent materials for imaging, the need to present a targeting ligand for recognition and binding by the biomarker is counterbalanced by the need to minimize interactions with plasma components, and to avoid recognition by the immune system. We have designed a microbubble imaging probe for ultrasound molecular imaging that uses a cloaked surface architecture to minimize unwanted interactions and immunogenicity. Here, we examine the utility of this approach for *in vivo* molecular imaging. In accordance with previous results, we showed a threefold increase in circulation persistence through the tumor of a fibrosarcoma model in comparison to controls. The cloaked microbubbles were then activated for targeted adhesion through the application of the primary acoustic radiation force applied specifically to the tumor region. Using a clinical ultrasound scanner, microbubbles were uncloaked, imaged and silenced. Results showed a twofold increase in ultrasound radiation force enhancement of acoustic contrast intensity for

cloaked microbubbles, while no such increase was found for exposed-ligand microbubbles. We conclude that use of cloaked microbubbles for ultrasound molecular imaging bridges the demand for low immunogenicity with the necessity of maintaining targeting efficacy and imaging conspicuity *in vivo*.

11:15

2aBAb6. Shape stability of an encapsulated microbubble undergoing translational motion. Yunqiao Liu (Key Lab. of HydroDynam., Shanghai Jiao Tong Univ., Shanghai, China), Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu), and Qianxi Wang (School of Mathematics, Univ. of Birmingham, Birmingham, United Kingdom)

Encapsulated microbubbles (EMBs) are associated with a variety of important diagnostic and therapeutic medical applications, including sonography, drug delivery, and sonoporation. The nonspherical oscillations, or shape modes, of EMBs strongly affect their stability and acoustic signature, and thus are an important factor in the design and utilization of EMBs. Under acoustic forcing, EMBs often translate with significant velocity, which can excite shape modes, yet few studies have addressed the effect of translation on the shape stability of EMBs. To investigate this phenomenon, an axisymmetric model is developed for the case of small shape oscillations. The exterior fluid is modeled as potential flow using an asymptotic analysis. Viscous effects within the thin boundary layer at the interface are included, owing to the no-slip boundary condition. In-plane stress and bending moment due to the encapsulation are incorporated into the model through the dynamic boundary condition at the interface. The evolution equations for radial oscillation, translation, and shape oscillation of an EMB are derived. These equations are solved numerically to analyze the shape mode stability of an EMB and a gas bubble subject to an acoustic, traveling plane wave. The findings demonstrate the counterintuitive result that translation more readily destabilizes an EMB than an uncoated gas bubble. The main factor responsible for mediating this translation-induced interfacial instability is the no-slip condition at the encapsulating membrane.

11:35

2aBAb7. Insights into the roles of radiation forces in bubble mediated sonothrombolysis. Christopher Acconcia (Univ. of Toronto, Toronto, ON, Canada), Alex Wright (Sunnybrook Res. Inst., Toronto, ON, Canada), Kevin Kiezun, and David Goertz (Univ. of Toronto, S665a, 2075 Bayview Ave., Toronto, ON m4k1x5, Canada, goertz@sri.utoronto.ca)

It is increasingly recognized that radiation forces on bubbles are relevant to a range of therapeutic applications such as sonothrombolysis. In this paper, we discuss work using high speed microscopy to investigate the interactions between ultrasound stimulated bubbles and the boundary of biologically relevant materials. Initial studies with fibrin gels (clots), demonstrated that bubbles could be stimulated to deform, penetrate, and damage the fibrin network. This highlighted two distinct processes where radiation forces play a central role: transporting bubbles from a vessel lumen to a target boundary and facilitating therapeutic effects at the target. To investigate the transport process, experimental data were acquired to assess translational bubble dynamics as a function of size and exposure condition, and this data were used to calibrate a model where both shell and history forces are captured. Experiments and modeling showed that both exposure scheme and flow rate can profoundly influence the number and size distribution of bubbles reaching a target site on a vessel boundary. Once at the boundary, cyclical deformations are induced by the bubble extending well into the material, and these are asymmetric on the forcing and recovery cycles. Deformation, as well as penetration and damage are found to be highly dependent on bubble size and exposure schemes. An improved understanding of these complex processes may facilitate improvements in applications involving large vessels, such as sonothrombolysis.

TUESDAY MORNING, 6 NOVEMBER 2018 CRYSTAL BALLROOM (FE), 9:00 A.M. TO 11:55 A.M.

Session 2aMU

Musical Acoustics and Signal Processing in Acoustics: Modeling Musical Instruments and Effects

Scott H. Hawley, Cochair
Chemistry & Physics, Belmont University, 1900 Belmont Blvd., Nashville, TN 37212

Vasileios Chatziioannou, Cochair
Department of Music Acoustics, University of Music and performing Arts Vienna, Anton-von-Webern-Platz 1, Building M, Vienna 1030, Austria

Chair's Introduction—9:00

Invited Papers

9:05

2aMU1. String synthesis using individually modeled termination scattering filters. Mark Rau, Julius O. Smith, and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 1618 Sand Hill Rd., Apt. 414, Palo Alto, CA 94304, mrau@ccrma.stanford.edu)

In the context of digital waveguide synthesis of string instruments, strings are modeled as pairs of digital waveguides which encounter a scattering junction at each boundary. The string scattering junction at the bridge can be modeled to match the measured driving point admittance at the bridge of a physical instrument, and is used in calculating the radiated sound. The string termination at the nut, or fingering end of the string, is typically modeled with a single scalar attenuation, independent of the fingering. However, driving point admittance measurements are presented confirming that string reflectance is a function of both frequency and fingering. A model is proposed which uses a unique, frequency-dependent string reflection scattering junction for each fingering position. The fingering location scattering junctions are calculated using a modal architecture based on driving point admittance measurements made on an acoustic guitar at the bridge, nut, and frets 1 through 13. These synthesized strings are compared to recordings of the physical instrument, as well as to the widely used scalar attenuation model.

9:25

2aMU2. Power-preserving nonlinear modal coupling, feedback frequency/phase modulation, and the stretched allpass filter. Tamara Smyth and Jennifer S. Hsu (Music, Univ. of California San Diego, 9500 Gilman Dr. MC 0099, La Jolla, CA 92093-0099, trsmyth@ucsd.edu)

Motivated by the sound produced by nonlinear modal coupling in acoustic systems, e.g., changes of sounding frequency in the form of pitch glides/chirps and the generation of harmonics, we model a nonlinear coupling between two modes of oscillation and derive expressions mapping "musical" parameters to the model's "synthesis" parameters. We begin by showing that a unitary power-preserving matrix formulation of the coupled modes leads to a feedback frequency (FM) equation, the instantaneous frequency of which is integrated over time to yield the phase in the corresponding—and preferred—phase modulation (PM) representation. Though the integration is made difficult by the feedback term, an analytic solution is presented by first observing that the real/imaginary parts of the PM equation are equivalent to the real/imaginary parts of the transfer function for a "stretched" allpass filter—one where the delay is not constrained to be a single sample and where the real part produces a periodic comb-like signal, that if made to vary over time (rather than frequency), produces the sounding frequency (a "musical" parameter) resulting from the nonlinear coupling. Ultimately, it is hoped this work may provide computationally efficient models of percussion instruments with rich nonlinear behaviour at little additional computational cost.

9:45

2aMU3. Efficient rendering of saxophone sound by modal synthesis and wave scattering. Esteban Maestre (CAML, McGill Univ., Barcelona, Spain), Gary Scavone (CAML, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, gary@music.mcgill.ca), and Julius O. Smith (CCRMA, Stanford Univ., Stanford, CA)

We present a digital filter design technique for the modeling of impedance and radiation transfer functions as measured from an alto saxophone. For each fingering position, both the input impedance and sound radiation are jointly modeled via a recursive parallel filter structure akin to a modal expansion, with the filter coefficients estimated in the frequency domain by means of constrained pole optimization and least-squares. For modeling the transition between fingering positions, we propose a simple model based on linear interpolation of impedance and radiation models. For efficient sound rendering, the common impedance-radiation model is used to construct a joint reflectance-radiation filter realized as a digital waveguide termination that is interfaced to a reed model based on non-linear scattering.

Contributed Papers

10:05

2aMU4. Physics-based resynthesis of clarinet performance using numerical optimization. Vasileios Chatziioannou, Montserrat Pàmies-Vilà, Sebastian Schmutzhard, and Alex Hofmann (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Inst. 22, Vienna AT1030, Austria, pamies-vila@mdw.ac.at)

The functioning of musical instruments has been an active area of research since the beginning of the last century, using both theoretical approaches and experimental measurements. Regarding wind instruments, the subtleties of embouchure control are of particular interest, since the actions of the player's lips and tongue evade direct measurement. One approach towards analyzing such actions is by extracting relevant information from signals that allow non-intrusive measurements, such as the blowing pressure, the mouthpiece pressure and the reed bending [Pàmies-Vilà *et al.* (2018) Frontiers in Psychology 9: 617]. Alternatively, it is possible to formulate a physical model of the player-instrument interaction and estimate the actions of the player while resynthesizing the oscillations of the instrument [Chatziioannou *et al.* (2012) Acta Acustica united with Acustica 98(4): 629–639]. This work presents such an attempt based on measurements carried out on expert clarinet players. An excerpt of Weber's clarinet concerto No. 2 is performed by the players and resynthesized using a refined physical model taking embouchure effects into account. Using inverse modeling, the underlying model parameters are being tracked in order to extract information on the physical phenomena that take place at the excitation mechanism.

10:20–10:40 Break

10:40

2aMU5. Acoustical impedance of brass instruments: Measurements and modeling for undergraduate physics course. Kurt Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Whitman College, Walla Walla, WA 99362-2067, hoffman@whitman.edu)

We will present recent work developing experimental and modeling exercises for an intermediate physics course focusing on vibrations and waves with applications to acoustics. The measurements described here utilize a simple acoustical impedance measuring system that can be used

to identify the resonant frequencies of a variety of brass instruments. For modeling purposes, we will have the students build approximate brass instruments from PVC tubing and couplers to simplify the shape of the instrument. Despite the abrupt changes in radius of the PVC instrument, adjustment of the cylindrical tube lengths results in a regular pattern of resonances that matches quite well to the first five or six harmonic intervals required in brass instruments. We have developed a Mathematica based program to model the impedance of the PVC brass instruments. The model results can be compared to the measured impedance resonances of the tube. In addition, the resonances can also be measured by lip buzzing the instrument to demonstrate the modification of the resonance frequencies by the vibrating lips at the mouthpiece. We will also discuss the some pedagogical benefits of using acoustical systems as an introduction to solving differential equations and boundary value problems.

10:55

2aMU6. Profiling musical audio processing effects with deep neural networks. Scott H. Hawley, Benjamin L. Colburn (Chemistry & Phys., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu), and Stylianos I. Mimilakis (Fraunhofer Inst. for Digital Media Technol., Ilmenau, Germany)

Deep learning has demonstrated great performance in audio signal processing tasks such as source separation, dereverberation, and synthesis. A challenging effort in deep learning is to devise models that operate directly on the raw waveform signals (i.e., end-to-end). In this talk, we present attempts of an end-to-end task in audio signal processing, that we denote as profiling (i.e., emulation). Our objective is deep learning based profiling of a set of audio production and editing effects, including nonlinear time-dependent effects such as dynamic range compression and time alignment, as well as learning their parameterized controls (e.g., gain, attack time). We present some promising initial results. A consequence of using a data-driven approach to effects modeling is that it allows for the creation of novel audio effects purely via the construction and training on appropriate datasets, without explicitly devising a signal processing algorithm.

11:10

2aMU7. Dynamic characterization of grand and upright piano action. Philip P. Faraci, John A. Case, Jillian M. Farrell, Eoin A. King, and Robert Celmer (Acoust. Program and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfied Ave., West Hartford, CT 06117, faraci@hartford.edu)

A dynamic characterization was performed on a grand piano and an upright (vertical) piano action assembly. The action assembly consists of a key and a system of levers that transfers a force from the key to the hammer. Measurements of the hammer acceleration profile for a given key force were compared to dynamic simulation results from a computer model constructed in SolidWorks. Single-key action assemblies typically used by piano technicians were used for the dynamic measurements and as a guide for the physical dimensions of the computer model. Measurements of hammer velocity were made using a Laser Doppler Vibrometer, which was differentiated to determine the hammer acceleration. The purpose of the study was to establish a relationship between key force and hammer acceleration for the action assemblies and to construct a computer model whose dynamic behavior closely matches the measured profile. Preliminary results between the measured hammer acceleration and the simulated acceleration will be discussed. [Work supported by a University of Hartford CETA Student Engagement Grant.]

11:25

2aMU8. Characterization and modeling of the Erhu. Chris Waltham and Laura Kim (Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada, cew@phas.ubc.ca)

The erhu is a two-string bowed instrument characterized in Chinese organology as a silk instrument (owing to the traditional string material), and by Hornbostel and Sachs as a spike fiddle. The overall length of the instrument is about 80 cm, and the vibrational length of the strings is around half of that. The soundbox is a tube (hexagonal, octagonal, or cylindrical) of about 13 cm length and 9 cm diameter. One end of the tube is covered by a membrane of tensioned python skin and the other is mostly open, usually with a grille. The two strings are stretched over a small bridge positioned centrally on the membrane. In spite of the great difference in soundbox structure between the erhu and a Western violin, the radiated sounds of the two instruments have a lot in common with each other (and with the human voice). Vibrational and acoustic measurements have been made on several erhus: cavity modes, surface velocities and radiated sound. A straightforward coupled-oscillator model has been found to reproduce many of the measurements, and also to allow mechanical characterization of the membrane *in situ*.

11:40

2aMU9. Heuristic finite element models of the Japanese *Koto*. Iran Sanadzadeh and Angela K. Coaldrake (Elder Conservatorium of Music, Univ. of Adelaide, Rm. 8.08, Schulz Bldg., 17 Kintore Ave., Adelaide, SA 5005, Australia, iran.sanadzadeh@adelaide.edu.au)

Modeling a traditional, hand-carved performance-grade Japanese *koto* presents many challenges. A COMSOL Multiphysics finite element model of the instrument, based on CAT scan data, is the most rigorously accurate solution to date. However, a substantially less computationally intensive idealized box model, for example, can provide qualitative assessment of its performance. This paper discusses a COMSOL model based on the physical properties of a solid plank of Australian *paulownia* wood, its validation, and its elaboration into an idealized but unstable hollow box with internal struts and sound holes of varying geometries. It then reports on the development of the next stage model which has 12 cross-sections lofted along a spline to create curvature as the natural progression towards realism. Comparison of these three models and the complex CAT scan model of the *koto* is presented. It shows that the box and lofted models are useful in qualitatively interpreting the results of the complex model. Isotropic models, as used in the literature, were unsuccessful in predicting responses, but using best available approximations for anisotropic elastic constants was helpful. Finally, each stage was able to provide timely feedback to inform the development of the next stage model and guide experimentation.

Session 2aNS

Noise, Physical Acoustics, Structural Acoustics and Vibration, and Architectural Acoustics: Emerging Technologies for Noise Control

Kirill Horoshenkov, Cochair
University of Sheffield, Mappin Street, Sheffield S1 3JD, United Kingdom

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Chair's Introduction—8:00

Invited Papers

8:05

2aNS1. Future trends in noise control technology. J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S. Russell St., West Lafayette, IN 47907-2099, bolton@purdue.edu)

Although acoustics plays an important role in noise control, noise control is arguably more difficult than acoustics owing to the constraints that are usually imposed on noise control solutions by manufacturers. That is, ideally, noise control solutions should occupy no space, weigh nothing, cost nothing, not impact the operation of the device being treated, and remain effective for 20 years without maintenance. Hence, identifying effective noise control solutions is a challenging, multidimensional problem. In this presentation, future trends and opportunities in noise control will be highlighted in four categories: targets, measurement techniques, predictive tools, and noise control methods and materials.

8:25

2aNS2. Noise reduction using metamaterials and metasurfaces. Yun Jing (North Carolina State Univ., 911 Oval Dr., EB III, Campus box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

We are surrounded by noise. Planes, trains, cars, crowds, cooling fans — just about everything is a potential source of noise. Traditional noise abatement methods have a number of limitations and they are particularly ineffective for low frequency noise. As we are becoming increasingly aware of the noise pollution issue, there is a pressing need to develop novel materials for more effective noise reduction. In this talk, I will give an overview of recent development on noise reduction using metamaterials or metasurfaces. I will cover structures such as decorated membrane resonators, hybrid resonators, and coiled Helmholtz resonators.

8:45

2aNS3. Correlating dynamical properties of organic aerogel with increased sound insulation of sandwich wallboard systems. Ning Xiang, Mathew A. Whitney (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu), Sadeq Malakooti (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX), Habel G. Churu (Mech. Eng., LeTourneau Univ., Longview, TX), Nicholas Leventis (Chemistry, Missouri Univ. of Sci. and Technol., Rolla, MO), and Hongbing Lu (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX)

Low thermal conductivity of classical aerogels has long been exploited, while their dynamic properties are yet to be further investigated. This work investigates a new class of acoustic materials based on porous hierarchical three-dimensional assemblies of organic nanoparticles. Primary polymer nanoparticles are formed via chemical reactions in solution and self-assemble to fractal porous secondary particles, which in turn become the fundamental building blocks of larger globular, fibrous or hybrid structures that form the microscopic network of highly porous monolithic materials referred to as polymer-crosslinked (or X-) aerogels. Their low-cost, thin-panel form has found potential applications as constrained damping layers integrated into gypsum wallboards. The standardized chamber-based random-incident tests demonstrate drastically increased sound transmission losses without significantly increasing the board thickness and weight when light-weight X-aerogel panels of less than 1 cm in thickness are integrated into gypsum wallboards in sandwich structure. For better understanding of its excellent effect as constrained damping layers, this paper discusses experimental investigations on broadband dynamic properties of X-aerogels with the intention to correlate their intriguing dynamic properties with increased sound insulations which will have high potentials as emerging building materials for advanced noise and vibration controls and beyond.

9:05

2aNS4. Complex frequency analysis of the scattering matrix for the design of perfect absorbers in reflection and transmission problems. Noé Jiménez (Instituto de Instrumentación para Imagen Molecular (I3M), Consejo Superior de Investigaciones Científicas (CSIC), Edificio 8B, Acceso N, 1ª Planta, Camino de Vera s/n, Valencia 46022, Spain, nojigon@upv.es), Vicente Romero-Garcìa, Vincent Pagneux, and Jean-Philippe Groby (UMR CNR 6613, Laboratoire d'Acoustique de la Université du Mans, Le Mans cedex 9, France)

The design of deep-subwavelength structures for low frequency sound perfect absorption is challenging. Subwavelength perfect absorption implies increasing of the density of states at low frequency while maintaining impedance matching to the surrounding medium. Therefore, the study of the eigenvalues and eigenvectors of the scattering matrix in the complex frequency plane appears extremely powerful to analyze such systems and derive the critical coupling condition. The latter consists in exactly compensating the leakage of the structure by using the intrinsic losses. Two simple structures, the thicknesses of which are 88 times and 40 times smaller than the perfectly absorbed wavelengths, are presented for reflexion and transmission problems, respectively.

9:25

2aNS5. Are nano-fibers an emerging noise control solution? Kirill Horoshenkov, Alistair Hurrell (Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk), and Mohan Jiao (Univ. of Sheffield, Sheffield, South Yorkshire, United Kingdom)

This paper presents experimental evidence that laboratory measurements of the acoustical and related non-acoustical data is not a trivial task. This paper also shows that the prediction of the acoustical properties of this seemingly simple material is far from easy. First, there is a problem of characterising the non-acoustical properties of nano-porous membrane with standard laboratory methods. Second, there is a high uncertainty in the value of the membrane thickness which can lead to a substantial change in its predicted acoustical and related non-acoustical properties. Thirdly, there is an argument whether or not sound propagation in a nano-porous membrane can actually be treated as that in a classical porous media. The paper attempts to respond to these questions and to present new evidence that we may not know well yet the true acoustical properties of these fibres. Therefore, the key question is *Are nano-fibres still an emerging noise control solution?*

9:45

2aNS6. Are aerogels an emerging noise control solution or it is just beefed up fiber glass? Kirill Horoshenkov and Hasina V. Begum (Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk)

Recently, there has been a number of publications which suggested that aerogels can offer high acoustical performance in terms of their ability to absorb or to insulate against sound. It seems that a bulk of these publications report laboratory experiments on the acoustical absorption or transmission loss performance for samples of particular thickness and lateral dimensions. None of these results appear to be rigorously linked to the aerogel's chemistry or micro-structure. This paper attempts to fill in this gap in information by explaining what these materials actually are, why they work acoustically and what is the underpinning physical phenomenon of aerogel. It is shown that the main reason that these materials are used in applications is that they are fire retardant or their elastic modulus is relatively independent of frequency and that they highly hydrophobic. It is also shown that the absorption performance of fibre glass impregnated by aerogel is generally controlled by the fibre diameter of fibreglass and by the density of the composite structure.

10:05–10:20 Break

Contributed Papers

10:20

2aNS7. Is anything radically new in noise control research? Kirill Horoshenkov (Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk) and Mark Swift (Armacell, London, United Kingdom)

This paper attempts to review the state of art of noise control research globally and answer a key question: *Has anything been radically new in noise control research lately?* There is anecdotal evidence that no radical noise control solution has emerged from noise control research since the invention of active noise control by Paul Lueg in 1936 (U.S. Patent 2,043,416). Among passive noise control solutions used for sound absorption the market is dominated by mineral wool and fibreglass (43% in 2017, Global Insulation Market Report, 2024). Technical foams used for acoustic absorption cover pretty much of the rest. Similar situation is for common solutions used for noise and vibration insulation. Their acoustical properties and *in-situ* performance can be predicted within experimental and material parameter uncertainties using a commercial package such as Comsol or semi-empirical models such as by Delany and Bazley. There seem no radical solutions has emerged recently to challenge this status quo. Metamaterials are one promising solution for the future, but they remain largely in the realm of academic laboratories or fancy theories. Therefore, another key question is: *What has true impact of noise control research been so far and what fundamental ideas need to emerge to radically improve this situation?*

10:35

2aNS8. Destructive interferences in resonators created using additive manufacturing. Umberto Berardi (DAS, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca)

The broad absorption of low-frequency noise typically requires the utilization of large thicknesses of sound absorbing materials. As this creates space issues, there is a demand for thin broad low-frequency absorbers. This research aims to develop an acoustic transparent panel for low-frequency broad sound absorption. Destructive sound resonance tubes were created for this scope using modern tools for generating complex geometries in systems of a few centimeters of thickness. The capability of digital fabrication to enable design freedom and to support the production of complex geometries is explored in this paper. Destructive interference means that two interfering sound waves that are in counter-phase, cancel each other. Digital fabrication is shown to play an important role in developing customized sound absorbing components to absorber below 500 Hz, and especially well below 200 Hz. A parametric script using Grasshopper through Rhinoceros to generate automatically the geometric parameters of the 3d resonators is described.

Different materials for 3d printing the resonators are compared. Finally, the challenges of additive manufacturing techniques are finally reported.

10:50

2aNS9. A non-negative artificial neural network configuration for noise predictions. Lucas D. Spies, Sterling McBride, Ricardo Burdisso, and Corina Sandu (Mech. Eng., Virginia Tech, 1706 Boxwood Dr., Apt. D, Blacksburg, VA 24060, lucass19@vt.edu)

Artificial Neural Networks (ANN) provide an accurate relationship between user-defined inputs and outputs for complex problems. Their basic units are called neurons. The computation that takes place in each neuron is a weighted summation of its inputs. These results are then implemented as an input to an activation or transfer function, which in turn defines the final neuron's output. The transfer function configuration is very important when defining the final purpose of the ANN. There are many standard activation functions like pure linear, tangent sigmoid, logistic sigmoid, among others. In this study, an ANN capable of predicting tire pavement interaction noise (TPIN) is constructed. Its outputs correspond to sound pressure in narrowband. Thus, the main requirement is that the outputs must be positive. Several neuron transfer functions were investigated for both hidden and output layers. Finally, a customized hybrid linear-sigmoid transfer function was selected. The final ANN configuration is able to capture typical TPIN spectral content within 400 Hz–1600 Hz. In addition, the error is significantly reduced if compared to those obtained after implementing standard transfer functions.

11:05

2aNS10. Characterization of laptop computer noise and vibration using nearfield acoustic holography. Seonghun Im, Won-Suk Ohm, Inman Jang (School of Mech. Eng., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, seonghun.im@yonsei.ac.kr), and Heoungkil Park (Samsung Electro-Mech., Suwon, South Korea)

Noise from a laptop computer can be a source of annoyance, especially when the computer is used in a quiet environment for a long period of time. Apart from the fan noise, an equally troublesome noise, often characterized as the "buzz," is produced by the vibration of the internal electronics components that transmits through the circuit board and eventually manifests itself as sound radiation. Because the buzz is very low in SPL (but loud enough to cause annoyance in a quiet environment), its measurement requires a nearfield technique that is specialized to a low-intensity sound source. In this talk, we discuss the application of nearfield acoustic holography (NAH) for this purpose, and present an acoustic camera system that can measure and visualize the noise and vibration of a laptop computer. The acoustic hologram measured in the nearfield of a laptop computer is utilized to obtain the two-dimensional maps of the acoustic intensity and the velocity of vibration on the surface of the laptop as well as the total radiated power. The NAH-based acoustic camera system described here proves to be an effective tool for characterization and control of noise and vibration present in mobile electronic devices.

11:20

2aNS11. Noise exposure assessment of a modular construction manufacturing factory. Sanam Dabirian, Joonhee Lee, and Sanghyeok Han (Dept. of Bldg., Civil, and Environ. Eng., Concordia Univ., EV 6.218, 1515 Rue Sainte-Catherine O, Montréal, H3G 2W1, Montreal, QC H3G 2W1, Canada, sanamdabirian@gmail.com)

The National Institute for Occupational Safety and Health (NIOSH) has laid down specific regulations to prevent harmful impacts of noise on workers. However, due to the lack of awareness of irreparable noise damages, the potential of the auditory and non-auditory diseases is increasing. In the modular construction manufacturing, which is increasingly applied as a construction method, construction workers are substantially exposed to high-noise levels. In order to provide a healthier workplace, assessment of the noise exposure is necessary while in this industry has not been fully studied yet. Therefore, this paper presents noise exposure assessment of a modular construction factory. After investigating patterns of the tasks, task-based method (TBM) was applied as a measurement strategy. K-means clustering was used to determine the noise level of each task to calculate the noise exposure levels in different locations of the factory. The results of this paper can provide a useful guideline to manage the noise efficiently in the modular construction sector.

11:35–11:50 Panel Discussion

Session 2aPA

Physical Acoustics and Engineering Acoustics: Novel Approaches to Acoustic and Elastic Wave Experimentation: Concepts, Hardware, and Novel Processing Methods

Michael R. Haberman, Cochair
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Nele Börsing, Cochair
Earth Sciences, ETH Zürich, Sonneggstrasse 5, No. H32, Zürich 8092, Switzerland

Theodor S. Becker, Cochair
Earth Sciences, ETH Zurich, Sonneggstrasse 5, Institute of Geophysics, No. H 41.1, Zürich 8092, Switzerland

Invited Papers

8:15

2aPA1. Time reversal focusing for nondestructive evaluation of cracks. Brian E. Anderson, Sarah M. Young (Dept. of Phys. & Astron., Brigham Young Univ., N145 ESC, N283 ESC, Provo, UT 84602, bea@byu.edu), Marcel Remillieux (EES-17, Geophys. Group, Los Alamos National Lab., Los Alamos, NM), Pierre-yves Le Bas, and Timothy J. Ulrich (Q-6, Detonator Technol., Los Alamos National Lab., Los Alamos, NM)

Time reversal (TR) is a technique that may be used for intentional focusing of wave energy. TR has been developed for crack and defect detection in solid media. Because TR provides a localized focus of elastic energy, it can be used to study the local nonlinear properties at point(s) of interest that are indicative of the presence of cracks and defects. Source transducers may be bonded to the sample under test or to a chaotic cavity that is coupled to a sample under test. A receiver, such as a laser Doppler vibrometer, is placed at a location of interest. Impulse responses are obtained between each source and the receiver and then the impulse responses are reversed in time. The reversed impulse responses are then simultaneously broadcast from the respective source transducers and a focus of energy occurs at the receiver position. Since the largest amplitude is obtained at the receiver position, any amplitude-dependent (nonlinear) vibration that occurs at the receiver position dominates nonlinearities, such as harmonics, that are detected in the signal recorded at the receiver position during focusing. This presentation will review the progress made in the detection of cracks using time reversal and present some recent results.

8:45

2aPA2. Spherical harmonic formulations for passive and active control of acoustic scattering around the sphere. Mihai Orita, Stephen Elliott, and Jordan Cheer (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd, Southampton, Hampshire SO17 1BJ, United Kingdom, mo4g11@soton.ac.uk)

Acoustic cloaking can be implemented using different strategies: passive, active, or a combination of both. The literature on this topic is rich when it comes to the passive methods, specifically on metamaterial designs. However, active and hybrid methods are less well explored. In this work, spherical harmonic decomposition is used to calculate the scattering from a spherical obstacle subjected to an incident, monochromatic plane-wave. Its surface is defined through boundary conditions equivalent to a uniform, locally reacting impedance (Robin condition). Secondary sources represented by point-monopoles on the sphere are used to form an active control system. In the absence of active control, at low-frequencies, the least scattering is achieved if the surface impedance is negative, purely imaginary and has a large absolute value. At high-frequencies, the best performance is realised with a real, positive impedance that is close to that of the exterior propagation medium. When adding active control, only the low-frequency regime shows improvement, given any impedance, and it is most beneficial to enhance for the previous case that exhibited the best passive performance. Thus, a promising hybrid cloaking approach is a combination of: a stiffened-controlled boundary with multi-channel active control at low-frequencies, and a passive, absorptive boundary at high-frequencies.

9:15

2aPA3. Immersive experimentation: A new paradigm for low-frequency acoustic and elastic wave propagation experimentation.
Dirk-Jan van Manen, Theodor S. Becker, Nele Börsing (Geophys., ETH Zurich, Sonneggstrasse 5, Zürich 8092, Switzerland, dirkjan.vanmanen@erdw.ethz.ch), Henrik R. Thomsen, Miguel Moleron (Geophys., ETH Zurich, Zurich, Switzerland), and Johan O. Robertsson (Geophys., ETH Zurich, Zürich, Switzerland)

Immersive experimentation is a new paradigm for wave-based laboratory experimentation aimed at overcoming the laboratory- and sample-size related limitations of conventional approaches and thereby significantly extending downwards the usable frequency range. Using so-called immersive boundary conditions, a physical experimentation domain or elastic rock volume can be immersed in an arbitrary larger numerical domain in such a way that the waves in the physical domain drive the simulation in the numerical domain and vice-versa. Waves propagating towards the edges of the experimentation domain are sensed by dense receiver arrays and extrapolated to/ decomposed at the boundary where they are actively suppressed. Waves incident on the boundary are also extrapolated through the larger numerical domain using pre-computed Green's functions and re-emitted into the experimentation domain, enabling arbitrary order scattering interactions between the laboratory or sample and the numerical domain. We present two laboratories currently under construction at ETH Zurich: WaveLab, an acoustic lab, which implements immersive experimentation in real-time using a low-latency, high-performance acquisition, compute and control system and matrix, an elastic experiment, which implements immersive experimentation using a robotized three-component scanning LDV and novel methods for wavefield separation and injection at an elastic free surface. Implementation, results, and applications are discussed.

Contributed Papers

9:45

2aPA4. Characterisation of complex fluid media by an angle-scanning ultrasound diffractometer.
Mathew J Francis, Melvin J. Holmes, Malcolm J. Povey (School of Food Sci. and Nutrition, Univ. of Leeds, Leeds, West Yorkshire, United Kingdom), Valerie J. Pinfield (Chemical Eng. Dept., Loughborough Univ., UK, Loughborough University, Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk), Artur L. Gower (School of Mathematics, Univ. of Manchester, Manchester, United Kingdom), Jinrui Huang, Derek M. Forrester (Chemical Eng. Dept., Loughborough Univ., UK, Loughborough, Leicestershire, United Kingdom), William J. Parnell, and Ian D. Abrahams (School of Mathematics, Univ. of Manchester, Manchester, United Kingdom)

The characterisation of complex fluid media (particles in liquids) by ultrasound can provide information on the particle size distribution, concentration, and aggregation of the particles. In the Rayleigh scattering regime, systems of liquid particles are dominated by monopole scatter (potentially producing thermal waves), whereas solid particle suspensions are dominated by dipole scatter (producing shear waves); thus the angular dependence of the diffracted field is expected to be different in these cases. Since mode conversion mechanisms are affected by the proximity of particles and thus by the degree of aggregation, the effect of particle type and degree of aggregation is expected to be observed in the angle-dependent signals. We report an experimental investigation of the angle- and frequency-dependent diffracted signal from cylindrical samples (in tubing) of emulsions and suspensions. The experimental results are compared with theoretical predictions using effective homogeneous properties for the complex fluids derived from multiple scattering models. We identify the differences in the diffracted signals for complex media in which the scattering mechanisms are monopole or dipole-dominated.

10:00

2aPA5. Numerical simulations of acoustic cloaking in a real laboratory that deploys immersive boundary conditions.
Nele Börsing, Theodor S. Becker (Geophys., ETH Zürich, Sonneggstrasse 5, NO H32, Zürich 8092, Switzerland, nele.boersing@erdw.ethz.ch), Andrew Curtis (Mathematical GeoSci., Univ. of Edinburgh, Edinburgh, United Kingdom), Dirk-Jan van Manen (Geophys., ETH Zürich, Zürich, Switzerland), Thomas Haag, Christoph Bärlocher (Geophys., ETH Zürich, Zurich, Switzerland), and Johan O. Robertsson (Geophys., ETH Zürich, Zürich, Switzerland)

Immersive boundary conditions (IBCs) act as a wavefield injection method that couples a physical wave propagation experiment to an arbitrary virtual domain. They allow novel, real-time applications for acoustic wave experimentation such as interactions between physical and virtual domains, broadband cloaking, and holography. The implementation of IBCs relies on actively injecting a particular wavefield on the boundary of the physical domain from a dense array of transducers. The injected wavefield must honour the real-world physical and the virtual or computational domains. To calculate the required inputs, a dense array of recording transducers inside the boundary is used from which the future wavefield arriving at the source transducers can be predicted using a discretised Kirchhoff-Helmholtz integral. Whereas IBCs are effectively exact for purely numerical applications, a physical implementation in a laboratory suffers from limitations mainly associated with spatial and temporal discretization issues and the imprint of the electrical devices such as transfer functions or radiation characteristics of transducers. We present a comprehensive numerical sensitivity study of a cloaking experiment in a 2D acoustic waveguide. This defines physical limitations of real-world IBCs, and systematic errors introduced due to subsampling of the recording and injection surfaces and to radiation characteristics of the transducers.

10:15–10:30 Break

10:30

2aPA6. Study of low-hydrostatic pressure dependent elasticity of porous alumina saturated with various gas media.
Ashoka Karunarathne, Josh R. Gladden (Phys. and Astronomy & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu), and Gautam Priyadarshan (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

Poroelasticity describes how the elasticity of porous materials depend on the material properties: porosity, pore saturation medium and its pressure, nature of pore arrangement, and the skeleton material. The primary goal of this study is to investigate the low-hydrostatic pressure dependent elastic properties of porous alumina saturated with different gas media using resonant ultrasound spectroscopy (RUS). RUS is a precise experimental approach for investigating the elastic properties of solid materials which is capable of measurements at different temperatures and hydrostatic pressures. In RUS, the elastic stiffness tensor of crystalline and polycrystalline solids is determined from the vibrational resonance spectra. In this study, we have used commercially available porous alumina ceramic material which has ~40% of porosity and 2 μm of pore diameter. Here we are reporting the variation of elasticity and acoustic attenuation of porous alumina during the low-hydrostatic pressure cycles from 760 torr (1 atm) to 0.1 torr. An experimentally observed discontinuity of elastic stiffness and acoustic attenuation during the pressure region ~150 torr to 0.1 torr will be described by the transition from viscous to molecular flows, quantified by Knudsen number. The effects of saturated gas medium to the elasticity will also be discussed based on the RUS measurements taken during the air, He, and Ar gas saturation.

10:45

2aPA7. Robust downhole ultrasonic pitch-catch measurements of compressional and shear wave velocities. Naoki Sakiyama (Schlumberger K.K., 2-2-1 Fuchinobe, Chuo, Sagamihara 252-0266, Japan, NSakiyama@slb.com), Evgeniya Deger, Hiroshi Hori, Shin'ichi Houshuyama, Atsushi Oshima, and Hiroshi Nakajima (Schlumberger K.K., Sagamihara, Kanagawa, Japan)

Probing acoustic wave velocities of earth formations, both in azimuth and radius, is important when evaluating anisotropy and/or heterogeneity of the formations. A conventional approach for evaluating these complex formations is to employ a sonic logging tool that operates from approximately 0.1 kHz to 20 kHz with an approximately 600 mm spatial resolution. With these standard sonic tools, evaluating formation acoustic wave velocities with an order smaller spatial resolution to detect thin beds and/or azimuthal variation of formation properties is very challenging. To overcome this challenge, this paper discusses an ultrasonic pitch-catch measurement that can be applied to a downhole acoustic tool. Downhole conditions are typically harsh, especially for ultrasonic frequencies. Thus, a robust measurement system is required that maximizes the signal-to-noise ratio. One of the ways to do this is to minimize the distance between the transmitter and the array receiver (and the receiver-receiver spacings of the array). Numerical modeling corroborated by experimental study indicates such a measurement system can primarily detect refracted compressional waves and surface mode waves related to the pseudo-Rayleigh mode. Characteristics of the pseudo-Rayleigh mode measured with such system and a way to estimate shear wave velocity from the measured mode are discussed.

11:00

2aPA8. Ultrasound elastography unveils the shear wave field: Prospects for medical and geophysical imaging. Johannes Aichele, Chadi Zemzemi, Stefan Catheline (Labtau, INSERM, Cours Albert Thomas 152, Lyon 69003, France, johannes.aichele@inserm.fr), and Philippe Roux (Isterre, CNRS, Grenoble, France)

Transient elastography is an ultrasonic imaging method primarily applied in medical imaging. Through correlation of echography images it reconstructs the particle velocities and displacements of the transversal wave-field inside a medium of interest. Elastography measurements are therefore not limited to an acquisition surface. We present developments in processing and experimental design that broaden the application horizon for elastography imaging. First, we propose a multiple source-array for shear wave beamforming that adaptively recovers the impulse response of the medium through time-reversal. This allows for targeted shear wave focusing and thus significantly enhances the SNR compared to single source elastography. Due to the time-reversal robustness the focusing holds true for transmission through a rigid barrier as well, which we show by applying the method to a human skull model. Second, we present a geophysical laboratory experiment. Combining a friction test-bed with ultrasound elastography enables us to recover the effect of rupture dynamics on shear wave propagation. Albeit the use of hydro-gels our densely sampled wave-field perfectly illuminates typical rupture phenomena. These include super-shear fronts, sub-Rayleigh ruptures, and stick-slip behavior. In the future, passive elastography, an equivalent to seismic coda correlation, could be used for monitoring local elasticity during the rupture.

11:15

2aPA9. Acoustic transducer design for active reflection cancellation in a finite volume wave propagation laboratory. Eli Willard, Michael R. Haberman (Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, eliw@utexas.edu), Dirk-Jan van Manen, Johan O. Robertsson, Theodor S. Becker, and Nele Börsing (ETH Zurich, Zürich, Switzerland)

This report describes the design, fabrication, and experimentally obtained electro-acoustic response of an acoustic transducer suite constructed for use in the Wave Propagation Laboratory (WaveLab) at ETH Zürich. WaveLab aims to immerse a physical acoustic experiment within a real-time numerical environment, by implementing exact boundary conditions (EBCs) [J. Acoust. Soc. Am., **134**(6), EL492–EL498, (2013)]. When scale-model ultrasonic experimentation is not possible, a system with EBCs allows for low frequency, reflection-free acoustic measurements in a small physical domain. The physical experiment of the WaveLab facility consists of a water tank measuring only 2 m on a side, while WaveLab's EBCs are realized through a massive computational engine coupled with a dense array of sensing and emitting acoustic transducers. The transducers are used to sense outgoing acoustic waves and to create a reflection-cancelling surface. Criteria for the transducers are discussed in terms of individual and overall system response. The design parameters and associated models include sensitivity, scattering strength, directivity, frequency response, noise floor, and the dynamic range of the system. The transducer designs and models are presented alongside their physical prototypes and experimental results.

11:30

2aPA10. Physical implementation of immersive boundary conditions in acoustic waveguides. Theodor S. Becker, Nele Börsing, Dirk-Jan van Manen (Earth Sci., ETH Zurich, Sonneggstrasse 5, Inst. of Geophys., No. H 41.1, Zürich 8092, Switzerland, theodor.becker@erdw.ethz.ch), Thomas Haag, Christoph Bärlocher (Earth Sci., ETH Zurich, Zurich, Switzerland), and Johan O. Robertsson (Earth Sci., ETH Zurich, Zürich, Switzerland)

The physical implementation of so-called immersive boundary conditions (IBCs) allows the construction of anechoic chambers, where wavefield reflections from the boundary of a physical domain, such as a wave propagation laboratory, are actively suppressed by emitting a secondary wavefield at the domain boundary that destructively interferes with the reflected waves. Moreover, IBCs enable immersive wave propagation experimentation by linking the wave propagation in the physical domain with the propagation in a numerical domain enclosing the physical domain. In this case, IBCs correctly account for all wavefield interactions between both domains, including higher-order scattering. The physical implementation of IBCs is achieved by densely populating the boundary surrounding the physical domain with transducers that enforce the necessary boundary conditions. The signals required at the injection boundary are predicted with the help of a secondary surface of transducers that record the wavefield on a surface slightly inside the physical domain. The recorded wavefield is extrapolated to the boundary by evaluating a Kirchhoff-Helmholtz integral in real-time using an FPGA-enabled data acquisition, computation and control system. Here, we present results of the active suppression of broadband boundary reflections in 1D and 2D acoustic waveguides.

Session 2aPP

Psychological and Physiological Acoustics, Speech Communication, and Musical Acoustics: Music, Speech, and the Brain

Tian Zhao, Cochair
Institute for Learning and Brain Sciences, University of Washington, Box 367988, Seattle, WA 98195

Patricia K. Kuhl, Cochair
Institute for Learning & Brain Sciences, University of Washington, Box 357920, Seattle, WA 98195

Chair's Introduction—9:00

Invited Papers

9:05

2aPP1. Music, speech, and temporal information processing. Tian Zhao and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu)

Music and speech share many acoustic characteristics. Both utilize dynamic acoustic cues (e.g., frequency, time, intensity) to convey information and emotion. We focus on temporal information processing at a slower time scale (e.g., musical timing, speech rhythm) to examine connections between music and speech: (1) transfer effects, (2) neural mechanisms and (3) theoretical frameworks. I will focus on early infancy. We observed a transfer effect from music experience to both music and speech processing in 9-month-old infants (Zhao & Kuhl, 2016). Infants were randomly assigned to either a 12-session music intervention that focused on triple meter (waltz) or a 12-session control group (no music) during the "sensitive period" for phonetic learning. The music intervention group exhibited enhanced neural detection of both mistimed musical beats and speech syllables, quantified by the mismatch response (MMR) measured with magnetoencephalography (MEG). The effects were observed in auditory and prefrontal regions of the cortex. Following the "predictive coding" and "dynamic attending" theories, I will discuss a current two-pronged approach that can extend this research to understand the effects of music on infants' higher-level cognitive skills as well as their early stage sensory encoding of sounds, and implications of these findings for theories of language development.

9:20

2aPP2. Neural entrainment to missing pulse rhythms. Edward W. Large, Charles S. Wasserman, Erika Skoe, and Heather L. Read (Psychol. Sci., Univ. of Connecticut, 406 Babbidge Rd., Unit 1020, Storrs, CT 06269-1020, edward.large@uconn.edu)

Pulse is the perceptual phenomenon in which an individual perceives a steady beat underlying a complex auditory rhythm, as in music. The neural mechanism by which pulse is computed from a complex rhythm is a topic of current debate. Neural Resonance Theory (NRT) predicts that synchronization itself is the neural mechanism of pulse perception and is supported by studies that demonstrate neural entrainment to complex rhythms. Here we report a behavioral and an EEG study of stimulus rhythms that have no spectral energy at the perceived pulse frequency. A dynamical systems model based on NRT predicts that endogenous oscillation will emerge at the "missing" pulse frequency. We observed 1) strong pulse-frequency steady-state evoked potentials (SS-EPs) to isochronous and missing pulse rhythms, but not to a random control; 2) strong coherence between model-predicted SS-EPs and brain responses for all rhythms; 3) differing pulse-frequency activation topographies for missing pulse rhythms versus isochronous and random controls; and 4) differing frequency-following response (FFR) amplitudes for events in missing pulse rhythms versus isochronous and random rhythms. These observations support the theory that the perception of pulse results from the entrainment of emergent population oscillations to stimulus rhythms.

9:35

2aPP3. Distributed neural systems for musical time processing. Takako Fujioka (CCRMA (Ctr. for Comput. Res. in Music and Acoustics), Dept. of Music, Stanford Univ., 660 Lomita Dri, Stanford, CA 94305-8180, takako@ccrma.stanford.edu)

My talk will feature recent MEG and EEG studies for examining time processing, and, together with other findings, discuss how distributed neural systems for musical time work together in managing top-down and bottom-up prediction, ensemble coordination, and learning-induced neuroplasticity. It has been long known that passive listening of metronome-like isochronous stimuli involves auditory cortices for simple prediction of the stimulus interval and detection of changes. Recently, however, more studies are aimed at capturing natural neural dynamics that encode or predict time without such "perturbation" approach. We show that in addition to auditory cortices, various motor-related brain areas contribute to neural oscillatory power modulations in listening to isochronous stimulus, and that beta-band frequency of this modulation may explain benefits of auditory-sensorimotor training in clinical populations with movement

impairments. Moreover, modulation patterns can encode stimulus intervals, metric hierarchies in perception and imagery, as well as anticipation of upcoming tempo change (e.g., *accelerando* and *ritardando*). Listening paradigm is also used to show neuroplastic changes in naïve subjects after short-term piano training in the reward-related systems. Of further interest is frontal lobe's contribution to musical ensemble coordination sensitive to player's role asymmetry and agency (e.g., human vs. computer).

9:50

2aPP4. Finding the beat: An ethnographic approach to beat perception. Alexander K. Khalil (Inst. for Neural Computation, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0523, akhalil@ucsd.edu)

People commonly use words such as "find," "hold," "keep," and "drop" to describe actions related to the musical beat. While such language implies that musical beats exist in an empirically observable, externalized way, musical beats exist only in our perception. This is most clearly evident when people of different cultures perceive different musical beats while listening to the same musical stimuli. At an early age, the ability to discriminate rhythms uncommon to one's surrounding culture declines steeply. However, it has also been shown that rhythmic training may slow or even reverse this decline. This paper explores this process from an ethnographic perspective, observing the way in which American schoolchildren engage with the different temporal dynamics and aesthetics they encounter as they participate in a multicultural music program across several inner-city schools in Southern California.

10:05–10:20 Break

10:20

2aPP5. Neural convergence of rhythm and language: From trait to state. Leyao Yu (Psych., Vanderbilt Univ., Nashville, TN), Brett Myers, and Reyna L. Gordon (Dept. of Otolarygnology, Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, MCE 10267, Nashville, TN 37232, reyna.gordon@vanderbilt.edu)

As a growing literature has shed light on associations between cognitive mechanisms for music and language, an important role of rhythm and timing is emerging for individual differences in language skills in children. In this talk we will address: A) how musical rhythm relates to individual differences in spoken grammar traits in children ages 5 to 8; and B) potential neural synchronization by which an overlap in processing rhythm and language might occur. Results in the current study show strong correlations between musical rhythm perception and spoken grammar production, converging with prior work. In addition, we found that individual differences in neural synchronization to the speech envelope (measured with EEG) during passive listening are associated with performance on prosodic and musical rhythm behavioral task performance. Additional EEG data (during musical rhythm beat perception) has also been collected from $N = 50$ children with typical language development and $N = 13$ children with grammatical impairments (all children had non-verbal IQ in the normal range); data analysis is underway. In light of these studies and others showing that rhythmic listening can impact grammatical task performance, we will discuss other ongoing translational work that takes first steps towards exploring a causal influence of rhythm on grammar states.

10:35

2aPP6. Redundancy in the speech signal helps amusics perceive prosody. Adam Tierney (Birkbeck College, Birkbeck College, London WC1E 7HX, United Kingdom, a.tierney@bbk.ac.uk)

Speech is information dense, rapidly conveying segmental, semantic, syntactic, and prosodic information through the manipulation of a small handful of acoustic cues. As a result, impaired perception of one of these cues could impede acquisition of linguistic structure at multiple levels. For example, impaired perception of temporal patterns has been linked to developmental language delays. However, auditory impairments do not necessarily lead to language delays, suggesting that some people are able to compensate for difficulties with sound perception. This may be due to the perceptual redundancy of speech—the fact that multiple auditory dimensions often convey the same linguistic information. We tested this hypothesis by examining prosody perception in individuals with amusia, i.e., severe difficulties with the perception of musical melodies. First, we find that amusics show decreased functional connectivity between auditory and motor cortices during speech perception, suggesting that amusia is a domain-general deficit in pitch perception stemming from decreased auditory-motor connectivity. Second, we find that amusics compensate for their impairment by relying upon preserved perceptual abilities, suggesting that redundancy makes speech somewhat robust to individual differences in perceptual skill.

Contributed Papers

10:50

2aPP7. The flow of language: Respiration and the rhythmic motor control of speech. Alexis D. MacIntyre and Sophie K. Scott (Inst. of Cognit. Neurosci., Univ. College London, 17 Queen Square, London WC1N 3AR, United Kingdom, a.macintyre.17@ucl.ac.uk)

Speech rhythm can be described as the temporal patterning by which sequences of vocalic and gestural actions unfold, within and between interlocutors. Despite efforts to quantify and model speech rhythm across languages, it remains a scientifically enigmatic aspect of prosody. For example, the existence and/or form of a basic speech rhythmic unit is hotly contested. One challenge is that the primary means of speech rhythm research has been the analysis of the acoustic signal. As speech is multimodal and motoric, investigations of speech rhythm will likely benefit from a greater range of complementary measures, including physiological recordings. The current experiment explores respiratory effort as a contributing factor in speech rhythm production. Undergoing simultaneous inductive plethysmography and acoustic recording, participants produce speech by reading novel prosaic texts aloud in single and dyadic conditions. Speech-associated breathing patterns are analysed in conjunction with traditional acoustic-phonological approaches to speech rhythm. It is hypothesized that, like spoken utterances, speech breathing is subject to rhythmic constraints. Preliminary results are therefore interpreted with a focus on the temporal relationship between inhalation events and previously proposed phonological rhythmic units (e.g., inter-stress, p-centre, syllable), underscoring breathing as a necessary, yet often overlooked, component in speech rhythm planning and production.

11:05

2aPP8. Improving lyrical intelligibility in live music and concert settings: Evaluating the application of alternative sound reinforcement techniques. Jared H. Koshiol (Integrative Studies, Northern Kentucky Univ., Nunn Dr., Newport, KY 41099, koshiolj1@nku.edu) and Greg DeBlasio (Commun., Northern Kentucky Univ., Highland Heights, KY)

Thirteen individuals were asked to listen to eight short passages of music from two songs containing sung lyrics. They compared the music as it was presented at 94dBA through three different systems; one control emulating a standard live music performance, another with the vocal channel assisted by digital effects (Compression, an Adaptive Limiter, EQ and a Harmonic Exciter), and a third where vocals were removed from the left and right speakers and instead played through a single center speaker of the same make while instruments remained in the left and right speakers. This third design is similar to common mixing of 5.1 theatrical audio. Participants indicated their perception of lyrical intelligibility and pleasurability of the sound for each passage and were asked to ignore artistic choices. Responses indicated that the use of effects processing for vocals somewhat improved lyrical intelligibility and pleasurability for one song, while slightly decreasing these values for the other. The center speaker system for vocals greatly improved (in one case almost doubled) listeners' ability to perceive the lyrics as well as their overall enjoyment of the sound. This improvement is likely linked to spatial hearing; a musical extension of the *Cocktail party effect*.

11:20

2aPP9. Vowel content influences relative pitch perception in vocal melodies: A comparison of models based on brightness vs. intrinsic pitch of vowels. Frank A. Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M4L 3T4, Canada, russo@ryerson.ca) and Dominique Vuvan (Psych., Skidmore College, Saratoga Springs, NY)

Past research involving real and synthesized instrumental timbres has found that note-to-note changes in brightness can influence perception of interval size. Changes that are congruent with changes in pitch led to an expansion, whereas changes that are incongruent lead to a contraction. In the case of singing, the brightness of individual notes (as measured by the spectral centroid) will vary as a function of vowel content. In a recent study, we investigated whether note-to-note changes in the brightness of sung notes were capable of influencing the perception of interval size. While results were consistent with past work on instrumental timbres, we were not able to completely rule out an alternative explanation concerning a perceptual correction for the intrinsic pitch of vowels (e.g., f0 of /i/ tends to be produced higher than /a/). In the present study, we created 288 unique note pairs that varied with regard to absolute change in f0 as well as vowel content. Vowels were sampled from across the vowel space, which allowed us to generate unique predictors for change in brightness (spectral centroid) and changes in intrinsic pitch (F2). Regression analyses will compare the effectiveness of competing models.

11:35

2aPP10. Extended high-frequency hearing enables better talker and singer head orientation detection. Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 75 Francis St., Boston, MA 02115, monson@illinois.edu)

Why is the human auditory brain endowed with sensitivity to acoustical energy at frequencies beyond 8 kHz? While this "treble" range is known to be important for music quality, we hypothesize that extended high-frequency hearing capability has been obtained and retained, in part, due to its utility for the detection and perception of conspecific vocalizations (*i.e.*, human speech and voice). More specifically, we hypothesized that access to the highest frequencies enables better detection of a vocalizer's head orientation. To test this hypothesis, we assessed human listeners' ability to detect changes in talker and singer head orientations, and the effect of removing access to the highest frequencies on that ability. Detection of head orientation was significantly impaired by low-pass filtering at 8 and even 10 kHz. One implication is that extended high-frequency hearing loss might impair a listener's ability to detect whether a talker is facing the listener—a skill presumably useful for "cocktail party" listening.

11:50–12:00 Panel Discussion

Session 2aSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustic Metamaterials

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

Contributed Papers

9:00

2aSA1. Numerical investigation of non-reciprocal wave propagation in mechanically-modulated elastic metamaterials. Benjamin M. Goldsberry (Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, Austin, TX)

Acoustic systems that break reciprocity have recently received great attention due to their potential to increase control over wave propagation and to have far-reaching impacts on engineering applications, including more efficient acoustic communication devices and vibration isolation. One means to break reciprocity is by spatiotemporal modulation of material properties, typically via electromagnetic actuation. In the present work, tunable mechanical metamaterials are investigated as a platform to achieve non-reciprocity, where effective dynamic property modulations are induced by an external pre-strain of a two-dimensional cellular material. These elastic metamaterials are studied using a nonlinear finite element model, which is suitable for unit cells having complex geometry and undergoing large deformation caused by the external pre-strain [Goldsberry *et al.*, J. Appl. Phys., **123**, 091711 (2018)]. A small-on-large approximation is used to analyze linear elastic, non-reciprocal wave propagation in the presence of a slowly-varying pre-strain. Results on non-reciprocal wave propagation in negative stiffness honeycombs, a structure exhibiting large stiffness modulations due to the presence of a mechanical instability [Correa *et al.*, Rapid Prototyping J., **21**(2), 193–200, (2015)], are shown as a case example. [Work supported by NSF.]

9:15

2aSA2. Dual-frequency sound-absorbing metasurface based on visco-thermal effects with frequency dependence. Wonju Jeon and Hyeonbin Ryoo (KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, wonju.jeon@kaist.ac.kr)

We propose an acoustic metasurface with near-perfect absorption at two frequencies and design it by using two-dimensional periodic array of four Helmholtz resonators in two types. By considering how fluid viscosity affects acoustic energy dissipation in the narrow necks of the Helmholtz resonators, we obtain effective complex-valued material properties that depend on frequency and on the geometrical parameters of the resonators. We furthermore derive the effective acoustic impedance of the metasurface from the effective material properties and calculate the absorption spectra from the theoretical model, which we compare with the spectra obtained from a finite-element simulation. As a practical application of the theoretical model, we derive empirical formulas for the geometrical parameters of a metasurface that would yield perfect absorption at a given frequency. Whereas previous works on metasurfaces based on Helmholtz resonators aimed to absorb sound at single frequencies, we use optimization to design a metasurface composed of four different Helmholtz resonators to absorb sound at two target frequencies.

9:30

2aSA3. A low frequency underwater meta structure composed by helix metal and viscoelastic damping rubber. Ruihao Zhang (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an Shaanxi, 710072, P.R. China, Xi 710072, China, zhangruihao@mail.nwpu.edu.cn)

Many underwater acoustic absorption materials have been proposed, but good low frequency acoustic absorption remains a challenge. We report an underwater metastructure with excellent acoustic absorption that is constructed by perforating helixes of metal into viscoelastic damping rubber. Finite element analysis shows that this metastructure can achieve an acoustic absorption coefficient of 0.75 at about 100 Hz with a thickness of only 1/68 wavelength at the same frequency. Compared with a homogeneous viscoelastic rubber material, the addition of the helix structure improves the acoustic absorption ability in the range of 0–1,000 Hz. An analysis of vibration displacement maps indicates that waveform transformation, multiple scattering, and reflection energy dissipation mechanisms are the critical factors affecting the absorption performance. Different geometries and materials can adjust the sound absorption characteristics below 1,000 Hz. The proposed metastructure has the advantages of high acoustic absorption ability, broader frequency bandwidth, and regular geometry, providing more possibilities for underwater acoustic manipulation applications.

9:45

2aSA4. Efficient emission of directional sound waves by using subwavelength meta-cavities. Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu), Xudong Fan (Univ. of MS, University, MS), Jiajun Zhao (GOWell Int., LLC, Austin, TX), Bin Liang, Jianchun Cheng (Nanjing Univ., Nanjing, Jiangsu, China), Ying Wu (King Abdullah Univ. of Sci. and Technol., Jeddah, Saudi Arabia), and Maryam Landi (Univ. of MS, Oxford, MS)

Efficient emission of directional sound waves is critical in imaging and communication, yet is held back by the inefficient emission at low frequencies, especially for a small source. A subwavelength enclosure with degenerate Mie resonances was implemented to experimentally enhance the sound power emitted to the far field where the radiation directivity pattern is preserved [L. Maryam *et al.*, Physical Review Letters 120 (11), 114301, 2018]. When considering the efficient emission of directional sound waves, a subwavelength meta-cavity of hybrid resonances can be used to convert the monopole sources to multipole emission [X. Fan *et al.*, Physical Review Applied 9 (3), 034035, 2018] or Mie resonances with spatial asymmetry can be used to even emit directional sound beams [J. Zhao *et al.*, Scientific reports 8 (1), 1018, 2018]. The work offers a practical path toward applications that demand miniaturization of speakers for efficient emission.

10:00

2aSA5. Finite element modeling of fluid-saturated metallic foams from micro-computed tomography. Mark J. Cops, James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, mcops@bu.edu), Elizabeth A. Magliula, and David J. Bamford (Naval Undersea Warfare Ctr. Div., Newport, RI)

Open-cell metallic foams are high stiffness-to-weight cellular materials whose microstructure allows for saturation of viscous liquid. Such a composite has advantages for underwater sound absorption over traditional rubbers due to minimal compression from hydrostatic pressure, composite tunability, and potential for specific gravity less than one. Semi-phenomenological and hybrid numerical models have been shown to predict sound absorption performance of metallic foams, however difficulty arises in determining parameters for the numerical models such as tortuosity, viscous characteristic length, thermal characteristic length, and flow resistivity. Such models also assume that the porous frame is rigid, an assumption valid for only a limited frequency range. In this presentation, finite element models of fluid-saturated metallic foams are created from micro-computed tomography scans and analyzed to determine sound absorption performance. The advantage of this method is that the entire foam microstructure and surrounding fluid can be accurately modeled through a finite element mesh. In addition, experimental measurement of model parameters is not required and the rigid frame assumption can be removed.

10:15–10:15 Break

10:15

2aSA6. An investigation into the direct control of the effective material properties of a fluid medium using active noise control. Joe Tan (ISVR, Univ. of Southampton, Highfield Campus, University Rd., Southampton SO17 1BJ, United Kingdom, j.tan@soton.ac.uk), Jordan Cheer, and Stephen Daley (ISVR, Univ. of Southampton, Southampton, Hampshire, United Kingdom)

Acoustic metamaterials are subwavelength-engineered structures that can exhibit behaviour not seen in conventional materials. The effective material properties for acoustic metamaterials are the bulk modulus and density. Negative effective material properties can create band gaps, where wave propagation is forbidden. Hence, acoustic metamaterials can achieve high levels of noise control performance. Active control has been combined with passive acoustic metamaterials to enhance the level of attenuation or broaden the bandwidth of the band gap. However, the width of the band gaps for these materials are still limited by the use of resonators. This study will investigate whether active control can be employed to directly minimise the effective material properties, which requires an optimization procedure. To begin to understand this optimization problem, the effective material properties have been calculated as a function of the real and imaginary parts of the control source strength at various frequencies for monopole and dipole sources. The resulting error surfaces demonstrate that the two configurations could achieve negative effective material properties; however, the cost functions are not always convex without constraints. Therefore, to adaptively manipulate the effective material properties using an active control system would require the development of an advanced optimization procedure.

Invited Papers

10:30

2aSA7. Acoustic-like behavior in periodic metal structures. Andrew Norris and Xiaoshi Su (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Pentamode materials are of interest for acoustic metamaterials because of their property of having only one elastic mode, similar to the hydrostatic response of an acoustic fluid. This static property does not take into account dynamic frequency dependent effects related to the necessary spatial inhomogeneity in the structural realization. In this talk, we argue that pure pentamode behavior is not desirable, and instead AMM users should focus on the one-wave dynamic response. Experience with metal structures designed to provide acoustic properties close to those of water using metal structures, e.g., Su *et al.* (JASA 2017, doi: 10.1121/1.498519), shows that the most useful response is in a one-wave band gap in which shear waves are non-propagating. Surprisingly, the one-wave region is generally broadband relative to the cutoff frequency of the shear waves at low frequencies, and is non-dispersive, with constant phase speed over the one-wave gap. The talk will attempt to explain these surprising and unexpected features using numerical and analytical results. The latter include an approximation of the dynamic effective properties of 2D and 3D metallic structures displaying broadband one-wave regions using simplistic models.

10:50

2aSA8. Magnetoactive acoustic metamaterials. Qiming Wang (Univ. of Southern California, 920 Downey Way BHE 222, Los Angeles, CA 90004, qimingw@usc.edu)

Acoustic metamaterials with negative constitutive parameters (modulus and/or mass density) have shown great potential in diverse applications ranging from sonic cloaking, abnormal refraction and superlensing, to noise canceling. In conventional acoustic metamaterials, the negative constitutive parameters are engineered via tailored structures with fixed geometries; therefore, the relationships between constitutive parameters and acoustic frequencies are typically fixed to form a 2D phase space once the structures are fabricated. Here, by means of a model system of magnetoactive lattice structures, stimuli-responsive acoustic metamaterials are demonstrated to be able to extend the 2D phase space to 3D through rapidly and repeatedly switching signs of constitutive parameters with remote magnetic fields. It is shown for the first time that effective modulus can be reversibly switched between positive and negative within controlled frequency regimes through lattice buckling modulated by theoretically predicted magnetic fields. The magnetically triggered negative-modulus and cavity-induced negative density are integrated to achieve flexible switching between single-negative and double-negative. This strategy opens promising avenues for remote, rapid, and reversible modulation of acoustic transportation, refraction, imaging, and focusing in subwavelength regimes.

Contributed Papers

11:10

2aSA9. Vibration suppressions using a dynamic absorber embedded with acoustic black hole features. Tong Zhou and Li Cheng (Dept. of Mech. Eng., Hong Kong Polytechnic Univ., Hong Kong 852, Hong Kong, mmlcheng@polyu.edu.hk)

By capitalizing on the acoustic black hole (ABH) phenomena, an ABH-featured vibration absorber is proposed for the broadband vibration suppressions of a primary structure. As an add-on device to be attached to the primary structure, the proposed absorber embraces the principles of both dynamic vibration absorbers and waveguide absorbers. Its design and implementation avoids the tedious parameter tuning, thus showing robustness to accommodate structural variations in the primary structure. Using a beam as a benchmark example, both numerical simulations and experiments show that multiple resonances of the primary structure can be significantly reduced, and the same absorber can be used for different primary systems. Analyses reveal the existence of three types of vibration reduction mechanisms, manifested differently and dominated by different physical process, *i.e.,*structural interaction, damping enhancement, and their combination. Comparisons with a conventional uniform beam absorber shows that the superiority of the proposed absorber is attributed to its ABH-specific features exemplified by the enriched system dynamics and the enhanced broadband damping.

11:25

2aSA10. Acoustic carpet cloaks based on anomalous reflectors. Marc Martí-Sabaté (Dept. of Phys., Universitat Jaume I, Castellón de la Plana, Spain), Yabin Jin (I2M, Univ. of Bordeaux, Bordeaux, France), and Daniel Torrent (Dept. of Phys., Universitat Jaume I, Av. de Vicent Sos Baynat, s/n, Castellon de la Plana, Castellón 12071, Spain, dtorrent@uji.es)

We present an acoustic carpet cloak based on acoustic anomalous reflectors. The cloak consists on a engineered diffraction grating, designed in such a way that a wave incident with a given angle is "retroreflected," so that a pyramidal surface reflects a normally incident wave as if it were a flat surface. Different geometries and angles of the surface are tested, showing that the surface can have an arbitrary shape, as long as the retrorreflection conditions be satisfied locally. Finally, the experimental characterization of these cloaks is discussed.

TUESDAY MORNING, 6 NOVEMBER 2018 SALON A (VCC), 8:15 A.M. TO 11:50 A.M.

Session 2aSC

Speech Communication, Biomedical Acoustics, and Signal Processing in Acoustics: Recent Advances in Experimental, Computational, and Clinical Research in Voice Production and Perception

Zhaoyan Zhang, Cochair
UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Center, Los Angeles, CA 90095

Michael Doellinger, Cochair
University Hospital Erlangen, Erlangen, Germany

Chair's Introduction—8:15

Invited Papers

8:20

2aSC1. Reconsidering the nature of voice. Jody E. Kreiman (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Voices play essential roles in human experience, and carry many kinds of meaning. At present, the study of voice is consistently tied to the speech chain model, so that voice production, acoustics, and perception are treated as separate, independently studied stages. Such studies provide no insight into how meaning in voice is constructed, and create a situation where voice comprises something different to everyone who studies it. This paper presents empirical and theoretical arguments leading to the conclusion that voice must be viewed as a single integral process, and not as a concatenation of separate stages. We argue that voice production, acoustics, aeroacoustics, biomechanics, perception, and all the rest, are conceptually inseparable, and that none can be understood without knowing how its relationship to the others contributes to its structure and function. We submit that development of such a comprehensive theory of voice should be the primary goal of voice research, whatever the discipline. Such a theory would form a basis on which to build realistic explanations of how voice functions socially and communicatively, how it has evolved, and how it conveys meanings of all sorts, leading to a deeper understanding of what it means to be human and to communicate.

8:40

2aSC2. Computational medicine in voice research. Nicole Y. Li-Jessen (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8/F, Montreal, QC H3A1G1, Canada, nicole.li@mcgill.ca)

Computer models have been widely used to numerically simulate the physical dynamics of vocal fold oscillation in phonation. Computer simulations have allowed the exploration of a much wider parameter space and a much longer time scale than what would be very costly or sometimes impossible with animal and human models. In the past 10 years, our team uses a combined *in vivo, in vitro,* and *in silico* approach to understand the biological mechanism underlying the formation of vocal lesions. The ultimate goal is to identify key biological and physical factors associated with the pathogenesis of and recovery from vocal fold injury that will be useful for personalized voice restoration. Tissue injury and repair is a complex process that involves multiple cells types and molecular signals. We developed 3D agent-based models at a physiological scale to simulate and visualize the cellular and molecular response to vocal injury and treatment. Significant progress has made to improve the model's computational speed and visualization features using our hybrid CPU-GPU computing platform. Gaps however still exist between the model and the reality. In this talk, I will present what we have achieved so far as well as what challenges we are facing in biological computing.

9:00

2aSC3. Studying vocal fold non-stationary behavior during connected speech using high-speed videoendoscopy. Maryam Naghibolhosseini, Dimitar Deliyski (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, naghib@msu.edu), Stephanie Zacharias (Laryngotracheal Regeneration Lab, Mayo Clinic, Phoenix, AZ), Alessandro de Alarcon (Div. of Pediatric Otolaryngol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and Robert Orlikoff (College of Allied Health Sci., East Carolina Univ., Greenville, NC)

Studying voice production during running speech can provide new knowledge about the mechanisms of voice production with and without disorder. Laryngeal high-speed videoendoscopy (HSV) systems are powerful tools for studying laryngeal function and, if coupled with flexible fiberoptic endoscopes, they can provide unique possibilities to measure vocal fold vibration with high temporal resolution during connected speech. Hence, we can measure the non-stationary behaviors of the vocal folds, such as the glottal attack and offset times in running speech. In this study, a custom-built flexible fiberoptic HSV system was used to record a "Rainbow Passage" production from a vocally normal female. Automated temporal and spatial segmentation algorithms were developed to determine the time stamps of the vibrating vocal folds and the edges of the vocal folds during phonation. The glottal attack time and offset times were then measured from the temporally and spatially segmented HSV images. The amplification ratio was computed during the phonation onset and the damping ratio was calculated at the offset of sustained portion of phonation. These measures can be used to describe the laryngeal mechanisms of voice production in connected speech.

9:20

2aSC4. High-fidelity image-based computer modeling of voice production—From muscle contraction to flow-structure-acoustics interaction. Qian Xue, Xudong zheng, Weili Jiang, Ngoc Pham, and Biao Geng (Mech. Eng., Univ. of Maine, 213 Boardman Hall, Orono, ME 04469, qian.xue@maine.edu)

This study aims to develop a high-fidelity computer model of voice production which can employ the image-based realistic three-dimensional geometries of larynges and simulate the complex mechanical process of voice production and control, from muscle contraction to flow-structure-acoustics interactions. Such a model will advance our understanding of the relationship between muscle contraction, vocal fold posturing, vocal fold vibration, and final voice outcome, which has important clinical implications for voice management, training, and treatment. The key components of the model include a sharp-interface-immersed-boundary-method based impressible flow model, a finite-element method based nonlinear structural dynamics model and a hydrodynamics/acoustics splitting method based acoustics model. A Hill-based contractile model is coupled in the finite element analysis to capture the active response of vocal fold tissues, and a fiber-reinforced model is employed for the passive response. A series of validations have been performed. The coupled Hill-based model and fiber-reinforced model demonstrated a good agreement with literature experimental data for dynamics, concurrent tissue stimulation, and stretching. The flow-structure-acoustics interaction model was validated with excised canine experiments using realistic geometric and material properties. The simulations showed a good agreement on the fundamental frequency, vocal fold maximum divergent angle, flow rate, and intraglottal velocity and pressure fields.

9:40

2aSC5. Bayesian inference of tissue properties from glottal area waveforms using a 2D finite element model. Paul J. Hadwin and Sean D. Peterson (Dept. of Mech. and Mechatronics Eng., Univ. of Waterloo, 200 University Ave. W, Waterloo, ON N2L 3G1, Canada, pjhadwin@uwaterloo.ca)

Bayesian inference has recently been demonstrated to be effective in estimating stationary and non-stationary reduced-order vocal fold model parameters, along with the associated levels of uncertainty, from simulated glottal area waveform measures (Hadwin *et al*., 2016, 2017). In these studies, the fitting model was a three mass body-cover model with two degrees of freedom in the cover layer. While demonstrative, restricting the fitting model to two degrees of freedom assumes a priori that this is sufficient to capture salient vocal fold dynamics, thus limiting future clinical applicability. To overcome this, we employ Bayesian inference to directly estimate tissue properties of a two-dimensional (2D) finite element vocal fold model from glottal area waveforms generated by both numerical simulations and recorded videos of synthetic vocal fold oscillations. We demonstrate that the 2D finite element model is not only capable of producing meaningful estimates with reasonable uncertainties, but is also capable of distinguishing between different experimental tissue properties, which is an essential step towards the development of patient specific models.

10:00

2aSC6. Advancement of flow velocity measurements in excised canine larynx model. Liran Oren, Charles Farbos de Luzan, Alexandra Maddox, Ephraim Gutmark, and Sid M. Khosla (Univ. of Cincinati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

In the classic source-filter model of speech production, flow modulation is the source of sound. Modulation of the flow refers to the fact that the volume flow, Q, is changing as the glottis opens and closes. Furthermore, studies have shown that the maximum flow declination rate (MFDR), which occurs during the latter part of the closing phase, is highly correlated with acoustic intensity (loudness) and acoustic energy in the higher harmonics. Therefore, measurements of the glottal airflow can provide important insights into voice mechanisms, dysfunction and efficiency. In recent years, particle image velocimetry (PIV) have become the method of choice for measuring Q because the technique can quantify, non-intrusively, the spatial and temporal information of the flow. The discussion includes progress of PIV measurements, from 2D to tomographic measurements, in the excised canine larynx model. Key findings related to intraglottal flow and glottal geometry, and their extension to modeling of voice mechanisms, are described.

10:20–10:35 Break

Contributed Papers

10:35

2aSC7. Systematic analysis of fluid-structure-acoustic interactions in an ex-vivo porcine phonation model. Marion Semmler (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Med. School, Dept. of Otorhinolaryngology, Head & Neck Surgery, Waldstr. 1, Erlangen, Bavaria 91052, Germany, marion.semmler@uk-erlangen.de), David Berry (Dept. of Head and Neck Surgery, David Geffen School of Medicine at UCLA, Los Angeles, CA), Anne Schützenberger, and Michael Döllinger (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Med. School, Dept. of Otorhinolaryngology, Head & Neck Surgery, Erlangen, Bavaraia, Germany)

Background: The phonation process is a complex interaction of different components, namely, the airflow from the lungs, the mucosal tissue in the larynx, and the resulting acoustic signal. From clinical experience, it is assumed that the presence of a pre-phonatory gap, oscillation asymmetries, and aperiodicities constitute detrimental factors to the resulting voice quality. In order to gain an in-depth understanding of the cause-effect chain and enable effective therapeutic measures, the crucial elements of the process need to be studied in isolation. *Materials & Methods:* The presented experimental set-up allows a systematic control and monitoring of the individual components of the excised porcine larynx. Varying degrees of asymmetric adduction, pre-phonatory gap size and flow rate reproduce irregular oscillation patterns associated with pathologic voice production. *Results:* The statistical analysis based on k-means clustering allows the identification of structural and aero-dynamic factors which influence voice quality. In addition to an increase in glottal closure and oscillation symmetry, an increase in glottal flow resistance produced a beneficial effect on the acoustic output signal and the closely coupled subglottal pressure. *Conclusion:* The clinical applicability and therapeutic value of these insights must be assessed in further studies.

10:50

2aSC8. The importance of medial surface vertical thickness to the control of voice production. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

While vocal fold vibration is often studied from a superior view, the vertical thickness of the vocal fold medial surface (or the portion that forms the glottal constriction) has long been hypothesized to play an important role in the control of voice production. This importance of the vertical thickness is confirmed in our recent computational studies, which showed that the vertical thickness, not the degree of vocal fold approximation, has a dominant effect on the vocal fold contact pattern and the spectral shape of the produced voice. Specifically, thicker folds often vibrate with a larger closed quotient, a lower H1-H2 value, and stronger excitation of higher-order harmonics. Thicker folds are also more likely to exhibit irregular vibration. These studies also showed an important effect of the transverse stiffness of the vocal fold in the coronal plane on the resulting voice quality, but a relatively small effect of the longitudinal tension on voice quality except around phonation onset. These results suggest that in addition to the superior view, more attention should be given to vocal fold shape along the vertical dimension and its control. [Work supported by NIH grant R01DC011299.]

11:05

2aSC9. Relationship between intraglottal geometry, vocal tract constriction, and glottal flow during phonation of a canine larynx. Charles Farbos de Luzan, Sid M. Khosla, Liran Oren, Alexandra Maddox (Dept. of Otolaryngol. - HNS, Univ. of Cincinnati, 231 Albert Sabin way, MSB 6313, Cincinnati, OH 45267-0528, farboscs@ucmail.uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

Hypothesis: The rapid reduction of the glottal volume flow rate, which usually occurs in normal phonation during the closing of the vocal folds, is measured by a quantity known as the maximum flow declination rate (MFDR). Our preliminary work highly suggests that intraglottal flow separation vortices (FSV), which form near the superior aspect of the divergent folds during closing, can directly affect MFDR. In this project, we hypothesize that the strength of the vortices is highly correlated with higher MFDR, larger divergence angles and stronger FSV. *Methodology*: We use particle image velocimetry to measure the strength of the FSV as well as the intraglottal pressures during phonation in four excised canine larynges with an attached mechanical vocal tract. Two vocal tract models are used, one with a false fold gap of 7 mm, one with a gap of 3 mm. *Results*: Stronger FSV (measured by vorticity) are correlated with glottic efficiency, larger glottal divergence angles and/or with decreased distance between the false folds. *Conclusion*: We show correlation of the strength of the FSV with different variables. These data are currently being used to validate computational flow structure models. These computational models then can be used to determine causation.

11:20

2aSC10. Multitaper harmonic analysis of infant vocalizations. Gordon Ramsay (Dept. of Pediatrics, Emory Univ. School of Medicine, Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, gordon.ramsay@emory.edu)

Infant vocalizations exhibit many special complexities relative to adult speech that make acoustic analysis difficult, even though the basic physics of sound production in the vocal tract is identical. Fundamental frequency is raised, resulting in wider-spaced harmonics that make it hard to accurately estimate the spectral envelope or locate formant resonances, which are higher than in adult voices, due to anatomical scaling effects. Irregular vocal fold vibratory regimes are often observed, involving period doubling and other nonlinear phenomena not seen in adult speech, along with atypical patterns of turbulent noise production. Unsurprisingly, traditional analysis techniques that depend on estimating the spectral envelope from the speech signal do not work well when applied to infant vocalizations. An alternative approach is proposed, using multitaper analysis to calculate the time-varying amplitude and phase of every harmonic component of the voice, along with the residual noise component, recovering the vocal tract transfer function from the results. The new technique is compared against linear

prediction and cepstral analysis using home audio recordings of 20 infants collected from 0 to 24 months using LENA technology. Developmental progressions in the acoustic structure of the infant voice are identified that cannot be found using traditional methods.

11:35

2aSC11. Adult imitating child speech: A case study using 3D ultrasound. Colette Feehan (Linguist, Indiana Univ., Bloomington, 107 s Indiana Ave., Bloomington, IN 47404, cmfeehan@indiana.edu)

Voice actors are interesting for linguistic study because of their unique abilities to manipulate their vocal tract and convey different social identities. This is essentially a field of professional folk linguistics where professionals manipulate their vocal tracts to convey socially indexed, linguistic features. This study uses Ultrasound paired with acoustic analyses to address what one amateur voice actor does while imitating a child voice. Previous studies have looked at anatomical and acoustic variations defining different character types such as laryngeal setting (Teshigawara, 2003; Teshigawara & Murano, 2004) and breathy voice in Anime (Starr, 2015). This study addresses specific tongue morphology resulting from an adult imitating a child's voice and serves as a pilot for future study of professional actors. The participant is one adult, amateur actor who produced CV syllables at different places of articulation with different vowel qualities. Notable manipulations are hyoid bone raising, gesture fronting, and tongue "troughing" where the sides of the tongue are used to narrow the oral cavity. The actor constricts the filter in multiple ways to shrink the usable space in the oral cavity and imitate the acoustic signal from a child's vocal tract. This presentation explores the anatomical manipulations the actor utilizes.

TUESDAY MORNING, 6 NOVEMBER 2018 — OAK BAY 1/2 (VCC), 8:45 A.M. TO 11:45 A.M.

Session 2aUW

Underwater Acoustics: Signals and Systems I

Danelle E. Cline, Chair

R&D, MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039

Contributed Papers

8:45

2aUW1. Deconvolved conventional beamforming applied to the SWellEx96 data. Tsih C. Yang (Ocean College, Zhejiang Univ., Bldg. of Information Sci. and Electron. Eng., 38 Zhe Da Rd., Hangzhou, Zhejiang 310058, China, tsihyang@gmail.com)

Horizontal arrays are often used to detect/separate a weak signal and estimate its direction of arrival among many loud interfering sources and ambient noise. Conventional beamforming (CBF) is robust but suffers from fat beams and high level sidelobes. High resolution beamforming such as minimum-variance distortionless-response (MVDR) yields narrow beam widths and low sidelobe levels but is sensitive to signal mismatch and requires many snapshots of data to estimate the signal covariance matrix, which can be a problem for a moving source. Deconvolution algorithm used in image de-blurring was applied to the conventional beam power of a uniform line array (spaced at half-wavelength) to avoid the instability problems of common deconvolution methods and demonstrated with real data (T. C. Yang, IEEE J. Oceanic Eng., 43, 160–172, 2018). The deconvolved beam output yields narrow beams, and low sidelobe levels similar to MVDR and at the same time retains the robustness of CBF. It yields a higher output signal-to-noise ratio than MVDR for isotropic noise. The method is applied here to the horizontal array data collected during the SWellEx96 experiment. Bearing time record are created to compare the performance of various beamforming methods and used to track the source position.

9:00

2aUW2. Using convolutional neural networks on perceptual and spectral features: Classification experiments on the TREX'13 dataset. Tara J. LeBlanc and John Fawcett (Defence Res. and Development Canada, Atlantic, 9 Grove St., P.O. Box 1012, Dartmouth, NS B2Y 2E2, Canada, tara.leblanc@drdc-rddc.gc.ca)

Distinguishing objects of interest on the seafloor from clutter remains a key problem facing the automatic target recognition (ATR) community. Because scattering at high frequencies relates more to the geometry of the scatterer, traditional ATR on high frequency imaging sonars is known to suffer in areas of high clutter. Recently, there has been interest in low frequency (1–50 kHz) wideband imaging sonars, because the complex combination of external (geometric) and internal (elastic) scattering that occurs in this frequency range is thought to improve classification in areas of high clutter. This work investigates the classification problem of distinguishing unexploded ordnance (UXO) from clutter using image-based Convolutional Neural Networks on the TREX'13 dataset. This dataset consists of experimental and modelled acoustic backscatter for objects interrogated acoustically at 3–30 kHz across a range of aspects. The model data are used exclusively for training, and the experimental data are used exclusively for testing purposes. The generated feature-set is a combination of acoustic colour and perceptual features, which are derived from modelling how humans perceive timbre. Further in an effort to address the differing amplitude modulations between sets of model and experimental data and improve classified performance, several moving window normalizations will be investigated.

9:15

2aUW3. Out-of-band beamforming in shallow water with horizontal arrays. Alexander S. Douglass (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu), Shima Abadi (Mech. Eng., Univ. of Washington, Bothell, Bothell, WA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Out-of-band beamforming methods, frequency-difference and frequency-sum beamforming, are array signal processing methods that allow a user to process signals at either a lower, below-the-signal-band difference frequency, or higher, above-the-signal-band sum frequency. Frequency-difference beamforming has previously been investigated for shallow-ocean propagation with sparse vertical arrays, demonstrating its capability to overcome the negative effects of spatial aliasing [Douglass, Song, and Dowling (2017). J. Acoust. Soc. Am. **142**, 1663–1673]. In the work presented here, data from the 2012 COAST (Cascadia Open-Access Seismic Transects)

experiment is utilized to explore out-of-band methods with horizontal arrays, where variations in environmental characteristics (bathymetry, sound speeds, etc.) may be more pronounced across the array. The abundance of array elements (636 in total) and total array length (7.9 km) in this experiment provide a means for controlling the sparsity and signal coherence within a chosen subarray to determine the capabilities and limitations of out-of-band beamforming. In addition, the bandwidth of the signal of interest is sufficient to compare out-of-band methods to conventional methods where the difference or sum frequencies match the in-band frequency used with conventional beamforming. [Sponsored by ONR.]

9:30

2aUW4. Underwater acoustics array spatial reversal convolution and beamforming. Anbang Zhao, Lin Ma, Chunsha Ge, Xuejie Bi (Acoust. Sci. and Technol. Lab.,Key Lab. of Marine Information Acquisition and Security (Harbin Eng. University),College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, HLJ 150001, China), (malin@hrbeu.edu.cn)

A novel underwater acoustic array signal processing method based on spatial reversal and convolution, aiming at improving detection performance, is proposed to meet the engineering application demand of the underwater acoustic array detection system. The theoretical and mathematical model of the proposed method is deduced and presented in detail. Based on the extended covariance matrix of the physical array, the array sensitivity equalization and the noise suppression technique of the underwater acoustic array are studied. The additional gain provided by sensitivity equalization and noise suppression techniques is analyzed and exploited. Computer modelling and various numerical simulations are applied to demonstrate the proposed method's effectiveness. At-sea experimental data are processed to test the performance in real underwater acoustic environment. These results indicate that the proposed array processing method can effectively improve the detection ability. The proposed techniques in this paper provide an insightful way to improve the detection performance of the underwater acoustic array.

9:45

2aUW5. Adaptive beamforming for uniformly-spaced linear hydrophone array using temporal convolutional neural networks. Dezhi Wang, Lilun Zhang, Changchun Bao (College of Meteorol. and Oceanogr., National Univ. of Defense Technol., Sanyi St., Kaifu District, Changsha 410073, China, wang_dezhi@hotmail.com), Kele Xu (School of Comput., National Univ. of Denfense Technol., Paris, France), Boqing Zhu (School of Comput., National Univ. of Denfense Technol., Changsha, China), and Zengquan Lu (College of Meteorol. and Oceanogr., National Univ. of Defense Technol., Changsha, China)

In oceanic remote sensing, large discrete linear hydrophone arrays are usually utilized to enhance the signal-to-noise ratio (SNR) by means of beamforming that attenuates noise coming from the directions outside the target direction. Although the widely adopted delay-and-sum (DS) and filter-and-sum (FS) beamforming techniques can improve the performance of linear hydrophone arrays in some scenarios, both DS and FS methods have limited adaptability to the changing oceanic environments especially with spatially correlated noise. Moreover, these beamforming techniques aims to optimize the SNR, which is not completely consistent with some objectives such as target recognition. In this work, a neural network adaptive beamforming framework for an uniformly-spaced linear hydrophone array is proposed to make use of the large-scale array signals and address the above issues. In particular, an architecture consists of temporal convolutional neural networks is designed to predict the beamforming filter coefficients in time domain by taking the raw multi-channel waveforms of sound pressure as the input. The filter prediction networks are also jointly trained with a convolutional neural network (CNN) based classification model with the purpose of increasing the target recognition accuracy. The proposed approach is validated by experiments carried out in the south coastal waters of China.

10:00–10:15 Break

10:15

2aUW6. Detection and classification of whales calls using band-limited energy detection and transfer learning. Danelle E. Cline and John P. Ryan (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, dcline@mbari.org)

The Monterey Bay Aquarium Research Institute has been recording since July 2015 almost continuously at the Monterey Accelerated Research System (MARS) cabled observatory in Monterey Bay, California, USA. This long-term recording contains thousands of whale calls to help further our understanding of interannual, seasonal, and diel patterns. Here we report on our highly accurate detection and classification method developed to classify blue whale A, B, and D calls and Fin whale 20 Hz pulses. The foundation of the method is a computationally efficient and tuned decimation filter to convert the broadband hydrophone 128 kHz signal to 2 kHz which preserves the low-frequency signal and avoids any high-frequency aliasing. Detection is done using a band-limited-energy-detection filter to find potential calls in the decimated data. Spectrograms are then generated for potential calls and enhanced with local image normalization followed by smoothing by convolving in either time or frequency. Classification is done using the Google Inception v3 model with a transfer learning method. Overall, false positive rates are very low despite variability in whale call shape and background noise.

10:30

2aUW7. Passive source depth discrimination in deep-water. Rémi Emmetiere (ENSTA Bretagne, 2 rue Francois Verny, Brest Cedex 9 29806, France, remi.emmetiere@ensta-bretagne.org), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Marie Gehant Pernot (Thales Underwater Systems, Sophia Antipolis, France), and Thierry Chonavel (IMT Atlantique, Plouzané, France)

In this work, we address the problem of passive source depth discrimination using a horizontal line array (HLA). The scope is restricted to low-frequency sources (f<500 Hz), broadband signals (a few Hertz bandwidth), deep-water environment (D>1000 m), and distant sources (at least several kilometers). The proposed method is based on the underlying physics driving the propagation. It notably uses the concept of mode trapping and waveguide invariant. In deep water, the waveguide invariant largely depends on source depth, and is thus an interesting input for source depth discrimination. An algorithm is proposed to compute energy ratio in modal groups. The input data for the algorithm are a range-frequency intensity, as measured on a HLA. The modal groups are thus defined based on their respective waveguide invariant values, which is a main difference with existing depth discrimination methods adapted to shallow water context. This idea is explored, and extended to propose a source depth discrimination which is performed as a binary classification problem. As long as the sound speed profile features a surface thermocline, the algorithm does not require detailed knowledge about the environment and it allows one to classify sources under two hypotheses, above or under a threshold depth.

10:45

2aUW8. A Bayesian approach to a passive range estimation method of underwater sound source in shallow water waveguide. Xiaoman Li and Shenchun Piao (Harbin Eng. Univ., No.145, Nantong St., Nangang District, Harbin, Heilongjiang, Harbin 150001, China, lixiaoman@hrbeu.edu.cn)

In this paper, a matched mode processing method is used to estimate the range of underwater sound source in shallow water, in which the replica field is calculated with the P and Q normal mode model. In this normal mode model, the replica field is described with the bottom phase shift coefficient, the sea depth and the average sound speed profile. The influence caused by the uncertainty of environment parameters, on the accuracy of range estimation is discussed with Bayesian approximation. Bayesian approximation is a good way to estimate the unknown parameters based on probability of data. The Gaussian distribution model and Markov Chain Monte Carlo method are used in this paper for calculating the posterior probability density. The effect of the uncertainty or error of environment parameters to the ranging result is obtained. The practicality of the ranging method is raised significantly. The procedure and the discussed results are given.

11:00

2aUW9. Experimental performance evaluation of underwater active detection and positioning system. Lin Ma, Anbang Zhao (Acoust. Sci. and Technol. Lab.,Key Lab. of Marine Information Acquisition and Security (Harbin Eng. University),College of Underwater Acoust. Eng., No. 145, Nantong St., Harbin, HLJ 150001, China, malin@hrbeu.edu.cn), T. Aaron Gulliver (Dept. of Elec. & Comput. Eng., Univ. of Victoria, Victoria, BC, Canada), and Caigao Zeng (Acoust. Sci. and Technol. Lab., Key Lab. of Marine Information Acquisition and Security (Harbin Eng. University), College of Underwater Acoust. Eng., Harbin, China)

Underwater targets active detection and positioning are of increasing challenges in various underwater practical engineering applications. The performance of the proposed detection and positioning algorithms are validated and evaluated by designed open-lake experiments. The widely used matched-filter, conducted in frequency-domain, is improved by normalized least mean square algorithm to achieve the better detection ability. The improved detector is applied to the underwater targets detection in these experiments. Post-processing technique based on autocorrelation is proposed to enhance the performance of detection by cancelling the noise existing in frequency-domain. The underwater target is located on the basis of the time-delay and azimuth estimation. The open-lake experimental results show that the detection and positioning system is robust and effective in the complex underwater environment. The designed model and these proposed techniques provide a better way to underwater target active detection and positioning.

11:15

2aUW10. Optimization of deployment depth for active towed array sonar (ATAS) using simulated annealing. Sangkyum An (Ocean Eng., Seoul Univ., Dept. of Naval Architecture and Ocean Eng., Collage of Eng., Seoul National Univ., Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, navy60@snu.ac.kr), Keunhwa Lee (Defense Systems Eng., Sejong Univ., Seoul, South Korea), and Woojae Seong (Ocean Eng., Seoul Univ., Seoul, South Korea)

The active towed array sonar (ATAS) is an active operating sonar towed behind the surface ship and deployed at various depths. The performance of ATAS may vary with the deployment depth. This study proposes a method which calculates the optimal deployment depth using simulated annealing (SA) in order to reduce the total computational time spent in the direct search of the deployment depth. We define a sonar performance function (SPF) with the probability of detection, which represents as a degree of how well the ATAS might perform at particular deployment depth. Then, the optimal depth is defined as the depth where the SPF is maximized. The SPF depends on the acoustic environment, target position and source-receiver depth. The optimization results are compared with the direct calculation of SPF at all deployment depth of source and receiver. Also, our study provides the optimal depths in the area of East Sea of Korea.

11:30

2aUW11. Bi-directional equalization for long-range underwater acoustic communication in East Sea of Korea. Hyeonsu Kim, Sunhyo Kim, Jee Woong Choi (Marine Sci. and Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 15588, South Korea, hskim00@hanyang.ac.kr), and Ho Seuk Bae (The 6th R&D Inst., Agency for Defense Development, Changwon, South Korea)

The long-range underwater acoustic communications using a sound channel in deep sea have been studied for many years. The acoustic waves transmitted through the sound channel can be propagated by long distance because there is no interaction with the sea surface and bottom interfaces. However, the transmitted signals are distorted by multipath propagation. Time reversal combining followed by a single channel equalizer can reduce efficiently the effects of multipath communication channel and thus increases signal-to-noise ratio by obtaining temporal and spatial diversity. However, the conventional time reversal tends to increase the entire error rate when burst errors occur. Long-range communication signals are particularly vulnerable to burst errors due to the weak signal strength. In this talk, the bi-directional equalization is applied to compensate for the distortion of long-range communication signals where the burst errors occur. Since the bi-directional equalization combines the soft outputs of forward and backward equalization, the effect of error propagation due to burst error can be reduced. The long-range communication experiment data acquired in the East Sea of Korea in October 1999 is used to verify the performance of the bi-directional equalization. The results show that the bi-directional equalization can be effectively applied when burst errors exist. [Work supported by the ADD(UD170022DD) and the National Research Foundation of Korea (NRF-2016R1D1A1B03930983).]

TUESDAY AFTERNOON, 6 NOVEMBER 2018 THEATER (VCC), 1:00 P.M. TO 4:15 P.M.

Session 2pAA

Architectural Acoustics and ASA Committee on Standards: Auditorium Acoustics and Architectural Design: Challenges and Solutions II

Jin Yong Jeon, Cochair
Department of Architectural Engineering, Hanyang University, Seoul 133-791, South Korea

Thomas Scelo, Cochair
Marshall Day Acoustics, 1601, 16/F The Hollywood Centre, 233 Hollywood Road, Sheung Wan 0000, Hong Kong

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. One project begets many: The role of one seminal project in the development of many design solutions (Part 1). Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

This paper, presented in three parts, describes the role of Threshold Acoustics' study and corrective treatments to Verizon Hall at the Kimmel Center for the Performing Arts, as a seminal project in the firm's work. Elements of the study of and solutions developed to improve the hall have proven to be repetitive themes in our work, both in the renovation of other halls and in the design of new spaces. This initial paper describes the interactions with the Philadelphia Orchestra, and the how the subjective concerns expressed by the orchestra led to unexpected conclusions regarding the overhead canopy and walls at the downstage edge. This portion of the presentation will focus on lessons learned for the interactions between the stage canopy weight, spatial density, and height and how these have informed subsequent design work.

1:25

2pAA2. One project begets many: The role of one seminal project in the development of many design solutions (Part 2). Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

In the second part of this multi-part paper, the solution for the cap of the organ chamber at Verizon Hall will be presented. The original cap was a lightweight material, allowing significant low frequency energy—critical to the success of any pipe organ—to escape the acoustic volume of the room. The solutions implemented to improve the sound of the organ were limited by structural capacities, and were ultimately developed as lightweight but very stiff honeycomb core panels. The strategy of lightweight-yet-stiff materials has been implemented in numerous other projects to achieve low frequency reflectivity. Examples will be presented illustrating development of this strategy over the past decade.

1:45

2pAA3. One project begets many: The role of one seminal project in the development of many design solutions (Part 3). Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

In the third and final part of this multi-part paper, computer modeling will be described as a tool for presentation to client groups (often skilled listeners but not expert acousticians). Over the course of the Verizon Hall project, computer models were used to present visual illustrations that explained the acoustic concerns experienced by the orchestra. Starting with this project, Threshold began to develop computer models as tools to translate acoustic phenomena in ways that clients could understand visually and, over time, aurally. The development of customized computer modeling techniques has further supported communication with non-acousticians, and examples of these techniques will be presented.

2:05

2pAA4. Vibration isolation for world-class performance spaces, Part I: A brief history. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

World-class performance spaces are typically located in central urban areas, often surrounded by sources of groundborne noise such as subways and streetcar lines or sources of structureborne noise within the building itself which, if not addressed, can be disruptive to the performances inside the buildings. The advent of digital recording within performance spaces has also been a major motivation for reducing background noise, often to levels below the threshold of human perception. The history and development of vibration isolation incorporated into such spaces will be discussed.

2:25

2pAA5. Vibration isolation for world-class performance spaces, Part II: Examples. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

This paper will present examples of vibration isolation incorporated into world-class performance spaces in North America and oversees, including acoustical and vibration measurements demonstrating the effectiveness of the vibration isolation to reduce groundborne/structureborne noise.

2:45–3:00 Break

Contributed Papers

3:00

2pAA6. Identifying trends in worship space acoustics: 3 decades of design. Stephanie A. Ahrens, Erica E. Ryherd, Lauren M. Ronsse (Architectural Eng., Univ. of Nebraska - Lincoln, Durham School of Architectural Eng. & Construction, Omaha, NE 68182-0816, sahrens@unomaha.edu), and David T. Bradley (Inclusion, Diversity and Equity, Smith College, Northampton, MA)

Worship Space Acoustics: 3 Decades of Design—published by the Acoustical Society of America in 2016—is a book rich with detailed acoustic and architectural information for 67 worship venues with diverse acoustic designs. The venues represent a range of worship space types, various geographic locations around the globe, an assortment of architectural typologies, and a variety of sizes (including seating capacities ranging from 100 to 21,000). This project aimed to synthesize the acoustic and written content presented in the book in order to identify design trends across the 67 venues. Qualitative data included venue typology, location of materials, sound system design, floor plate and room shape, intended use of space, and noise sources, which were mined and analyzed using text analytics software. Quantitative data included reverberation time, background noise level, seating capacity, and room volume. By compiling the various qualitative and quantitative data across all venues, acoustic data and design strategies were synthesized to illustrate popular trends in worship space acoustic design over the last three decades.

3:15

2pAA7. The modern sophistication of acoustics in education environments. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

In the past 10 years the use of acoustic design and products in educational environments, from elementary schools to universities, has grown exponentially. While acoustics in general has been more widely embraced to solve problems related to modern design technologies, the use of more sophisticated designs and materials has been widely accepted as an alternative to the simple absorptive materials of the past. Some secondary school designs challenge the finest college and university campuses for both creative and effective acoustic spaces. Likewise, some colleges and universities challenge the finest commercial venues, studios and rehearsal spaces in both function and design. Examples will be shown in the presentation including a new convertible acoustic system at Tabor College designed to make the hall usable for both classical performance and sound reinforced, high volume contemporary performances as well. Discussions as to the principals behind the use of acoustics in these environments will be offered.

3:30

2pAA8. The preference of orchestra ensemble on sound absorption design in a concert stage. Yi Run Chen (Architecture, National Taiwan Univ. of Sci. and Technol., No.43, Keelung Rd., Sec.4, Taipei 10607, Taiwan, D10413009@mail.ntust.edu.tw), Lawrence Huang (none, Kaohsiung, Taiwan), and Weihwa Chiang (Architecture, National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan)

Musicians in a symphony orchestra have to deal with more sound balance problems in a relatively compact stage when appropriate spacing between orchestra sections become unavailable. Under such circumstances, applying sound absorptive surfaces around the stage might benefit the condition of either hearing oneself or hearing each other. This paper explored the musicians' preference on energy-restrained design with varying frequency characteristics, amount of coverage and location. Experiments took place with a well-reputed university orchestra at their rectangular concert hall in the routine practices. Room acoustic parameters were measured on site and in an 1/10 scale model, empty and occupied, to be compared with computer simulations.

3:45

2pAA9. Case study: Retrofitted solution to address sound flanking via window mullions of stacked residences. Pier-Gui Lalonde (Integral DX Eng. Ltd., 907 Admiral Ave., Ottawa, ON K1Z6L6, Canada, pier-gui@integraldxengineering.ca)

This paper presents a case study concerning a noise transmission issue between stacked residences in a condominium building. The façade design included vertical window mullions that were continuous and common to the upper and lower units, creating a sound flanking path. Low acoustic privacy and disturbing intrusive noise resulted. A solution was required to mitigate the noise transmission issue while meeting additional requirements: fully reversible modifications to façade components were preferred, no interference to the function of the façade components was allowed, and occupant expectations regarding the appearance of the final solution needed to be met. This paper will present the approach that was followed in order to diagnose the issue, as well as the analysis which led to the solution that was ultimately implemented.

4:00

2pAA10. Noise and vibration measurements of a machine-room-less elevator system. Andrew Williamson (RWDI, 301-2250 Oak Bay Ave., Victoria, BC V8R 1G5, Canada, andrew.williamson@rwdi.com)

A trend in the design of new condominium buildings is the use of "machine-room less" (MRL) elevator systems. The primary difference between MRL elevator systems and more traditional elevator systems is that, rather than locating the hoist machinery in a rooftop penthouse, the hoist machinery is installed in the elevator shaft. In this location (which is typically closer to residential units), there is greater potential for the operation of the machinery to result in complaints from residents of excessive noise and vibration levels. This paper presents a case study of noise and vibration measurements that were conducted in a condominium building with a MRL elevator system. Measurements were conducted both within the affected residential unit as well as in the elevator shaft. In conducting the measurements, particular attention was paid to determining the effectiveness of the vibration isolation that was provided for the elevator hoist machinery within the elevator shaft.

Session 2pAB

Animal Bioacoustics and Signal Processing in Acoustics: Anything You Can Do I Can Do Better: Bat Versus Dolphin Biosonar

Laura Kloepper, Cochair
Biology, Saint Mary's College, 262 Science Hall, Saint Mary's College, Notre Dame, IN 46556

Brian K. Branstetter, Cochair
National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106

Chair's Introduction—1:00

Invited Papers

1:05

2pAB1. The acoustic world of bat biosonar. Rolf Müller, Michael J. Roan, Mohammad Omar Khyam, and David Alexandre (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Bats and toothed whales (odontocetes) have both independently evolved sophisticated biosonar systems. This raises the question how similar the functional principles of these systems are. Could, for example, insights gained from bats be assumed to hold for odontocetes or vice versa? Could both systems be lumped together as a single source of inspiration for novel engineering approaches to sonar sensing? Similarities and differences between the biosonar systems of bats and odontocetes are likely to depend on the respective acoustic environments in which these systems have evolved as well as on the evolutionary starting points and capabilities for adaptations of these two very different phylogenetic groups. In this presentation, the focus will be on comparing the acoustic environments of bats and odontocetes. The acoustics of biosonar sensing can be organized into three different aspects: (i) the properties of the propagation medium, (ii) the geometry and material of the boundaries that limit the propagation channel (this includes targets of interest and clutter), and (iii) the time-frequency and spatial properties of the sources. In this presentation, these aspects will be reviewed for the in-air biosonar of bats. In the companion talk, the same will be done for the underwater biosonar of the odontocetes.

1:25

2pAB2. Jittered echo delay resolution in bats and dolphins. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Ryan A. Jones, Jason Mulsow, Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), and Patrick Moore (National Marine Mammal Foundation, Spring Valley, CA)

Fine-scale echo delay resolution has been investigated using a "jittered" echo paradigm, where animals discriminate between electronic echoes with fixed delay (i.e., simulating fixed range) and echoes with delays that alternate (jitter) on successive echoes. The data consist of the animals' discrimination performance (e.g., error rate) as a function of the amount of jitter in the echo delay (the time interval over which the echoes jitter). Results of jitter delay experiments in big brown bats (*Eptesicus fuscus*) are extraordinary (and controversial) because they suggest that bats can extract information from within the envelope of the crosscorrelation function between an emitted signal and its received echo and therefore may be capable of operating as a coherent receiver. Recently, the jitter delay paradigm has been adapted for use with bottlenose dolphins (*Tursiops truncatus*) at jitter delays down to 1 μs. Jitter delay acuity results for dolphins show qualitative similarities to those from bats: error function peak widths are below the envelope of the biosonar pulse autocorrelation function and dolphins can perceive a jitter in echo polarity with no change in echo delay. However, further testing with submicrosecond jitter values are required to determine if dolphins possess "hyperacuity" as reported for bats.

1:45

2pAB3. Reconstruction of acoustic scenes in biosonar. Chen Ming (Neurosci., Brown Univ., 814 Cascade Ct., Blacksburg, VA 24060, chen_ming@brown.edu), Michael J. Roan, Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Biosonar mechanisms are highlighted by comparing acoustically reconstructed scenes derived from generalized methods with the performance of echolocating animals. Bioinspired computations replacing standard methods then offer a path towards understanding the animal's solutions. Seemingly simple outdoor spaces present flying bats with complex scenes when vegetation and the ground are factored in, and even sound-treated research spaces such as flight rooms evolve into complex scenes as transmitted sounds propagate through the space. Underwater scenes, particularly for shallow water, are much more complicated because longer propagation distances create multipath reverberation that often overlaps with echoes. Interpulse intervals of biosonar emissions must be short to support rapid updating of tracking and perception, but reverberation means that pulse-echo ambiguity occurs. Two reconstructive methods are

useful—the HARPEX method for visualizing the progression of echoes and reverberation following a sound, and a HARPEX-like method reconfigured as a forward-looking sonar. Acoustic datasets visualize airborne and underwater sonar scenes by reconstructing sound images for successive time frames following a transmitted sound using a tetrahedral soundfield microphone or a forward-looking sonar head. The complex acoustics of vegetation are examined with a simulation model built to understand the bats' perception of vegetation. (Work supported by ONR.)

2:05

2pAB4. Biosonar-mediated prey tracking in air and in water—The fast and the persistent. Danuta M. Wisniewska (Ctr. d'Etudes Biologiques de Chizé, Ctr. National de la Recherche Scientifique, 405 Rte. de La Canauderie, Villiers-en-Bois 79360, France, danuta@stanford.edu), Mark Johnson (ZooPhysiol., BioSci., Aarhus Univ., St. Andrews, United Kingdom), Heather Vance (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St Andrews, St. Andrews, United Kingdom), Laura Stidsholt, and Peter T. Madsen (ZooPhysiol., BioSci., Aarhus Univ., Aarhus, Denmark)

Microchiropteran bats and toothed whales are the only taxa to echolocate for prey. The two biosonar systems have evolved independently, in media with very different physical properties that determine the production, propagation, and reflection of sound, as well as the means and speeds with which the echolocating predators and their prey can move and maneuver. Yet, they have converged on remarkably similar acoustic behavior; the animals adjust the timing and level of their sound emissions as they near their targets, eventually entering the buzz phase with high rates of low-amplitude signals. But while the buzz of a bat lasts a fraction of a second, prey pursuits by toothed whales are one to two orders of magnitude longer. Here, we explore these differences and their consequences for biosonar performance and prey capture success. We use echograms recorded with acoustic dataloggers on porpoises to show that, like some bat species, porpoises hunt in different environments targeting several types of prey and adjusting their biosonar outputs to manage different acoustic soundscapes. However, we further show that, unlike bats, porpoises are able to carefully track their prey with adjusted click intervals and dynamically accommodate prey movements to provide high spatial resolution without range ambiguities.

2:25–2:40 Break

Contributed Papers

2:40

2pAB5. Flexibility means adaptability: Bats adapt to jamming scenarios better than dolphins. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Bats and dolphins both use sonar to image their environment, but frequency modulated (FM) bats demonstrate greater flexibility in echolocation than their marine counterparts, modifying the time-frequency shape of their calls. Bats' FM signals can change depending on foraging phase, environment, or even density and proximity of conspecifics. By changing their time-frequency call structure, bats can acquire different information than by using a stereotyped signal, such as the impulsive signal produced by dolphins. In my talk, I argue that bats have the superior sonar system due to this flexibility. I compare the signals between bats and dolphins and demonstrate new data on how both bats and dolphins adjust echolocation signals in jamming scenarios.

2:55

2pAB6. Biosonar capabilities of large-brained dolphins and small-brained bats: Size does matter. Kaitlin R. Van Alstyne, Brian K. Branstetter, Sam H. Ridgway (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, katie.vanalstyne@nmmpfoundation.org), James J. Finneran, and Mark J. Xitco (US Navy Marine Mammal Program, SSC Pacific Code 71510, San Diego, CA)

The large encephalized brain of the bottlenose dolphin, with a small corpus callosum, functions to maintain continuous, 24-hour biosonar signal processing. Bats are not faced with the dolphin's challenge of maintaining constant vigilance, as bats are able to retreat to roosts for safety from predators. Although both bats and dolphins echolocate, dolphins face greater challenges in the aquatic environment. Dolphins must locate prey of a similar density to that of the environment, which itself has a high speed of sound transmission. Bats can readily distinguish between prey from the surrounding air because of a strong impedance mismatch, and the speed of sound is much slower. Perhaps stemming from these challenges, dolphins have developed a highly sophisticated echolocation system in which they are able to emit signals and receive echoes, decide, and respond accordingly at a remarkably rapid speed. Dolphin brains may also be specially equipped for early auditory development in utero. The dolphin body has a close impedance match to water, exposing the fetus to environmental sounds. Furthermore, auditory fibers develop myelin early during gestation. Thus, although dolphins and bats both echolocate, the dolphin's experience during life in the ocean promoted the development of a large and specialized brain.

3:10

2pAB7. Don't bring a knife to a gunfight: Humpback whale biosonar reigns supreme. Eduardo Mercado (Dept. of Psych., Univ. at Buffalo, SUNY, Buffalo, NY 14260, emiii@buffalo.edu)

Stroboscopic echo imaging of the sort used by most bats and dolphins is well suited for the task of rapidly gulping prey. As an information seeking strategy, however, strobing is about as sophisticated as duct-taping a flashlight onto a helmet. Humpback whale biosonar signals, in contrast, create a temporally-extended echoic soundscape the likes of which no strober could ever conceive. By spectrally interleaving quasi-CF units with packets of shorter-duration FM units, humpback whales can likely detect multiple targets simultaneously from multiple bearings, including targets more than 5 km away, while at the same time tracking movements of individual targets. By maintaining relatively stationary positions and depths while signaling, humpback whales can potentially use not only direct echoes from targets, but also modulations in familiar acoustic scenes ("acoustic glints") to interpret the movements of distant targets. By modulating temporal substructure within fixed-rate streams of signals, humpback whales may be able to extract self-generated echoes from a sea of potential maskers. The scope of vocal control, auditory processing, perceptual flexibility, and echoic memory required for a whale to construct and interpret such rich percepts far exceeds the kinds of delay-tuned, chirplet detection that bats and dolphins rely on to bump into food.

3:25

2pAB8. Welcome home: Brazilian free-tailed bats adjust their echolocation calls to detect cave opening. Kathryn McGowan and Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, kmcgowan01@saintmarys.edu)

While both bats and dolphins use sonar to navigate their environment, the sensory challenges between the two are not equal since dolphins typically navigate in the open ocean and bats often navigate in cluttered environments. Brazilian free-tailed bats (*Tadarida brasiliensis*), for example,

adjust their frequency modulated (FM) call structure based upon their environment, producing lower-bandwidth calls in open environments, and higher-bandwidth calls in cluttered environments. In this study, we examined how Brazilian free-tailed bats change the bandwidth of calls when locating the cave opening in a flat, non-cluttered environment. We extracted individual echolocation calls from two locations around the cave, one away from the cave opening and one next to edge of the cave opening, with the only distinctive difference in the environment of the locations being the presence of the cave edge. We found higher starting and lower stopping frequencies for the FM calls at the cave edge, resulting in an increased bandwidth. This higher bandwidth suggests the bats may rely on edge detection to locate the cave opening and change the bandwidth of their signals to improve target resolution when returning to the roost.

3:40

2pAB9. Dolphins' echolocation based on reverberations through skull?
Lapo Boschi (ISTEP, Sorbonne Univ., 4 Pl. Jussieu, Paris 75005, France, lapo.boschi@upmc.fr), Michael Reinwald, Quentin Grimal (Biomedical Imaging Lab., Sorbonne Univ., Paris, France), Jacques Marchal (IJLRA, Sorbonne Univ., SAINT-CYR-L'ECOLE, France), and Stefan Catheline (LabTAU, INSERM, Lyon, France)

The sensitivity of odontocetes to changes in sound source azimuth and, in particular, elevation (minimum audible angles) has been reported to be superior to those of humans and bats. It has been suggested that binaural/spectral cues might be insufficient to account for this skill. We investigate bone-conducted sound in a short-beaked common dolphin's mandible and attempt to determine whether and to what extent it could contribute to the task of localizing a sound source. Experiments are conducted in a water tank by deploying, on the horizontal and median planes of the skull, sound sources that emit synthetic clicks between 45 and 55 kHz. Elastic waves propagating through the mandible are measured at the pan bones and used to localize source positions via binaural cues, as well as a correlation-based full-waveform algorithm. We find that by making use of the full waveforms, and, most importantly, of their reverberated coda, the accuracy of source localization in the vertical plane can be enhanced. While further experimental work is needed to substantiate this speculation, our results suggest that the auditory system of dolphins might be able to localize sound sources by analyzing the coda of biosonar echoes.

3:55

2pAB10. What bats may do better: Emission beampattern complexity.
Hui Ma, Yanan Zhao (Shandong University - Virginia Tech Int., Lab., Shandong Univ., Jinan, Shandong, China), Liujun Zhang (Mech. Eng., Virginia Tech, Jinan, China), and Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Horseshoe bats (family Rhinolophidae) and the related Old World round-leaf nosed bats (Hipposideridae) have conspicuous emission baffles ("noseleaves") that not only have a high degree of geometric complexity but can also change their shapes under active muscular actuation during emission of the biosonar pulses. Our work has shown that these shape changes occur in tight synchrony with pulse emission and are large enough to affect the acoustic diffraction process. Furthermore, our experiments with bats positioned on an experimental platform have shown that these noseleaf motions impart dynamic signatures onto the emitted ultrasonic wave packets. We also have collected pilot data from flying bats executing natural maneuvers that indicate geometric complexity as well as time-variance in the beampatterns based on a comparison with a loudspeaker that was placed in the same position as the bat to serve as a static reference. An information-theoretic analysis has shown that this emitter dynamics results in the encoding of additional sensory information that is useful, e.g., for direction finding, but future work is needed to determine if and how these effects may fit into the animals' natural biosonar behaviors. It remains to be seen if dolphins are make use of a similar biosonar emission dynamics.

4:10

2pAB11. Ultra high definition 3D tracking of biosonar of Amazon River dolphins and others mammals into the wild.
Gerald Blakefield (MADC EADM, Seattle, Washington), Marie Trone (Valencia Coll., Kissimee, FL), Valentin Barchasz (LIS, CNRS, AMU, Univ Toulon, Toulon, France), Valentin Gies (IM2NP, CNRS, Univ Toulon, Toulon, France), Dave Bonnett (MADC EADM, Seattle, Washington), and Herve Glotin (LIS, CNRS, AMU, Univ. Toulon, USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr)

We designed and built a high capacity digital analog converter : JASON Qualilife DAQ [1] which is capable of sampling at 1 MHz on 5 channels simultaneously to assess cetacean biosonar and the source position and orientation relative to the array using time delay of arrivals of the biosonar acoustical signals. We used JASON with underwater portable arrays housing between 3 and 7 hydrophones (for more than 4 hydrophones we connected two cards sharing one common hydrophone). This has been used to monitor Physeter m. biosonar in near field, and Inia. g. in the Peruvian Amazon from 2014 to 2018. Results demonstrate the efficiency of this advanced low cost scientific instrumentation : it allows with only 4 hydrophones to track in 3D the precise movements of the *I. g.* dolphin, and to determine key behavioral features, like the velocity of its rostrum rotation, while following its highly defined biosonar emissions. This is the first, at our knowledge, that wild amazon dolphin biosonar is described inside its ecosystem. Perspectives on automatic low power trigger [2] for counting of the dolphins and extraction of individual acoustic invariant are given, as well as research avenues on biosonar of other species. *[1] Gies et al, JASON Qualilife Daq, in DCLDE2018, smiot.univ-tln.fr,sabiod.org/bib [2] Fourniol et al., Low-Power Frequency Trigger for Environmental Internet of Things, in IEEE/ASME Mechatronic & Embedded Systems* http://sabiod.org/bib*, 2018 Samples and supp. material: http://sabiod.org/JASON.*

4:25

2pAB12. The micromechanics and bioacoustic behaviour of *Bunaea alcinoe* moth scales.
Zhiyuan Shen, Thomas R. Neil, Daniel Robert (School of Biological Sci., Univ. of Bristol, Office 2B09, Life Sci. Bldg., 24 Tyndall Ave., Bristol BS81TQ, United Kingdom, shenyuan675603@gmail.com), Bruce W. Drinkwater (Dept. of Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), and Marc W. Holderied (School of Biological Sci., Univ. of Bristol, Bristol, United Kingdom)

Through the 65 million year acoustic arms race between moths and bats, different moth species have evolved different defence strategies against bat echolocation. For non-toxic moth species without hearing capability, passive acoustic camouflage is thought to be the most efficient way to evade bat predation. Being the elementary building blocks covering moth wing surfaces, scales have been hypothesized as the main organ creating such acoustic camouflage. There is, however, no understanding for the relation between scale microstructure and wing acoustic performance. This report represents the first effort to numerically and experimentally characterize moth scale biomechanics and vibrational behaviour. 3D microstructures of *Bunaea alcinoe* moth scales have been characterized using various microscopies. A parameterized finite element model has been built to replicate the double-layered perforated scale bio-nanomaterial. Both experimental and numerical analyses have proved that the first three resonance frequencies of a single scale lie within the bat echolocation frequency range. Here, we propose numerical models that explain how the resonances can contribute to the acoustic performance of wings. This study contributes to the on-going discussion of the evolution of ultrasonic camouflage in moths. We aim to use our findings to generate biomimetic light-weight noise mitigation materials.

4:40

2pAB13. Dolphin and bat sonar: Nonlinear acoustic effects.
Thomas G. Muir, Jack A. Shooter, and Mark F. Hamilton (Appl. Res. Laboratories, Univ. of Texas at Austin, P/O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

Both dolphin and bat sonars operate in an amplitude and frequency regime where nonlinear effects can be observed, particularly distortion and the generation of harmonics. Computations illustrating these nonlinear

effects are presented, quantifying effects that could be observed in experiments on echolocating signals conducted with wideband sensors. The mathematical model used is the Khokhlov-Zabolotskaya-Kuznetsov (KZK) nonlinear parabolic wave equation, which includes diffraction and absorption for directional beams, with computations performed in the time domain using a finite-difference scheme. Inputs to the computations are sample dolphin echolocation transients (clicks) and bat echolocation tone bursts (FM Chirps—fundamental component), taken from the literature for a few species of high-amplitude emitters. Calculations begin in the water or air media. Propagating high-amplitude dolphin clicks are shown to distort, create harmonics, and suffer nonlinear attenuation at ranges of tens of meters. Propagating high-amplitude bat chirps are shown capable of undergoing some nonlinear distortion at ranges within 10 m. The results may be useful in understanding the significance of nonlinear effects for work with these animals. (Work supported by ARL:UT Austin.)

TUESDAY AFTERNOON, 6 NOVEMBER 2018 ESQUIMALT (VCC), 1:00 P.M. TO 3:40 P.M.

Session 2pAO

Acoustical Oceanography, Animal Bioacoustics, Underwater Acoustics, and Signal Processing in Acoustics: Machine Learning and Data Science Approaches in Ocean Acoustics II

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Shima Abadi, Cochair
University of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011

Invited Papers

1:00

2pAO1. Orcasound app: An open-source solution for streaming live ocean sound to citizen scientists and cloud-based algorithms. Scott Veirs (Beam Reach (SPC), 7044 17th Ave. NE, Seattle, WA 98115, sveirs@gmail.com), Val Veirs (Beam Reach (SPC), Friday Harbor, WA), Paul Cretu (Jotengine, San Francisco, CA), Steve Hicks (Einhorn Eng., Encinitas, CA), and Skander Mzali (Jotengine, San Francisco, CA)

A growing number of hydrophones stream ocean sound data in near-real-time to automated detectors. Many of these streams are also provided to the public, but none yet appeal for citizen scientists to provide real-time detection or classification services. Within the Orcasound hydrophone network—orcasound.net (WA, USA)—live human listeners have always been in friendly competition with real-time automated detectors. To facilitate the participation of citizen scientists in studies of signals from southern resident killer whales and other soniferous organisms of the Salish Sea, in 2017–2018, we developed a low-cost combination of hardware and open-source software to stream audio data from our hydrophones to the headphones of our listeners. The data are archived at and stream from AWS/S3 in real-time (<60 seconds latency) to a browser-based player that is platform- and device-independent. This sets the stage for human detection of signals—mainly the calls, whistles, or clicks of fish-eating orcas—but also novel signals for which we do not yet have classifiers. We will demonstrate the Orcasound app and explain how it interacts synergistically with automated detectors and classifiers, as well as crowdsourced citizen science projects.

1:20

2pAO2. Deep learning for ethoacoustical mapping: Application to a single Cachalot long term recording on joint observatories in Vancouver Island. Herve Glotin (LIS CNRS Univ. Toulon, USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr), Paul Spong, Helena Symonds (Orcalab, Alert Bay, BC, Canada), Vincent Roger (LIS CNRS Univ. Toulon, Toulon, France), Randall Balestriero (Rice Univ., Houston, TX), Maxence Ferrari, Marion Poupard (LIS CNRS Univ. Toulon, Toulon, France), Jared Towers (Fisheries & Oceans Canada, Alert Bay, BC, Canada), Scott Veirs (Orcasound.net, Seattle, Washington), Ricard Marxer, Pascale Giraudet (LIS CNRS Univ. Toulon, Toulon, France), james pilkinton (Fisheries & Oceans Canada, Victoria, BC, Canada), Val Veirs (Orcasound.net, Friday Harbor, Washington), Jason Wood (SMRU, Friday Harbor, Washington), John Ford (Fisheries & Oceans Canada, Victoria, BC, Canada), and Thomas Dakin (Univ. of Victoria, Victoria, BC, Canada)

During February and March, 2018, a lone sperm whale known as Yukusam was recorded first by Orcalab in Johnstone Strait and subsequently on multiple hydrophones within the Salish Sea [1]. We learn and denoise these multichannel clicks trains with AutoEncoders Convolutional Neural Net (CNN). Then, we build a map of the echolocations to elucidate variations in the acoustic behavior of this unique animal over time, in different environments and distinct levels of boat noise. If CNN approximates an optimal kernel decomposition, it requires large amounts of data. Via spline functionals we offer analytics kernels with learnable coefficients do reduce it. We [1-3] identify the analytical mother wavelet to represent the input signal to directly learn the wavelet support from scratch by gradient descend on the parameters of cubic splines [2]. Supplemental material http://sabiod.org/yukusam [1] Balestriero, Roger, Glotin, Baraniuk, Semi-

Supervised Learning via New Deep Network Inversion, arXiv preprint arXiv:1711.04313, 2017 [2] Balestriero, Cosentino, Glotin, Baraniuk, WaveletNet : Spline Filters for End-to-End Deep Learning, Int. Conf. on MachineLearning, ICML, Stockholm, http://sabiod.org/bib, 2018 [3] Spong P., Symonds H., *et al*., Joint Observatories Following a Single male Cachalot during 12 weeks—The Yukusam story, ASA 2018.

1:40

2pAO3. Applying machine-learning based source separation techniques in the analysis of marine soundscapes. Tzu-Hao Lin (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., 2-15, Natsushima, Yokosuka, Kanagawa 237-0061, Japan, schonkopf@gmail.com), Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan), Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan), and Katsunori Fujikura (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., Yokosuka, Japan)

Long-term monitoring of underwater soundscapes provides us a large number of acoustic recordings to study a marine ecosystem. Characteristics of a marine ecosystem, such as the habitat quality, composition of marine fauna, and the level of human interference, may be analyzed using information relevant to environmental sound, biological sound, and anthropogenic sound. Supervised source separation techniques have been widely employed in speech and music separation tasks, but it may not be practical for the analysis of marine soundscapes due to the lack of a database that includes a large mount of paired pure and mixed signals. Even when the paired data is not available, different sound sources with unique spectral or temporal patterns may still be separated by apply semi-supervised or unsupervised learning algorithms. In this presentation, supervised and unsupervised source separation techniques will be demonstrated on long-term spectrograms of a marine soundscape. Separation performances under different levels of simultaneous source influence will also be discussed. In the future, more advanced techniques of source separation are necessary to facilitate the soundscape-based marine ecosystem sensing. An open database of marine soundscape will promote the development of machine learning-based source separation. Therefore, we will open acoustic data tested in this presentation on the Asian Soundscape to encourage the open science of marine soundscape.

2:00

2pAO4. Identification of fish species in estuaries and rivers using recorded soundscape with supervised machine learning. Mohsen Badiey and Javier Garcia-Frias (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu)

Identification of various fish species in estuarine and riverine environments is an important ecological problem at the intersection of various disciplines, including underwater acoustics, signal processing, and fisheries. Advancements in this field can be made by applying machine learning and deep learning technologies. From the vast amount of existing work on machine learning techniques, of special relevance to this problem are those aiming at the detection and labeling of speakers on overlapping speech, where the existing literature is much more reduced. In this paper, we present a dataset collected with a passive recorder placed on a tripod in Delaware Bay estuary during summer 2013 to continuously measure environmental soundscapes for several weeks. By using a library of mid-Atlantic region fish sounds as prior knowledge, we adapt the aforementioned techniques to the detection and identification of various recorded sounds. This adaptation is challenging, as the statistical properties of fish sounds are not the same as those of speech, and requires careful research on how to perform data pre-processing, feature extraction, and model development. The application of the proposed machine learning techniques to future observations will enhance the potential for long-term autonomous ecological studies in bays and rivers.

2:20–2:35 Break

Contributed Papers

2:35

2pAO5. Temporal patterns in Pacific white-sided dolphin communication at Barkley Canyon, with implications for multiple populations. Kristen S. Kanes, Stan E. Dosso (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, kristen.kanes@gmail.com), Tania L. Insua (Ocean Networks Canada, Victoria, BC, Canada), and Xavier Mouy (JASCO Appl. Sci., Victoria, BC, Canada)

Acoustic data collection is a cost-effective approach to evaluating activity patterns of otherwise challenging-to-study offshore cetaceans. However, manual analysis of acoustic data is time-consuming and impractical for large data sets. This study evaluates diel and seasonal patterns in Pacific white-sided dolphin communication through automated analysis of 20 months of continuous acoustic data collected from the Barkley Canyon node of Ocean Networks Canada's NEPTUNE observatory, offshore of Vancouver Island, British Columbia, Canada. In this study, cetacean signals are manually annotated in a sub-set of the data, and 94 time and frequency features of these and other sounds are extracted and used to train random forest classifiers targeting Pacific white-sided dolphin pulsed calls. The performance of binary and multiclass models with various forest sizes, minimum leaf sizes, and confidence thresholds for Pacific white-sided dolphin classification are compared through nested 10-fold cross-validation to select the best model. Vocalizations are classified with the resultant classifier, manually verified, and examined for seasonal and diel patterns. Pacific white-sided dolphins are shown to be vocally active during dawn and day in spring and summer and at dusk and night year-round with reduced overall activity in fall and winter, suggesting both consistent nocturnal and migratory diurnal populations.

2:50

2pAO6. Towards the topology of autoencoder of calls versus clicks of marine mammal. Vincent Roger, Maxence Ferrari, Ricard Marxer (Univ. Toulon, AMU, CNRS, LIS, Toulon, France), Faicel Chamroukhi (LMNO UMR CNRS 6139, Caen, France), and Herve Glotin (Univ. Toulon, AMU, CNRS, LIS, USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr)

The goal is to learn the features and the representation adapted for cetacean sound dynamics without any priors. Thus, we develop data driven model to generate voicing and click of cetaceans audio signals. We learn representation and features of stationary or nonstationary emission using neural network from raw audio. We use different types of convolutions (causal, with strides, with dilation [1]), or gradient inversion [2]. Experiments are conducted on various kind of calls of humpback whales from nips4b challenge [3] or Orca whale. We compare the topology for transient encoding on Physeters and Inia g. For each model, we detail the resulting

filters and discuss on the topology. We acknowledge Region PACA and NortekMED for Roger's Phd grant, & DGA and Région Haut de France for Ferrari's Phd grant. [1] Oord, Dieleman, Zen, Simonyan, Vinyals, Graves *et al*. Wavenet : A generative model for raw audio, arXiv:1609.03499, 2016 [2] Balestriero, Roger, Glotin, Baraniuk, Semi-Supervised Learning via New Deep Network Inversion, arXiv:1711.04313, 2017 [3] Glotin, LeCun, Mallat *et al*. Proc. 1st wkp on Neural Information Processing for Bioacoustics NIPS4B, joint to NIPS Alberta USA, 2013 http://sabiod.org/nips4b/challenge2.html, http://sabiod.org/NIPS4B2013_book.pdf

3:05

2pAO7. PyEcholab: An open-source, python-based toolkit to analyze water-column echosounder data. Carrie C. Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), Rick Towler (Alaska Fisheries Sci. Ctr., NOAA NMFS, Seattle, WA), Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Boulder, CO), Randy Cutter (Southwest Fisheries Sci. Ctr., NOAA NMFS, La Jolla, CA), and J. Michael Jech (Northeast Fisheries Sci. Ctr., NOAA NMFS, Woods Hole, MA)

Water-column echosounder data are widely used for a diversity of research objectives from fisheries abundance to identifying methane seeps to characterization of the seafloor. However, these data are voluminous, complex, and recorded in instrument-specific binary file formats. Tools to process these data are limited to a few commercial applications or custom programs developed by researchers, leaving potential users without sufficient financial resources or programming knowledge to use the data. To address this problem, NOAA Fisheries, University of Colorado Boulder's Cooperative Institute for Research in Environmental Sciences, and NOAA National Centers for Environmental Information (NCEI) scientists are developing PyEcholab. PyEcholab is an open-source, python-based system for reading, processing, and visualizing water-column echosounder files. Currently, the system is being developed to meet existing NCEI processing and visualization needs but PyEcholab's base classes and open architecture provide a framework for developing new file readers, processing algorithms, and visualization techniques that can be modularized into the system. The long-term vision is to engage the community in an open-source effort that continually grows PyEcholab's capabilities and expands the accessibility of water-column echosounder data.

Invited Paper

3:20

2pAO8. Toward scalable, reproducible, and open ocean acoustic research. Valentina Staneva, Amanda Tan (eSci. Inst., UW-IT, Univ. of Washington, Campus Box 351570, 3910 15th Ave. NE, Seattle, WA 98105, vms16@uw.edu), Divya Panicker (Biological Oceanogr., Univ. of Washington, Seattle, WA), and Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

As our ability to collect long-term acoustical signals grows we face the challenges of analyzing large datasets, ensuring our findings are reproducible, and sharing our work with collaborators. In this talk, we will give an overview of frameworks and tools that can facilitate those tasks. We will show how the Jupyter Notebook environment serves as a hub for quick prototyping, literate programming, interactive analysis and visualization, and a gateway to cloud computing. We will provide examples of processing of larger-than-memory ocean acoustic recordings from the comfort of our own laptop. We will further outline best practices for sharing code and data, and discuss the needs and approaches to achieve scalable, reproducible, and open acoustic research.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 SIDNEY (VCC), 1:00 P.M. TO 3:45 P.M.

Session 2pBAa

Biomedical Acoustics and Physical Acoustics: Shock Waves and Ultrasound for Calculus Fragmentation

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Pennsylvania State University, Penn State, 201E Applied Sciences Building, University Park, PA 16802

Michael R. Bailey, Cochair
Applied Physics Lab, University of Washington, Center for Industrial and Medical Ultrasound, APL-UW, Seattle, WA 98105

1:00

2pBAa1. Update on clinical trials results of kidney stone repositioning and preclinical results of stone breaking with one ultrasound system.
Michael R. Bailey, Yak-Nam Wang, Wayne Kreider, Bryan W. Cunitz (Appl. Phys. Lab, Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, APL-UW, Seattle, WA 98105, mbailey@uw.edu), Jessica Dai, Jonathan Harper, Helena Chang, Matthew D. Sorensen (Dept. of Urology, Univ. of Washington, Seattle, WA), Ziyue Liu (Dept. of Biostatistics, Indiana University-Purdue Univ. Indianapolis, Indianapolis, IN), Oren Levy (SonoMotion, Inc., San Francisco, CA), Barbrina Dunmire (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA)

Our goal is an office-based, handheld ultrasound system to target, detach, break, and/or expel stones and stone fragments from the urinary space to facilitate natural clearance. Repositioning of stones in humans (maximum 2.5 MPa, and 3-second bursts) and breaking of stones in a porcine model (maximum 50 cycles, 20 Hz repetition, 30 minutes, and 7 MPa peak negative pressure) have been demonstrated using the same 350-kHz probe. Repositioning in humans was conducted during surgery with a ureteroscope in the kidney to film stone movement. Independent video review confirmed stone movements ($\geq$ 3 mm) in 14 of 15 kidneys (93%). No serious or unanticipated adverse events were reported. Experiments of burst wave lithotripsy (BWL) effectiveness on breaking human stones implanted in the porcine bladder and kidney demonstrated fragmentation of 4 of 4 stones on post mortem dissection. All clinical pathology, hematology, and urinalysis for a 1-week survival study with the BWL exposures in 10 specific pathogen free pigs were within normal limits. These results demonstrate that repositioning of stones with ultrasonic propulsion and breaking of stones with BWL are safe and effective. [Work supported by NIH P01 DK043881, K01 DK104854, and R44 DK109779.]

1:15

2pBAa2. Acoustic radiation force acting on a spherical scatterer in water: Measurements and simulation.
Maria M. Karzova, Anastasiya V. Nikolaeva, Sergey Tsysar (Phys. Faculty, Moscow State Univ., Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Vera Khokhlova, and Oleg Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle, USA; Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Acoustic radiation force (ARF) is successfully used in a recently proposed ultrasonic technology for kidney stone propulsion. The planning of the treatment requires calibration of ARF for stones of different dimensions and locations. However, such calibration remains a problem. Here, a method of measuring ARF acting on a mm-sized spherical object positioned on the axis of a focused ultrasound beam is proposed and tested. Acoustic field was generated by a single-element 1.072 MHz transducer of 100 mm aperture and 70 mm focal length positioned at the bottom of the water tank. Measurements were performed for nylon and glass spherical scatterers with diameters from 2 to 4 mm located at different distances from the source along the vertical beam axis. For each scatterer, the source power was gradually decreased until the scatterer started to fall down from the trap. At this threshold power of the source, the value of ARF was determined as a difference between the gravity and buoyancy forces acting on the scatterer. At other source power outputs, the value of ARF was linearly scaled. ARF was also calculated from pressure distributions reconstructed from acoustic hologram of the source and physical parameters of the scatterers. Experimental and theoretical results were found in a good agreement. It was shown that the most effective pushing was observed at distances where the beam was slightly wider than the scatterer. [Work supported by the stipend of the President of Russia (SP-2621.2016.4), RFBR №18-32-00659, and NIH P01 DK43881.]

1:30

2pBAa3. Generation of guided waves during burst wave lithotripsy as a mechanism of stone fracture.
Adam D. Maxwell (Urology, Univ. of Washington School of Medicine, 1013 NE 40th st, Seattle, WA 98105, amax38@u.washington.edu), Brian MacConaghy, Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg Sapozhnikov (Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation)

Burst wave lithotripsy (BWL) is an experimental method to noninvasively fragment urinary stones by short pulses of focused ultrasound. We are investigating physical mechanisms of stone fracture during BWL to better optimize this procedure. In this study, we used photoelasticity imaging as a method to visualize elastic wave dynamics in model stones. Epoxy and glass stone models were made into cylindrical, rectangular, or irregular geometries and exposed in a degassed water bath to focused ultrasound bursts at different frequencies. A high-speed camera was used to record images of the stone during exposure through a circular polariscope backlit by a monochromic flash source. Results showed development of periodic stresses in the stone body with a pattern dependent on frequency. These were identified as guided wave modes in cylinders and plates, which formed standing waves upon reflection from the distal surfaces of the stone model, causing periodic stress positions. Measured phase velocities compared favorably to specific numerically calculated modes dependent on frequency and material. Artificial stones exposed to BWL produced cracks at positions anticipated by this mechanism. These results support guided wave production and reflection as a mechanism of stone fracture in BWL. [Work supported by K01 DK104854 and P01 DK043881.]

1:45

2pBAa4. Impact of stone characteristics on cavitation in burst wave lithotripsy.
Christopher Hunter (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 ne 40th St., Seattle, WA 98105, chunter6@uw.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan Cunitz, Barbrina Dunmire (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Matthew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), James C. Williams (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), Michael Bailey, Akshay Randad, and Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Non-invasive kidney stone treatments such as shock wave lithotripsy (SWL) and burst wave lithotripsy (BWL) rely on the delivery of pressure waves through tissue to the stone. In both SWL and BWL, the potential to hinder comminution by exciting cavitation proximal to the stone has been reported. To elucidate how different stones alter prefocal cavitation in BWL, different natural and synthetic stones were treated *in vitro* using a therapy transducer operating at 350 kHz (peak negative pressure 7 MPa, pulse length 20 cycles, pulse repetition frequency 10 Hz). Stones were held in a confined volume of water designed to mimic the geometry of a kidney calyx, with the water filtered and degassed to maintain conditions for which the cavitation threshold (in the absence of a stone) matches that from *in vivo* observations. Stone targeting and cavitation monitoring were performed via ultrasound imaging using a diagnostic probe aligned coaxially with the therapy transducer. Quantitative differences in the extent and location of cavitation activity were observed for different stone types—e.g., "softer" stones (natural and synthetic) that disintegrate into dusty fragments produced larger prefocal cavitation clouds. Future work will focus on correlation of such cavitation metrics with stone fragmentation. [Funding support by NIH P01-DK043881, K01-DK104854.]

Invited Paper

2:00

2pBAa5. Modeling and numerical simulation of the bubble cloud dynamics in an ultrasound field for burst wave lithotripsy.
Kazuki Maeda (Univ. of Washington, Seattle, WA), Tim Colonius (California Inst. of Technol., 1200 E. California Blvd., Pasadena, CA 91125, colonius@caltech.edu), Adam D. Maxwell, Wayne Kreider, and Michael R. Bailey (Univ. of Washington, Seattle, WA)

Modeling and numerical simulation of bubble clouds induced by intense ultrasound waves are conducted to quantify the effect of cloud cavitation on burst wave lithotripsy, a proposed non-invasive alternative to shock wave lithotripsy that uses pulses of ultrasound with an amplitude of $O(1)$ MPa and a frequency of $O(100)$ kHz. A unidirectional acoustic source model and an Eulerian-Lagrangian method are developed for simulation of ultrasound generation from a multi-element array transducer and cavitation bubbles, respectively. Parametric simulations of the spherical bubble cloud dynamics reveal a new scaling parameter that dictates both the structure of the bubble cloud and the amplitude of the far-field, bubble-scattered acoustics. The simulation further shows that a thin layer of bubble clouds nucleated near a kidney stone model can shield up to 90% of the incoming wave energy, indicating a potential loss of efficacy during the treatment due to cavitation. Strong correlations are identified between the far-field, bubble-scattered acoustics and the magnitude of the shielding, which could be used for ultrasound monitoring of cavitation during treatments. The simulations are validated by companion experiments *in* vitro. [The work was supported by NIH P01-DK043881 and K01-DK104854.]

Contributed Papers

2:15

2pBAa6. Burst wave lithotripsy: An *in vivo* demonstration of efficacy and acute safety using a porcine model.
Yak-Nam Wang, Wayne Kreider, Christopher Hunter, Bryan Cunitz, Jeffrey Thiel, Frank Starr, Jessica Dai, Yasser Nazari, Donghoon Lee (Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, ynwang@uw.edu), James C. Williams (Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), Michael R. Bailey, and Adam D. Maxwell (Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a new non-invasive method for stone comminution using bursts of sub-megahertz ultrasound. A porcine model of urolithiasis and techniques to implement BWL treatment have been developed to evaluate its effectiveness and acute safety. Five human calcium oxalate monohydrate stones (6–7 mm) were hydrated, weighed, and surgically implanted into the kidneys of three pigs. Transcutaneous stone treatments were performed with a BWL transducer coupled to the skin via an external water bath. Stone targeting and treatment monitoring were performed with a co-aligned ultrasound imaging probe. Treatment exposures were applied in three 10-minute intervals for each stone. If sustained cavitation in the parenchyma was observed by ultrasound imaging feedback, treatment was paused and the pressure amplitude was decreased for the remaining time. Peak negative focal pressures between 6.5 and 7 MPa were applied for all treatments. After treatment, stone fragments were removed from the kidneys. At least 50% of each stone was reduced to $<$2 mm fragments. 100% of four stones were reduced to $<$4 mm fragments. Magnetic resonance imaging showed minimal injury to the functional renal volume. This study demonstrated that BWL can be used to effectively fragment kidney stones with minimal injury. [Funding support: NIH P01-DK043881, NIH K01-DK104854.]

2:30–2:45 Break

2:45

2pBAa7. Design of a transducer for fragmenting large kidney stones using burst wave lithotripsy.
Akshay P. Randad (Mech. Eng. Dept., Univ. of Washington, Stevens Way, Box 352600, Seattle, WA 98195, aprandad@u.washington.edu), Mohamed A. Ghanem (Dept. of Aeronautics and Astronautics, Univ. of Washington, Seattle, WA), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, APL, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a potential noninvasive treatment for breaking kidney stones. BWL requirements of high-pressure output, limited aperture for acoustic window, and specific focal length and frequency constrain the focal beam width. However, BWL is most effective only on stones smaller than the beam width. We tested a porous piezoelectric material (PZ36) to increase the output power and designed acoustic lenses that broaden the beam. A weighted iterative angular spectrum approach was used to calculate the source phase distribution needed to generate desired cross sectional focal beam profiles each of 12 mm width. The phase calculations were then 3D printed as holographic lenses placed over a circular aperture of 80-mm diameter, 350 kHz PZ36 to produce the desired beam at 85 mm depth. The difference in simulated beam width and that measured by hydrophone was $<$1 mm, and the structural-similarity index value was greater than 0.65. The differences in structures were due not to shape and size of the 6-dB contours but to amplitude distribution within the contour. In conclusion, this design approach combined with 3D printing provide a way to tailor focal beam profiles for lithotripsy transducers. [The work was supported by NIDDK P01-DK043881 and K01-DK104854.]

3:00

2pBAa8. Acoustic bubble coalescence and dispersion for enhanced shockwave lithotripsy.
Hedieh Alavi Tamaddoni and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, 2135 Carl A. Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, alavi@umich.edu)

The goal of this study is to develop methods to improve shockwave lithotripsy (SWL). Although cavitation on the surface of a stone may aid fragmentation, cavitation away from the stone may block subsequent shockwaves or cause tissue damage. In previous work, we have shown low amplitude acoustic bursts immediately *after* each shockwave can force cavitation bubbles to coalesce enhancing SWL efficacy. In this study we examined the feasibility of acoustically dispersing bubbles away from the propagation path immediately *before* the arrival of the next shockwave. A clinical Dornier lithotripter was used with an in-house made transducer to generate Acoustic Bubble Coalescence and Dispersion (ABCD) pulses fired at different timing with respect to each shockwave. Model stones were treated with 2500 shockwaves at 30 shocks/min or 120 shocks/min and four different ABCD pulse sequences *in-vitro*. Results showed fragments larger than 2 mm were significantly reduced for all four ABCD sequences cases at 120 shocks/min. The remnant mass of fragments larger than 2 mm was 0.16% at low rate of 30 shocks/min, and 15.81% at 120 shocks/min without ABCD, and 0.19% at 120 shocks/min with ABCD. These results suggest that dispersing residual bubbles can aid in fragmentation efficiency allowing faster SWL treatments.

3:15

2pBAa9. Confocal lens focused piezoelectric lithotripter.
Gilles Thomas, Jean-Yves Chapelon, Alain Birer, and Cyril Lafon (U1032, INSERM, 151, cours Albert Thomas, Lyon 69424, France, gilles.thomas@inserm.fr)

Usually, piezoelectric lithotripters consist of a high quantity of small flat piezoelectric discs tied to a spherical structure, where the focus is the

geometric center of the sphere. The small diameter of the ceramics compared to the distance to the focus of the lithotripter means that only a fraction of the surface pressure will arrive to the focus, and also results in a very small focal diameter. This work focus on the evaluation of a new type of piezoelectric lithotripter with similar dimensions of a commercial lithotripter and composed of either 3 or 4 large lens focused piezoelectric transducers with focal pressure up to 25 MPa each, set either in a confocal C-shape or confocal ring-shape. Each transducer is made with a 92 mm diameter flat piezoelectric ceramic disc of 220, 300, or 400 kHz thickness frequency and the acoustic lens shape was calculated using finite element optimization in order to maximize its focusing capability. Comparison of artificial stone comminution efficiency depending on the frequency, pressure, and the setup of the piezoelectric transducers were made and compared to commercially available lithotripters. [Work supported by an industrial grant from EDAP-TMS.]

3:30

2pBAa10. Experimental observations and numerical modeling of lipid-shell microbubbles with stone targeting moieties for minimally-invasive treatment of urinary stones. Yuri A. Pishchalnikov, William Behnke-Parks (Applaud Medical Inc., 953 Indiana St., San Francisco, CA 94107, yurapish@gmail.com), Kazuki Maeda (Dept. of Mech. Eng., Univ. of Washington, Seattle, WA), Tim Colonius (Dept. of Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Matt Mellema, Matt Hopcroft, Alice Luong (Applaud Medical Inc., San Francisco, CA), Scott Wiener, Marshall Stoller (Dept. of Urology, Univ. of California, San Francisco, CA), Thomas Kenny (Dept. of Mech. Eng., Stanford Univ., Stanford, CA), and Daniel Laser (Applaud Medical Inc., San Francisco, CA)

Products incorporating stone-targeting microbubbles have recently entered human clinical trials as a new minimally-invasive approach to treat urinary stones. Lipid-shell, gas-core microbubbles can be introduced into the urinary tract through a catheter. Calcium-binding moieties incorporated into the lipid shell can facilitate binding to stones. The microbubbles can be excited by an extracorporeal source of low-intensity ultrasound. Alternatively, the microbubbles can be excited by an intraluminal source, such as a fiber-optic laser. With either excitation technique, stone-targeting microbubbles can significantly increase rates of erosion, pitting, and fragmentation of stones, as has recently been reported for *in-vitro* experiments with synthetic stones [Wiener *et al*., J. Urology, v.199, no.4S, e322 (2018)]. We report here on new experiments using high-speed photography to characterize microbubbles expansion of cracks within a stone and resultant breaking-off of stone fragments. Numerical modeling shows that the direction of micro-jets produced by collapsing stone-bound microbubbles depends strongly on bubble shape and stand-off distance. For a wide range of stand-off distances and bubble shapes, microbubble collapse is associated with pressure increases of some two orders of magnitude compared to the excitation source pressures. This *in-vitro* study provides key insights into the use of stone-targeting microbubbles in treatment of urinary stones.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 COLWOOD 1/2 (VCC), 1:00 P.M. TO 5:10 P.M.

Session 2pBAb

Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications II

Guillaume Haiat, Cochair
Multiscale Modeling and Simulation Laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Pierre Belanger, Cochair
Mechanical Engineering, Ecole de technologie superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K/, Canada

Invited Papers

1:00

2pBAb1. Understanding the influence of grain scattering noise on array imaging and defect characterisation. Bruce W. Drinkwater, Long Bai, and Alexander Velichko (Mech. Eng., Univ. of Bristol, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

The performance of ultrasonic inspection degrades for coarse grained materials because of the multiple scattering effects caused by the material microstructure. The influence of grain scattering noise on detection and characterisation of defects is studied in this paper. Two parameters, mean grain size and grain size variation, are used to describe a particular grain structure, which is included in a finite element (FE) model to simulate the array data of a defect with the presence of grain noise. The ultrasonic attenuation and multiple

scattering effects become more significant at high frequencies, and so a low frequency is shown to be preferable for defect detection. However, the frequency should also be higher than a certain limit for the defect to be characterisable based on its scattering matrix. Here we extract the noise scattering matrix due to the grain scattering from the simulated array data and use its statistical distribution is used to determine the characterisation uncertainty. Preliminary results obtained at 2.5 MHz for stainless-steel show that 1 mm (i.e., 0.45λ) cracks can be sized reliably when the mean grain size is less than 0.2 mm. However, when the mean grain size exceeds 0.15 mm, it is difficult to distinguish the 1 mm cracks from volumetric defects of the same size.

1:20

2pBAb2. Ultrasonic characterization of synthetic three-dimensional polycrystals. Musa Norouzian, Nathanial Matz, Showmic Islam, and Joseph A. Turner (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

Ultrasonic nondestructive testing has been increasingly used to characterize heterogeneities of polycrystalline materials. With such techniques, the interactions of coherent ultrasonic waves with grain boundaries result in scattering. Such scattered waves carry information regarding the physical properties of the scatterer. Therefore, microstructural information can be obtained by quantifying the scattered response. Current diffuse ultrasonic backscatter models include several assumptions about the macroscopic and microscopic properties of the polycrystals. In this presentation, the sensitivity of grain size characterization to such assumptions is investigated using simulated microstructures. Several polycrystals with cubic crystal symmetry and randomly oriented grains are simulated using Dream.3D. This study applies the single scattering model in which the longitudinal-to-longitudinal configuration is considered for the incident and the scattered waves and limited to the weakly-scattering regime. In each configuration, the theoretical results are compared with results from the synthetic volumes. The results demonstrate distinct differences between the theory and the simulation. For example, the theoretical scattering cross section for a Voigt-averaged copper polycrystal at 15 MHz is found to be about three times larger than the value based on the Dream.3D microstructure. Finally, the influence of grain size distribution and grain elongation on the ultrasonic models are investigated.

1:40

2pBAb3. Finite element modelling of scattering of an acoustic wave by particles in a fluid: Shear and thermal effects. Valerie J. Pinfield, Derek M. Forrester (Chemical Eng. Dept., Loughborough Univ.,Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk), and Jinrui Huang (Chemical Eng. Dept., Loughborough Univ., UK, Loughborough, Leicestershire, United Kingdom)

We investigate, using finite element modelling, the effects of viscous and thermal dissipation in the region around a particle in a liquid subjected to an acoustic field. The linearized thermo-acoustic equations for propagation in a viscous liquid are used in the model. In the Rayleigh scattering regime, with Mega-Hertz frequencies and nano- or micron-sized particles, the shear and thermal decay lengths are of order micrometres and are therefore challenging to resolve accurately using finite element models. We report investigations of the effect of particle size and frequency on the magnitude of the thermal and shear wave fields and their decay length and compare with analytical predictions. We present a determination of the viscous and thermal power dissipation around a single particle, and for small clusters of particles. Here we identify the effect of interaction of the shear and thermal fields with neighbouring particles through mode reconversion, leading to a reduction in dissipation losses when interparticle separation is of the order of the thermal or shear decay length.

Contributed Paper

2:00

2pBAb4. Quantitative assessment of angiogenic microvasculature using Ultrasound multiple scattering with microbubble contrast agents. Kaustav Mohanty (Mech. and Aerosp. Eng., North Carolina State Univ., 3147 B, 911 Oval Dr., College of Eng., EB-3, Raleigh, NC 27606, kmohant@ncsu.edu), Virginie Papadopoulou, Isabel Newsome, Paul A. Dayton (Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

In tumors, angiogenesis (formation of new blood vessels) is established as a biomarker of malignancy. A random, isotropic, high-density vessel network is linked to tumor invasiveness. Ultrasound quantification of angiogenic micro-architecture could help increase the specificity of ultrasound for cancer diagnosis. We exploit multiple scattering by microbubbles populating angiogenic networks to characterize an effective medium diffusivity via the diffusion constant (D). The inter-element response matrix is acquired using an L11-4v linear array transducer operating at 7.8 MHz. D is computed from the time evolution of the incoherent contribution to the backscattered intensity in the near field. D is measured in fibrosarcoma tumors subcutaneously implanted in rats ($n = 16$), and in control, healthy tissue ($n = 18$). D is measured for two different orientations (Coronal and Transverse) and the anisotropy of the microvasculature is evaluated via the ratio of the D values obtained in the two different orientations. D was found significantly different between control (1.38 ± 0.51 mm$^2/\mu$s) and tumor (0.65 ± 0.27 mm$^2/\mu$s) ($p<0.01$) and anisotropy of the angiogenic network was observed in control cases (1.62 ± 0.911). For further validation, these results were corroborated with vascular density measurements from acoustic angiography data, confirming increased vessel density in tumors compared to controls.

Invited Papers

2:15

2pBAb5. Experimental assessment of guided waves far-field scattering around damage in metallic and composite structures. Patrice Masson, Nicolas Quaegebeur (Mech. Eng. Dept., Universite de Sherbrooke, 2500 Blvd. Universite, Sherbrooke, QC J1K 2R1, Canada, Patrice.Masson@USherbrooke.ca), Mohammad H. Sherafat, Larry Lessard, and Pascal Hubert (Mech. Eng., McGill Univ., Montreal, QC, Canada)

Experimental damage simulation is useful for designing ultrasonic guided-wave based systems for non-destructive evaluation (NDE) and structural health monitoring (SHM). However, simulating the scattering of guided waves with geometrical (rivets, thickness changes, stiffeners, and extrusions) or damage features (fatigue cracks, fillet cracks, delaminations, and disbonds) remains a challenge. The objective of this work is to assess to which extent the interaction of ultrasonic guided waves with typical damage can be captured with an experimental model for a metallic structure and a composite structure. For the metallic structure, real fatigue cracks around a rivet hole are simulated by machined notches, while, for the composite structure, the impact damage is simulated by a single artificial delamination introduced into the laminate using two circular Teflon tapes during manufacturing. This paper implements an experimental methodology for estimating the far-field scattering for both simulated and real damage. Two co-localized rectangular piezoceramics are used to generate the guided waves and non-contact measurement is performed using a three-dimensional laser Doppler vibrometer (3D-LDV) to extract the required information for evaluation of the reflection, transmission, as well as the scattering behavior of the waves. The corresponding coefficients as a function of frequency, incident angle, and type of damage are extracted.

2:35

2pBAb6. Interaction of guided waves with defects in a multilayered cylinder by an asymptotic approach. Aditya Krishna (I2M, CEA, Talence, Gironde, France), Eric Ducasse (I2M Arts et Metiers ParisTech, Talence, Gironde, France), and Marc Deschamps (I2M, Universite Bordeaux, 351 cours de la liberation, Talence, Gironde 33400, France, marc.deschamps@u-bordeaux.fr)

This work deals with the problem of the propagation of an elastodynamic field radiated by a source in a cylindrical layered medium which interacts with and is diffracted by a defect. At low frequencies, where the defect size is much smaller than the wavelength, this interaction can be approximated by a point source, located at the defect position. This secondary source is expressed by the Green function and its derivative. The Green function, which describes the response of an undamaged cylinder, is calculated using the canonical form of the wave equation initially expressed as a function of the spatial and temporal variables. Performing the Laplace in time and Fourier transforms along the cylinder axis, this equation is written as an ordinary differential equation with respect to the radial position. The solution for the transversely isotropic case is obtained by adopting the partial wave formulation, expressed as a combination of the modified Bessel's functions of the first and second kind. Having assembled the layers, numerical inverse transforms are performed to obtain the real wave fields. This technique allows for reduced computational costs and faster calculation times and could be used for the non destructive testing of embedded pipes and tubes.

2:55–3:10 Break

3:10

2pBAb7. Harmonic generation and frequency mixing at nonlinear imperfect interfaces. Shiro Biwa (Dept. of Aeronautics and Astronautics, Kyoto Univ., C-Cluster III, Katsura, Nishikyo-ku, Kyoto 615-8540, Japan, biwa@kuaero.kyoto-u.ac.jp)

Imperfect interfaces can be found in a variety of natural and artificial systems. Some examples of such interfaces that are objects of ultrasonic evaluation and diagnostics include contacting mechanical components as well as closed defects in structural components. A remarkable feature of these interfaces is that they exhibit highly nonlinear responses to ultrasonic waves. Harmonic generation (generation of, e.g., double frequency signals at the incidence of a wave with a certain frequency) and frequency mixing (generation of sum/difference frequency signals at the incidence of waves of two different frequencies) are typical phenomena at these interfaces. In this presentation, a theoretical analysis of these phenomena is presented based on modeling the interface as a nonlinear spring-type interface between two similar linearly elastic media. Unlike the corresponding nonlinear acoustic effects due to material nonlinearity, the interfacial nonlinear effects considered here do not require the spatial accumulation of nonlinearly generated harmonic or sum/difference frequency components. The propagation directions and amplitudes of these nonlinearly generated components are derived for plane mono- or dichromatic incident waves. Some relevant experimental results for a contacting interface of metallic blocks subjected to different applied pressures are also presented.

Contributed Papers

3:30

2pBAb8. Interfacial effects in underwater acoustic panel measurements. Jeffrey Szabo and Adam Bent (Atlantic Res. Ctr., Defence Res. and Development Canada, PO Box 1012, 9 Grove St., Dartmouth, NS B2Y3Z7, Canada, jeff.szabo@drdc-rddc.gc.ca)

Underwater acoustic panel measurements were conducted on single layers of rubber or metal, and on rubber/metal bilayers that had been fabricated using various adhesives or attachment methods. Reflection and transmission measurements were carried out on submerged panels using a parametric array source. It was found that the condition of the various interfaces in the system (water/rubber, water/metal, or rubber/metal) had a dominant effect on the measured acoustic properties. Single layer (metal or rubber) panels with smooth surfaces behaved in accordance with theoretical predictions, but panels with even a small amount of surface roughness did not. This was attributed to a non-ideal water/sample interface, most likely due to the presence of air bubbles. The effect of various surface treatments on the acoustic behaviour of the panels was investigated.

Different adhesion methods for fabricating aluminum/rubber bilayers were tested for their practicality and effects on underwater acoustic properties. Nitrile butadiene rubber and aluminum panels were adhered to one another using pressure sensitive adhesive, contact cement, banded duct tape, or epoxy, and tested in transmission and reflection. The results were strongly dependent on adhesion method, with only one method (epoxy cured under 15 psi pressure) giving results that matched theoretical predictions.

3:45

2pBAb9. Reflection of an ultrasonic wave from the bone-implant interface: A numerical study. Yoann Hériveaux (Laboratoire MSME, CNRS, UPEC - Fac des Sci. - Laboratoire MSME, 61, Ave. du général de Gaulle, Créteil 94010, France, yoann.heriveaux@u-pec.fr), Vu-Hieu Nguyen (Laboratoire MSME, Université Paris-Est, Creteil, France), and Guillaume Haiat (Laboratoire MSME, CNRS, Créteil, France)

Quantitative ultrasound are used to characterize and stimulate osseointegration processes at the bone-implant interface (BII). However, the interaction between an ultrasonic wave and the implant remains poorly understood. This study aims at investigating the sensitivity of the ultrasonic response to the microscopic and macroscopic properties of the BII and to osseointegration processes. The reflection coefficient R of the BII was modeled for different frequencies using a two-dimensional finite element model. The implant surface was modeled by a sinusoidal function with varying amplitude and spatial frequency and then by considering actual implant surface profiles. A soft tissue layer of thickness W was introduced between bone tissue and the implant in order to model non-mineralized fibrous tissue. For microscopic roughness, R is shown to increase from around 0.55 until 0.9 when $k.W$ increases from 0 to 1 and to be constant for $k.W>1$. These results show that R depends on the properties of bone tissue located at a distance comprised between 1 and 25 μm from the implant surface. For macroscopic roughness, R is highly dependent on the roughness amplitude, which may be explained by phase cancellation and multiple scattering effects for high roughness parameters.

4:00

2pBAb10. *In vitro* and *in vivo* comparison between methods based on quantitative ultrasound and on resonance frequency analysis to assess dental implant stability. Romain Vayron (Multiscale Modeling and Simulation Lab., CNRS, Créteil, France), Vu-Hieu Nguyen, and Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr)

Resonance frequency analyses (RFA) and quantitative ultrasound (QUS) methods have been suggested to dental assess implant stability. The aim of this study was to compare the results obtained using these two techniques *in vitro* and *in vivo*. Implants were inserted in bone phantoms with different values of density and cortical thickness to assess the effect of bone quality on the ultrasonic indicator (UI) and on the ISQ values. 81 identical implants were inserted in the iliac crests of 11 sheep. The QUS and RFA measurements were realized after different healing times. *ISQ* values increase and *UI* values decrease when i) the bone density and ii) cortical thickness increase. The error realized on the estimation of the trabecular density (respectively cortical thickness) with the QUS device is around 4 (respectively, 8) times lower compared to that made with the RFA technique. The error made on the estimation of the healing time using the QUS technique was 10 times lower than using the RFA technique. The results show that ultrasound technique provides a better estimation of different parameters related to the implant stability compared to the RFA technique, paving the way towards the development of a decision support system to dental surgeons.

Invited Paper

4:15

2pBAb11. Experimental behaviour of an acoustic wave in a medium comprising self-similar (fractal) objects. Elsy Mandelbrot (ENSAV, Versailles, France), Bernard Sapoval (LPMC, Ecole Polytechnique, Palaiseau, France), and Vincent Gibiat (ICA, Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr)

After the pioneering work of one of the authors (Sapoval *et al.* J. Acoust. Soc. Am. **104**, 1997), the study of acoustic propagation over fractal boundaries came up against the difficulty of experimenting with self-similar objects immersed in two- and three-dimensional space. The idea of one of us to use the capabilities offered by three-dimensional printing has made it possible to overcome this difficulty and produce numerous objects, as small or large as desired, in large numbers. We will present the experimental results obtained for acoustic propagation, reflection coefficient, attenuation and trapping in the case of an half fractal octahedron, isolated as well as in the case of a flat paving consisting of an important number of these objects.

Contributed Paper

4:35

2pBAb12. Phase velocity dispersion estimation of viscoelastic materials by two-point wavelet-based method. Piotr Kijanka (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, kijanka.piotr@mayo.edu), Lukasz Ambrozinski (Dept. of Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Ultrasound shear wave elastography (SWE) is a promising imaging modality used for noninvasive, quantitative evaluation of tissue mechanical properties. This method uses an acoustic radiation force to produce laterally propagating shear waves that can be tracked to obtain the wave velocity. One of the ways to explore the viscoelasticity is through examining the shear wave velocity dispersion curves. In this paper, we present an alternative method to the classical two-dimensional Fourier transform (2D-FT), a two-point wavelet transformation method. We use the Morlet wavelet function as the mother wavelet to filter two shear waves at different locations. We examined how starting distance from the push and distance between the two locations affected the shear wave velocity dispersion. We tested this method on digital phantom data created using local interaction simulation approach (LISA) in viscoelastic media and on data acquired from phantom experiments. We compared results from the two-point method with the 2D-FT technique. The two-point method provided dispersion curves estimation with lower errors over a wider frequency band. Tests conducted showed that the two-point technique gives results with better accuracy in simulation results and can be used to measure phase velocity of viscoelastic materials.

Invited Paper

4:50

2pBAb13. Ultrasonic and X-Ray tomography inspection of a woven glass reinforced composite damaged by fatigue loading. Nada Miqoi (LEM3 - UMR CNRS 7239, ENSAM - Arts et Métiers ParisTech, Metz, France), Pascal Pomarède, Nico F. Declercq (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France, nico.declercq@me.gatech.edu), Laurent Peltier, and Fodil Meraghni (LEM3 - UMR CNRS 7239, ENSAM - Arts et Métiers ParisTech, Metz, France)

An investigation of a polyamide 6.6/6 composite reinforced with woven glass fibers is described with a focus on its response to fatigue solicitations. The main purpose is to elaborate a quantitative and a qualitative study of the induced damage as well as its detectability. To do so, several nondestructive testing methods were used. The different damage mechanisms, relative to fatigue loading, were identified using X-Ray tomography as a reference for the ultrasonic investigations. As a result, each damage mechanism is visualized and a measure of the induced void is established for comparison. An ultrasonic investigation based on C-scans and guided waves is performed under different signal analysis approaches in an attempt to extract as much information as possible. The discoveries and experiences are important in the framework of a collaboration with the automotive industry en for biomechanical applications.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 — SALON A (VCC), 1:00 P.M. TO 3:30 P.M.

Session 2pED

Education in Acoustics: Measuring Educational Outcomes

Andrew A. Piacsek, Chair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Chair's Introduction—1:00

Invited Papers

1:05

2pED1. Measuring students' learning gains with pre/post assessment. John R. Buck (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu) and Kathleen E. Wage (ECE Dept., George Mason Univ., Fairfax, VA)

Pre/post protocols assess students with the same instrument at the start and end of a course to measure changes in their understanding. In contrast, traditional "final exam" assessment measures only the students' understanding but not their progress. Since pre/post testing measures learning gains, it is an essential tool for evaluating the effectiveness of different learning formats, e.g., standard lecture versus active learning. Concept inventories (CI's) are frequently administered as a pre/post assessment. A CI is a multiple-choice exam that emphasizes conceptual understanding over computational skills and has wrong answers designed to match common misconceptions. Inspired by Hake's [1998] pioneering study of pre/post test data from the Force Concept Inventory [Hestenes *et al*., 1992], we developed the Signals and Systems Concept Inventory (SSCI). In almost two decades since its development, scores of instructors have administered the SSCI to thousands of students in dozens of countries. Analyzing pre/post data from sixty-two signals classes at multiple institutions, we found students in active learning classes learned significantly more than students in lecture classes. Our findings are consistent with both Hake's pioneering study and with Freeman *et al*.'s 2014 meta-analysis of 225 studies that found that active learning classes reduced the failure rate by one-third.

1:25

2pED2. A new pre/post test to assess student mastery of introductory level acoustics and wave mechanics. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu)

The Force Concept Inventory is a widely used multiple-choice test that enables physics educators to assess the ability of students to apply basic concepts associated with Newton's laws and to identify specific misconceptions held by students. When administered at the beginning of a course in introductory physics, as well as at the end, the FCI provides a reliable measure of the *gain* in student understanding. The usefulness of the FCI, and its ease of use, has spurred the development of similar "inventories" and "baseline tests" for other subjects in the physical sciences, such as the Fluid Mechanics Concept Inventory, the Conceptual Survey in Electricity and Magnetism, and the Thermodynamics Concept Inventory. This presentation will describe a new concept inventory test designed to assess

students' conceptual understanding of the behavior of oscillators and mechanical waves, their understanding of the measurement and perception of sound, and their ability to interpret graphical representations of sound. Following the FCI model, this new Sound and Vibration Concept Inventory forces students to compare correct answers to those based on common misconceptions. Preliminary efforts to assess the reliability of this test will be described.

1:45

2pED3. Assessing the concept of resonance through the lens of discipline-based educational research. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

In the last few decades, many science and engineering fields have developed educational research programs specific to their fields. These discipline-based educational research programs are critical for improving the learning environments of the next generation of scientists and engineers. Acoustics is a highly interdisciplinary field, drawing from a wide variety of science and engineering fields. Although no major acoustics educational research program currently exists, acoustics educators should be aware of discipline-based educational research findings which most closely align to their particular branch of acoustics. Educational research can inform educators on how students develop content expertise as well as grow in their thinking about the practice of science and engineering. One of the ways that discipline-based educational research informs educators about the way students think about science content and practices is through the use of assessment instruments. Some assessment tools related to acoustics have been developed by the physics education community, although the selection remains somewhat limited. An attempt to develop an instrument for assessing the concept of resonance as well as the challenges for validating the instrument is presented.

2:05

2pED4. Using and assessing mechanical wave tutorials in introductory physics. Jack Dostal (Phys. Dept., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

The Tutorials in Introductory Physics curriculum produced by the University of Washington Physics Education Group includes tutorials on mechanical waves. The tutorials are written in the style of a Socratic dialogue on topics like superposition, reflection, and transmission of waves. These tutorials have a physics education research basis, leading the student to address common difficulties and misconceptions that have been broadly demonstrated by other students. The tutorials also come with homework assessments and pretest/posttest questions. I will describe my use of these and other tutorials in my own classes, both in standard calculus-based introductory physics courses, as well as in a non-major physics of music class.

2:25

2pED5. Measuring educational outcomes for ABET accreditation of the undergraduate acoustical engineering programs at the University of Hartford. Eoin A. King and Robert Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu)

Located in West Hartford, CT, USA, the University of Hartford has two ABET-accredited acoustical engineering programs: the Bachelor of Science in Mechanical Engineering (BSME) with Acoustics Concentration and a unique interdisciplinary Bachelor of Science in Engineering with a major in Acoustical Engineering & Music. The national Accreditation Board for Engineering & Technology accreditation process requires that an institution identify and provide evidence of Program Educational Objectives, Student Outcomes, Required Curricular Components, Relevant Facilities, Faculty Expertise, and Methods of Continuous Improvement. The differences and key points of these criteria will be discussed, along the manner by which each are measured. The results of a recent ABET accreditation visit will be described.

Contributed Paper

2:45

2pED6. Imperatives of acoustics and musical instrument acoustics education to the Nigerian educational system in tertiary institutions. Stephen G. Onwubiko, Adebowale O. .Adeogun (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com), Joy O. Nnabuchi (Dept. of Philosophy, Univ. of Abuja Nigeria, Gwagwalada, Abuja, Nigeria), Tobi E. Kemewerigha (College of Education Akamkpa Cross-River, Calabar State, Nigeria, Akakpa CrossRiver, Akakpa, Nigeria), and Ikechukwu E. Onwubiko (Dept. of Civil Eng., Federal Polytechnic Owerri, Owerri, Imo, Nigeria)

This paper discusses about the imperatives of acoustics and musical instrument acoustics education to the Nigerian educational system. It identifies the prospects, the long term educational effects, problems and proffered possible solutions to them. In achieving its objectives the study uses ethnographic and qualitative methods with simple percentages for eliciting and collation of data. The paper suggests that the Acoustics societies work with our curriculum planners, and the government to implant acoustics and its education to tertiary institution in Nigeria. The research questions raised guided the study. It proposes as part of its recommendations that other acoustical bodies provide necessary facilities and personnel for acoustics and its education to thrive as a core subject or vocational subject; and that the larger society must become educated on the usefulness of acoustics as a career subject worth pursuing by scholars.

3:00–3:30 Panel Discussion

Session 2pID

Interdisciplinary and Women in Acoustics: Panel Discussion: Mentoring Across Differences

Dominique A. Bouavichith, Cochair
Linguistics, University of Michigan, 611 Tappan St., #455C, Ann Arbor, MI 48109

Evelyn M. Hoglund, Cochair
Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43204

Kelly L. Whiteford, Cochair
Psychology, University of Minnesota, 75 East River Parkway, Minneapolis, MN 55455

Chair's Introduction—1:00

Panel Discussion: Mentoring Across Differences. Dominique A. Bouavichith (Linguist, Univ. of Michigan, 611 Tappan St, #455C, Ann Arbor, MI 48109, dbouavichith@gmail.com), Evelyn M. Hoglund (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), and Kelly L. Whiteford (Psych., Univ. of Minnesota, Minneapolis, MN)

A panel of educators in acoustics will discuss issues related to mentoring undergraduate and graduate students, with a focus on practices for creating successful mentoring relationships across differences. The panelists will briefly introduce themselves and describe their experiences as educators and mentors. The panelists will then answer questions from the audience.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 SALON C (VCC), 1:00 P.M. TO 5:00 P.M.

Session 2pNS

Noise, Structural Acoustics and Vibration, and Architectural Acoustics: Noise and Vibration from Fitness Activities

Matthew V. Golden, Cochair
Pliteq, 4211 Yonge St., North York, ON M2P 2A9, Canada

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Fitness centers in mixed-use development: Examples from practice. Benjamin Markham, Ethan Brush, and Robert Connick (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Spin classes utilize sound systems that generate levels in excess of 115 dBA, with subwoofers to match. Gyms located in office buildings feature Olympics style weight lifting technique that involves dropping 200 lb (or more) barbells from shoulder or overhead positions. Medicine balls get thrown at walls and floors. More "typical" sources of airborne and structureborne noise—treadmills, aerobics classes, weight machines, and the like—contribute to the din. The authors will present experiences and lessons learned from actual field experience with fitness centers located in commercial, residential, and mixed-use developments. Data will be presented on the effectiveness of mitigation methods installed in real buildings to attenuate airborne and structureborne noise. The session will include a

discussion of the various types of fitness franchises and associated acoustic concerns, illustrated by case study examples from the authors' acoustical consulting experience.

1:25

2pNS2. Further testing on the efficacy of inertia base weight training stations for CrossFit weight drops. Angela Waters (Kinetics Noise Control, 6300 Irelan Pl., Pataskala, OH 43017, awaters@kineticsnoise.com) and Scott W. Smith (Ballentine Walker Smith, Inc., Kennesaw, GA)

In an attempt to mitigate the adverse impact and vibration effects of CrossFit weight drops in a wood frame house in South Carolina, a concrete inertia base system was designed, constructed, and installed in 2013. The homeowners have been pleased with the system's performance, reporting major reductions in the felt vibration throughout the house. In 2017 initial testing was conducted that showed significant reduction in vibration in an adjacent corridor. This presentation reports on the results of subsequent and more detailed testing at remote locations in the house. New equipment became available which allowed the vibration levels on the inertia base and at the remote locations to be recorded simultaneously.

1:45

2pNS3. Specialty fitness centre noise issues—Case study. Brigette Martin (none, 1200 Lynn Valley Rd., Vancouver, BC v7j2a2, Canada, martin@bkl.ca)

This presentation will include a collection of case studies focusing on specialty fitness centres. There has been growth in the fitness industry for niche fitness facilities, many of which provide instructed classes in small spaces. These facilities can be located in mixed use buildings, where noise from fitness activities can result in complaints from other users in the building. This presentation provides a summary of some projects relating to these spaces, outline the noise issues, noise investigation, and solutions developed for the spaces. We have been invited to present at the Noise and Vibration from Fitness Activities session.

2:05

2pNS4. Long-term monitoring of fitness activity in a commercial gym. Samantha Rawlings, David W. Dong, and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com)

Noise and vibration from activity in fitness facilities is a common source of complaints in residential, commercial, and mixed use buildings. Most studies have focused on developing and measuring mitigation of the vibration from the activity. However, there is limited information on the characteristics of fitness activity and its effect on acceptability. It is assumed that the complaints will increase with the frequency of occurrence of the events, but there is little information on the temporal distribution of typical fitness activity. There is therefore a need for long term measurements of a large number of fitness facilities, comparable to noise data that have been collected around sources such as roadways and airports. Here, we begin to address this need and report results of long-term vibration monitoring of a commercial fitness facility.

2:25

2pNS5. Determining a relationship between mockup size and transmitted vibration. Paul Gartenburg (Eng., Pliteq Inc, 4211 Yonge St., Toronto, ON M2P2A9, Canada, pgartenburg@pliteq.com) and Matthew V. Golden (Eng., Pliteq Inc., Washington, District of Columbia)

Testing fitness flooring mockups is sometimes required to prove the efficacy of a fully installed floor. Constructing these *in-situ* mockups can be labour intensive and costly. Keeping mockup sizes low is beneficial from these standpoints but can be detrimental from an acoustical standpoint. The purpose of this paper is to understand the relationship between mockup size and transmitted vibration as a result of heavy weight impacts. To achieve this, mockups of varied sizes and buildups ranging from $2' \times 2'$ to $10' \times 0'$ were tested and resultant vibrations in the structural slab were recorded by two accelerometers adhered to the slab. This paper analyzes the recorded results for both locally reactive and resonantly reactive floors to provide insight in determining what mockup size is appropriate without significantly compromising the validity of the results.

2:45–3:00 Break

3:00

2pNS6. Investigation of vibration reduction methods from fitness activity. David W. Dong, John LoVerde, Richard Silva, and Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Noise and vibration from activity in fitness facilities, in particular drops of heavy weights, is a common source of disturbance and complaint in residential, commercial, and mixed-use building types. There is no standardized method for evaluating the reduction in noise or vibration provided by these products and systems. The authors previously reported (Internoise 2015, Noise-Con 2016, ASA Boston 2017) a preliminary test method to evaluate athletic tile flooring with heavy weight drops, based on the reduction in floor vibration (ΔL_V) achieved due to the insertion of the products. Preliminary results indicated that the ΔL_V measurement adequately described the impact reduction due to athletic tile, over a certain frequency range, reasonably independent of structure. This paper continues the research with measurements with additional materials and sources, and preliminary analytical investigations of the effects of mitigation methods.

3:20

2pNS7. Further prediction of heavy weight drops on resilient sports floors in existing buildings. Matthew V. Golden (Pliteq, 4211 Yonge St., North York, ON M2P 2A9, Canada, mgolden@pliteq.com) and Paul Gartenburg (Pliteq, Toronto, ON, Canada)

Currently, the assessment of noise and vibration due to heavy weight drops in an existing building can be cumbersome and time consuming, have low repeatability, and does not allow for evaluation of additional solutions not present at the time of testing. This paper will present further development of a method to solve those problems and is broken up into three parts. The first part is the laboratory assessment of the force pulse that a given weight will exert onto a resilient floor. The second part is a method to quickly obtain the in-situ transfer function between the force injected into the building and the resulting vibration elsewhere in the building. The third part combines the previous two items to predict the acceleration in a building due to any arbitrary combination of impact source and resilient floor. The results obtained will be compared to on-site weight drop measurements in order to qualify the predictive model. Extensions of the model into other structures, types of outputs (vibration vs sound pressure levels), and octave band vs narrow band results will be discussed.

3:40

2pNS8. Assessment and remediation of treadmill generated vibration increased due to a building modification. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

Structural modifications to an existing multi-story building installed to increase the load bearing capacity of a floor resulted in a noticeable increase in floor vibration from treadmills used in a fitness room on the floor above. Results from vibration measurements were combined with a finite element analysis (FEA) model of the building dynamics to determine the dynamic force generated by the treadmills in use. The FEA model and the calculated dynamic force were used to evaluate the effectiveness of options for vibration isolating the treadmills to reduce vibration to acceptable levels in the occupied office below.

Contributed Papers

4:00

2pNS9. Effects of noise exposure on hearing health evaluated through short interval otoacoustic emission monitoring: Preliminary results with low and moderate noise exposure groups. Vincent Nadon (École de technologie supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, vincent.nadon@etsmtl.ca) and Jeremie Voix (École de technologie supérieure, Montréal, QC, Canada)

Industrial workers are often exposed on a daily basis to noise doses that put them at risk of Noise-Induced Hearing Loss (NIHL), which remains the largest cause of indemnity in North America industries. To improve occupational hearing loss prevention programs, a method to continuously measure hearing fatigue using otoacoustic emissions (OAE) has already been proposed by the authors using a portable and robust OAE system designed for noisy industrial use. The effectiveness in the field of the aforementioned method is examined by comparing a control group of unexposed subjects in laboratory and noise exposed subjects in the field. All participants are equipped with an hearing protection earpiece which includes the usual OAE measurement components as well as a microphone mounted on the outer faceplate of the earpiece. At regular intervals over the course of the day, the growth function of distortion product OAEs is measured for both groups while simultaneously measuring noise levels inside and outside the ear. Medial olivocochlear reflexes are also measured pre and post exposure to monitor other potential effects on hearing. Preliminary results of this study are presented and analyzed in the broader context of the dose-response relationship that could indicate individuals' risk to develop NIHL.

4:15

2pNS10. Short-term annoyance due to construction work noise in a laboratory context. Jae Kwan Lee, Seo I. Chang (Energy Environment System Eng., Univ. of Seoul, 309, 2nd Eng. Bldg., 163, Seoulsiripdae-ro, Dongdaemun-gu, Seoul 02504, South Korea, jklee645@gmail.com), Jae Woong Jang, and Soo Il Lee (Mech. and Information Eng., Univ. of Seoul, Seoul, South Korea)

Among the various noises that cause environmental disputes, the noise of the construction site is particularly problematic. Psycho-acoustic experiment was used to develop an "annoyance model" due to construction site noise. In the experiment, four noises recorded at the construction site were used. There are two continuous noises and two impact noises. Some people were recruited for experiment and the classification was performed primarily. The subjects were classified into various criteria such as gender, age, sensitivity, disease, and the results were compared. The noise converted to 35–80dB (A) was randomly played. During the experiment, the annoyance was judged by the 11 point scale for every noise. The physical analysis of the sound used in the experiment was performed for the development of the annoyance model besides the auditory experiment. Annoyance model was developed by multiple regression analysis using Leq, Lmax, sound quality indices (loudness, sharpness, roughness, and tonality), and questionnaire score. Also, it was confirmed whether there was any difference in the annoyance model for each subject group.

4:30

2pNS11. The sound insulation performance of mass timber building elements. Jeffrey Mahn (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1C4N4, Canada, jeffrey.mahn@nrc-cnrc.gc.ca)

Mass timber constructions have been gaining popularity in Canada since provincial building codes were changed to allow for taller buildings of mass timber elements. Common mass timber elements include cross-laminated timber, nail-laminated timber, dowel-laminated timber, and glued laminated timber. Many of these elements have been in use for decades, but much still needs to be learned in terms of the transmission of structure-borne noise between elements and the transmission loss and radiation efficiency of the elements with and without linings. Recent and ongoing studies on cross-laminated and nail-laminated elements conducted at the National Research Council have investigated many of the unknowns and a summary of the results are presented in this paper.

4:45

2pNS12. Characterization, analysis, and noise control measures of a mechanical room. Deepak Deokant Jha, Joonhee Lee, and Mohammed Zaheeruddin (Bldg., Civil & Environ. Eng., Concordia Univ., E.V. 16.255, 1455 De Maisonneuve Blvd. W., Montreal, QC H3G 1M8, Canada, deepak.jha0509@gmail.com)

The acoustic environment of a mechanical room in buildings has received little attention because the space is occasionally occupied by building maintenance staff. However, the room typically includes the loudest noise sources in buildings. It is necessary to investigate noise exposure in the mechanical room for the staff. The aim of this study is, thus, to assess noise levels in the room by examining acoustic characteristics of the noise sources and mapping noise levels produced by the sources. This study also aims to provide optimal solutions to decrease the noise levels being produced. The framework consists of (1) identifying the major locations at which noise is being produced; (2) measurement of noise levels; and (3) performing noise level analysis of each activity. These findings will provide useful information to establish effective noise management plans for mechanical rooms in buildings.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 RATTENBURY A/B (FE), 1:00 P.M. TO 4:20 P.M.

Session 2pPA

Physical Acoustics, Architectural Acoustics, Noise, Signal Processing in Acoustics, and Underwater Acoustics: Challenges in Computational Acoustics

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290

Matthew G. Blevins, Cochair
U.S. Army Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61822

Chair's Introduction—1:00

Invited Papers

1:05

2pPA1. Numerical modeling of ultrasound propagation in heterogeneous media using a mixed domain method. Yun Jing and juanjuan Gu (North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

A mixed domain method (MDM) is presented in this paper for modeling one-way linear/nonlinear wave propagation in biological tissue with arbitrary heterogeneities, in which sound speed, density, attenuation coefficients, and nonlinear coefficients are all spatial varying functions. The MDM is based on solving a Westervelt-like equation. We validated the algorithm by studying a number of one-dimensional and two-dimensional problems. Overall, this study demonstrates that the MDM is a computationally efficient and accurate method when used to model wave propagation in biological tissue with relatively weak heterogeneities. We also proposed methods to improve the algorithm for moderately heterogeneous media.

1:25

2pPA2. High frequency propagation in a waveguide with an expanding section. Jerry H. Ginsberg (Dunwoody, GA) and David Feit (ASA, 1305 Walt Whitman Rd, Melville, NY 11747, dfeit@acousticalsociety.org)

A waveguide whose walls have a constant local impedance is readily analyzed if its cross section is uniform with a standard shape. This is not true if the cross section expands or contracts. The (one-dimensional) Webster horn equation typically is used for low frequency propagation in such configurations, but doing so restricts consideration to situations where the size of the cross section changes slowly over the scale of a wavelength. Somewhat paradoxically, a waveguide consisting of a sequence of different sized uniform segments has been the subject of numerous analyses that are not limited to low frequencies. The present work extends one such analysis, which uses a collocation technique at discontinuous junctions, to a two-dimensional configuration in which a transition section expands linearly. The field in each section is described as a modal series, whose terms are the product of a transverse standing mode and two phasors for axial propagation. Cartesian coordinates are used to describe the field within the uniform sections, whereas a polar coordinate description of the field in the expansion section facilitates satisfying rigid wall conditions. Various schemes for arranging the collocation points are discussed, and convergence of the series is examined.

1:45

2pPA3. Improving the speed of acoustic propagation modeling, for sonar training applications, with a 3D Gaussian ray bundling model. Sean M. Reilly (Physical Sci. and Systems (PS2), Raytheon BBN Technologies, 127 John Clark Rd., Middletown, RI 02842, sean.m.reilly@raytheon.com) and Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

This research improves the calculation speed of acoustic propagation modeling, for sonar training applications in littoral environments, by implementing a 3D Gaussian ray bundling model in the geodetic coordinates of the environmental databases. Tactical decision aids currently transform the three-dimensional environment into two dimensional radials, then compute acoustic propagation as a function of range and depth. This allows them to analyze millions of range, depth, and bearings for potential targets locations in a few minutes. But, it assumes that the cost of transformation is small compared to the speed benefit of computing propagation in Cartesian coordinates. This assumption is violated in applications where the geometry of the sensors and targets is rapidly evolving, results are strongly dependent on location, and answers must be produced faster than real time. Instead of relying on advancements in computer hardware, this research focuses on the development of a new algorithm to compute bistatic, broadband, transmission loss, and reverberation for 100 active sonar acoustic contacts on commodity laptop hardware, at rates ten times faster than the speed of sound, with no measurable impact on accuracy. [Work supported by the High Fidelity Active Sonar Training (HiFAST) Project at the U.S. Office of Naval Research.]

2:05

2pPA4. Optimizing acoustic diffusion of an architectural feature using finite-difference time-domain. Laura C. Brill and John T. Strong (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, lbrill@thresholdacoustics.com)

Parametric design is gaining in popularity as a design tool; however, rigorous analysis of the acoustic characteristics of parametrically designed surfaces typically requires testing physical mockups of a small sample of candidate designs. A computational method using a combination of finite-difference time-domain (FDTD) analysis and a genetic optimization algorithm has been developed to analyze a large number of candidate designs and select the best performers. In this case study, an architect's aesthetic vision was translated into geometric parameters used by a genetic algorithm to optimize acoustic diffusion performance. FDTD was used to simulate an impulse response and analyze the frequency response of reflective/diffusive geometry with a resolution of up to 11 kHz. The process utilized GPU computing clusters available through Amazon Web Services (AWS) to accelerate the analysis. The current availability and relative cost-effectiveness of cloud computing resources makes wave-based acoustic analysis a more viable and efficient option. The methods and resources discussed in this paper allow acoustic performance to be integrated into the architectural optimization process thereby making acoustic performance an active consideration in an iterative design approach.

2:25

2pPA5. Optimized geospatial tool for ambient soundscapes. Michael M. James, Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St, Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com), Mark K. Transtrum, Kent L. Gee, Katrina Pedersen, and Brooks A. Butler (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Ambient soundscape data are a critical component of numerous applications such as evaluating the environmental noise impacts of proposed activities and studying the physical and psychological impacts of a person's acoustic environment. The current approach to characterize ambient soundscapes requires intensive field measurement programs. A soundscape prediction tool capable of modeling the soundscape over large spatial regions and timeframes is needed to provide an alternative to intensive field measurements. To address this need, a machine learning based soundscape model trained with acoustic data and geospatial layers has been developed. The model's capability to generate A and flat-weighted exceedance levels across space, time (hourly, daytime, and nighttime), and frequency (one-third octave bands) will be demonstrated. [Work funded by an Army SBIR.]

2:45

2pPA6. Machine learning-based prediction of outdoor ambient sound levels: Ensemble averaging and feature reduction. Katrina Pedersen, Kent L. Gee, Mark K. Transtrum, Brooks A. Butler (Brigham Young Univ., N283 ESC, Provo, UT 84602, katrina.pedersen@gmail.com), Michael M. James, and Alexandria R. Salton (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Outdoor ambient sound levels can be predicted from machine learning-based models derived from geospatial and acoustic training data. To improve modeling robustness, median predicted sound levels have been calculated using tuned models from different supervised machine learning modeling classes. The ensemble-based model reduces errors at training sites for both overall levels and spectra, and produces more physically reasonable predictions elsewhere. Furthermore, the spread in the ensemble provides an estimate of the modeling accuracy. An initial analysis of feature importance metrics suggests that the number of geospatial inputs can be reduced from 120 to 15 without significant degradation of the model's predictive error, as measured by leave-one-out cross validation. However, the predictions from the reduced-feature modeling may be less physical in certain regions when all differentiating geospatial features are removed. These results suggest the need for more sophisticated data collection and validation methods.

3:05–3:20 Break

3:20

2pPA7. Optimizing calculation points using Gaussian process regression. Matthew Kamrath (CSTB, 72 Lyme Rd., Hanover, New Hampshire 03755, Matthew.J.Kamrath@erdc.dren.mil), Philippe Jean, Julien Maillard (CSTB, Saint-Martin-d'Hères, France), and Judicaël Picaut (IFSTTAR, Bouguenais Cedex, France)

Many applications in computational acoustics calculate a value (e.g., sound pressure level) at many different locations to approximate the value throughout a region using interpolation. Often, the points are uniformly or exponentially spaced without a rigorous procedure to optimize the locations because directly minimizing the interpolation error is too computationally expensive. Instead, the interpolation error can be indirectly reduced by minimizing the maximum variance estimated using Gaussian process regression. This approach is less expensive because the variance at a point is not a function of the value at that point. Thus, each evaluation of the objective function (i.e., the maximum variance) does not require additional acoustical computations, which mitigates the cost of the objective function. As an example, this procedure is applied to an outdoor sound propagation case to approximate the insertion loss of a 3 m tall T-shaped barrier compared to 3 m tall straight barrier, which are modeled using the boundary element method. In this case, the optimized locations have a smaller interpolation error than uniformly and exponentially distributed points.

3:35

2pPA8. A highly efficient approach to model acoustics with visco-thermal boundary losses. Martin Berggren, Daniel Noreland, and Anders Bernland (Dept. of Computing Sci., Umeå Univ., Campustorget 5, Umeå 90187, Sweden, martin.berggren@cs.umu.se)

For devices such as hearing aids, microphones, micro loudspeakers, and compression drivers, thermal and viscous boundary layer effects are often highly noticeable. These effects can be modeled in the linear regime by the linearized, compressible Navier-Stokes equations. However, the need for resolution of the very thin boundary layers typically makes numerical solutions of these equations computationally very expensive. Based on a boundary-layer analysis, we have derived for the pressure Helmholtz equation what appears to be a new boundary condition that accurately takes visco-thermal boundary losses into account. The model is valid when the wavelength and the minimum radius of curvature of the wall is much larger than the boundary layer thicknesses. In the special case of sound propagation in a cylindrical duct, the model collapses to the classical Kirchhoff solution. We assess the model in the case of sound propagation through a compression driver, a kind of transducer that is commonly used to feed horn loudspeakers. The transmitted power spectrum through the device calculated numerically using our model agrees extremely well with computations using a hybrid model, where the full linearized, compressible Navier-Stokes equations are solved in the narrow regions of the device and the pressure Helmholtz equations elsewhere. However, our model needs two orders of magnitude less memory and computational time than the more complete model.

3:50

2pPA9. Detection of bubble movements via wave phase conjugation. Amir Modarreszadeh, Evgeny Timofeev (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Montreal, QC H3A0C3, Canada, amir.modarreszadeh@mail.mcgill.ca), Alain Merlen, Olivier Bou Matar, and Philippe Pernod (Electronics, Microelectronics and NanoTechnol., Univ. of Lille 1, Villeneuve d'Ascq, France)

Wave Phase Conjugation (WPC) means reversing the propagation of waves so that initial spatial distributions of amplitudes and phases are conserved. One of the methods to conjugate ultrasonic waves is to modulate the speed of sound in a solid conjugator by an alternating electromagnetic field. The detection of bubble motions in a fluid to be considered in this work is among potentially interesting and omnipresent WPC applications in industry. As the ultrasonic waves have short wavelengths, high-order in space and time numerical methods are required for modeling. In this work, a modified version of the Nodal Discontinuous Galerkin method, which is based on the non-collocated solution and flux bases, is implemented for wave propagation in solids and liquids (for both linear and non-linear flow regimes) in an axisymmetric geometry. Being assured of the accuracy and performance of the numerical technique by evaluating some representative test cases, the detection of bubble motion and growth in a flow field using WPC is simulated with the inclusion of all elements of the WPC process: the transducer, the conjugator, and the bubbly liquid itself. The developed methodology and results can be used to design and improve measurement devices based on the WPC phenomenon.

4:05

2pPA10. Target-oriented waveform inversion. Tianci Cui, James E. Rickett (Geophys., Schlumberger Cambridge Res., High Cross, Madingley Rd., Cambridge CB30EL, United Kingdom, tcui2@slb.com), and Ivan Vasconcelos (Earth Sci., Utrecht Univ., Utrecht, Netherlands)

Full-waveform inversion (FWI) has demonstrated increasing success in estimating high-resolution medium parameters, which are obtained by minimizing the difference between synthetic and real data on the acquisition surface. In the standard approach, the whole propagation domain needs to be resolved. This highly nonlinear problem is usually solved via a gradient-based local optimization approach, which can be computationally very expensive. We propose target-oriented FWI utilizing novel redatuming techniques. From surface measurements, Marchenko redatuming retrieves full-waveform up- and downgoing wavefields inside the medium at no significant computational cost. We choose two datum locations inside the medium, one above and one below the target volume. Marchenko-redatumed wavefields at both target-enclosing boundaries consist of in- and outgoing wavefields. The outgoing wavefields are the response of the ingoing wavefields being scattered from the target volume only, without interference from the overburden or underburden. We conduct FWI by finding a target model whose response to the ingoing wavefields fit the outgoing wavefields best. The computational cost of targeted FWI, which involves solving the wave equation in the target volume only, will lead to savings over the traditional full-volume approach. We validate our method numerically on a 2D acoustic model.

Session 2pPP

Psychological and Physiological Acoustics and Speech Communication: Speech Perception in Children with Hearing Impairment

Kelly N. Jahn, Cochair
Speech and Hearing Sciences, University of Washington, 1417 NE 42nd Street, Box 354875, Seattle, WA 98105-6246

Mishaela DiNino, Cochair
Speech and Hearing Sciences, University of Washington, 1417 NE 42nd St., Seattle, WA 98105

Chair's Introduction—1:00

Invited Papers

1:05

2pPP1. Optimizing speech recognition in realistic environments for children who wear hearing aids. Ryan W. McCreery (Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ryan.mccreery@boystown.org), Elizabeth Walker (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), and Marc Brennan (Special Education and Commun. Disord., Univ. of Nebraska - Lincoln, Lincoln, NE)

Children with hearing loss are immersed in listening situations with poor signal-to-noise ratios and high levels of reverberation. Children's ability to effectively communicate in acoustically challenging listening situations has an impact on language development and academic and social functioning. Children with hearing loss who use hearing aids perform more poorly than peers with normal hearing when listening in noise and reverberation. However, speech understanding among children with hearing loss is heterogeneous and most previous research has focused primarily on the effect of a child's degree of hearing loss derived from the audiogram. Our research explores additional factors that contribute to individual differences in speech recognition in adverse acoustic conditions for children with hearing loss. This work has focused on three factors that explain individual differences in speech recognition in noise among children who use hearing aids: 1) the degree to which speech audibility is restored by the hearing aid(s), 2) the child's language abilities, and 3) the child's cognitive abilities, including working memory and selective attention. The relationships between speech recognition and these factors, as well as the clinical implications of these findings, will be described.

1:35

2pPP2. A functionally relevant measure of spatial release from masking in children with bilateral cochlear implants. Z. Ellen Peng and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, z.ellen.peng@wisc.edu)

The ability to segregate speech from noise is an important skill for children as they navigate everyday noisy environments. Spatial release from masking (SRM) is a measure of benefit observed when target speech is co-located versus spatially separated from maskers. SRM is typically studied with target-maskers in front versus maskers displaced to fixed angles towards the side. Children fitted with bilateral cochlear implants (BiCIs) typically show smaller SRM than children with normal hearing (NH), even for target-masker angular separation of 90°, and when monaural head shadow cues exist. Here, we used an open-set corpus that has high informational masking and better mimics realistic listening situations for children. First, SRM was measured by adapting target-masker separation, and quantified as the minimal angular separation needed to achieve 20% improvement in target intelligibility. Second, SRM was measured for target-masker separation of 180°. Preliminary results suggested that, in the BiCI group, all children achieved 20% SRM with a target-masker angular separation smaller than 180°. With the full 180° target-masker separation, children in the BiCI group achieved a much larger SRM (6–8 dB) than previously reported while NH children's SRM was larger. Acoustic cues available through CI processors will be discussed. [Work funded by NIH-NIDCD.]

Contributed Paper

2:05

2pPP3. Vowel confusion patterns in children and adults with cochlear implants. Kelly N. Jahn, Mishaela DiNino, Matthew Winn, Jacob Hulswit, and Julie G. Arenberg (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Box 354875, Seattle, WA 98105-6246, jahnk@uw.edu)

Suboptimal interfaces between cochlear implant (CI) electrodes and their target neurons may distort the spectral information that is important for vowel identification. The goal of this investigation was to relate vowel confusion patterns of CI users to their site-specific (i.e., frequency-specific) variation in electrode-neuron interface across the electrode array. The electrode-neuron interface is estimated using single-channel detection thresholds with a spatially focused electrode configuration; elevated thresholds indicate channels with relatively poorer frequency transmission. We hypothesized that higher focused thresholds on channels transmitting low frequency (<983 Hz) information would be associated with errors in vowel height, whereas higher thresholds on channels transmitting mid- and high-frequency (>984 Hz) information would be related to vowel advancement errors. Vowel recognition was assessed with a closed set of medial vowels in 4-talker babble noise. Focused thresholds were measured on electrodes 2–15. Vowel confusions were examined according to the direction of errors in 2-dimensional physical articulatory and acoustic F1-F2 space. Results showed that higher average focused thresholds on the 3 most basal electrodes tested (2327–4630 Hz) were associated with perception of front vowels as more back. These findings suggest that suboptimal electrode-neuron interfaces could systematically affect CI listeners' perception of vowel characteristics.

Invited Papers

2:20

2pPP4. Remote microphone system use in the homes of children with hearing loss. Carlos R. Benitez-Barrera, Gina Angley, and Anne Marie Tharpe (Hearing and Speech Sci., Vanderbilt Univ., 2031 Convent Pl, 2031, Nashville, TN 37212, carlos.r.benitez@vanderbilt.edu)

Remote microphone systems (RMS) are known to improve speech recognition skills of children with hearing loss in settings where noise and distance are present (e.g., the classroom, the home; Bertachini, 2016). However, although RMSs are widely recommended in the classroom setting, very few and inconclusive studies have examined the possible language benefits related to the use of RMS use in the home environment. In this presentation, we will discuss a series of projects in which we investigated the effects of home-use of an RMS on the communication of 10 families with young children with hearing loss. Language Environmental Analysis (LENA^{TM}) recorders were used during two consecutive weekends (one weekend with the remote microphone and one without). Caregiver talk, child-directed speech, and other caregiver communication strategies were compared across both weekends. Caregiver perceptions related to RMS use will also be discussed.

2:50–3:10 Break

3:10

2pPP5. Spectral modulation detection in adolescents with normal hearing or cochlear implants predicts some language skills, but not others. Susan Nittrouer, Joanna H. Lowenstein, and Donal Sinex (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2147, Gainesville, FL 32610, snittrouer@phhp.ufl.edu)

Previous outcomes of an ongoing longitudinal study involving children with cochlear implants (CIs) have revealed that these children demonstrate especially large deficits in two areas: (1) language skills dependent upon phonological structure; and (2) speech-in-noise recognition. Morphosyntactic skills appear closer to typical. We have hypothesized that both the phonological deficit and the speech-in-noise problems arise from poor spectral resolution. To test this hypothesis we measured children's spectral modulation detection (SMD). Participants were14-year-old children with normal hearing (49) or with CIs (42). A 3-interval, forced-choice procedure was used in which two stimuli were unmodulated noises and one was a noise modulated at a rate of 0.5 ripples per octave (rpo). The 70.7% threshold for SMD was estimated adaptively. Other measures included tests of phonological awareness and processing, vocabulary and syntactic abilities, and speech recognition in single- or multi-talker babble, as well as in speech-shaped noise. Children with CIs had slightly higher SMD thresholds, and poorer performance on all other measures. For both groups, SMD thresholds were related to phonological skills and speech-in-noise recognition, but not to vocabulary or syntactic abilities. Outcomes were interpreted as suggesting that phonological sensitivity and speech-in-noise recognition depend upon spectral resolution, but morphosyntactic skills do not.

Contributed Paper

3:40

2pPP6. Perception of Mandarin Chinese vowels in young hearing-impaired and normal-hearing children. Changxin Zhang (Dept. of Education and Rehabilitation, East China Normal Univ., Shanghai, Shanghai, China) and Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu)

The study investigated the perception of Mandarin Chinese vowels of 45 congenital hearing-impaired and 45 normal-hearing children aged 4 to 6. There were 30 children at each age group, half of which were hearing-impaired children. All of the hearing-impaired children received hearing aids or cochlear implants before the age of 5. In a picture identification task, listeners identified the target consonant-vowel word among 2–4 contrastive words. The target words and the other contrastive words differed only in consonants. Each target word represented a concrete object and was spoken by a young female native-Chinese talker. Sixteen of the target words ended with monophthong, twenty-two with diphthong and nine with triphthong. Age showed a significant effect on vowel perception for both groups. Normal-hearing children showed significantly better identification of all three types of vowels than hearing-impaired children at the age of 6, whereas the two groups had comparable performance at age of 4 and 5. For hearing-impaired children, a rapid development of diphthong perception occurred between 4 and 5 years old, while a rapid development of monophthong perception between 5 and 6. The effect of hearing loss severity and the use of hearing aids or cochlear implants will be discussed.

Invited Paper

3:55

2pPP7. Your ears never sleep: Auditory processing of nonwords during sleep in children. Adrienne Roman, Carlos Benitez, Sasha Key, and Anne Marie Tharpe (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 904 Erin Ln., Nashville, TN 37221, adrienne.s.roman@vanderbilt.edu)

The auditory system in humans is the only sensory system that remains active for the continuous scanning and processing of information during sleep; however, little research has investigated this phenomenon. This study examined whether a brief exposure to auditory stimuli during sleep could result in memory traces indicated by event-related potentials (ERPs) in children. Twelve preschool children with normal hearing (2–5 years) were presented with three randomly selected nonwords (250 trials; 45–50 dB SPL) for 10 min during a regular nap (sleep state verified by EEG). After the nap, children's memory for the stimuli was evaluated using a passive listening version of the "old/new" ERP paradigm that included three nonwords from the nap exposure (repeated condition, 45 trials), 3 new nonwords (sham repeated condition, 45 trials), and 45 other distinct nonwords (novel condition). Memory for the "nap exposure" words was reflected by increased positive amplitudes at midline parietal locations between 350 and 600 ms compared to the sham and novel conditions. This evidence of the memory trace for the auditory stimuli experienced during sleep can have implications for children with hearing loss who routinely remove their assistive devices during sleep. Preliminary results from children with cochlear implants will be discussed.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 UPPER PAVILION (VCC), 1:00 P.M. TO 2:30 P.M.

Session 2pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration (Poster Session)

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

All posters will be on display from and all author will be at their posters from 1:00 p.m. to 2:30 p.m.

Contributed Papers

2pSA1. Remote focusing of elastic waves in complex structures using time reversal of acoustic waves. Maxime Farin, Claire Prada, and Julien de Rosny (Acoust., Institut Langevin, Institut Langevin, 1, rue, Jussieu, Paris, France 75005, France, maxime.farin@espci.fr)

Remotely assessing the state of damage of thin plates or beams within a complex structure is a burning issue in many industrial problems. A time-reversal technique is used to focus a short and localized elastic impulsion on a thin plate using a network of loudspeakers. The generated plate vibration is measured using a laser vibrometer. We built an experimental setup to mimic a complex structure, with several thin plates attached close to each others inside a cylinder. We show that the acoustic method allows us to put a targeted plate in vibration inside the cylinder, without exciting the other plates in the same structure. The time-reversal technique takes advantage of the strong wave reverberation caused by the presence of the complex structure around the plate, which creates virtual sound sources, and improves the focusing on the plate compared to that obtained when the plate has nothing around it. The ratio of plate vibration energy over emitted acoustic energy is about 1%. Finally, we create a small defect on a plate and assess whether the defaults can be detected from changes in the plate vibration modes. Our experimental results are compared with finite elements simulations.

2pSA2. Road pavement defect assessment through vibration analysis inside vehicles. Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl) and Maciej Szczodrak (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland)

Miniature accelerometer sensors were used for the evaluation of road surface roughness. The device designed for installation in the vehicles is composed of a GPS receiver and of multi-axis accelerometers. Smartphones with built-in accelerometers were also used. Measurement data were collected through the recording of road trips employing 3 car types on diversified surface roughness roads and with varied vehicle speed on each investigated road section. The first step of data processing was the sensor alignment made to achieve proper values of acceleration vector. Subsequently, the influence of the car suspension system to the measurement results was diminished employing a designed filter. The magnitude of coefficients of Gabor transform analysis of accelerometer signals was calculated to discover differences between good and damaged surface. Research results show that road sections quality can be assessed by the applied vibration analysis. Although the precision of low-cost devices may be lower than the application of expensive professional laser profilograph scanning on road pavements, they can help to increase the effectiveness and the coverage of road surface damage detection, through the monitoring of road surfaces with typical cars instead of special test vehicles only. [Research was subsidized by the Polish National Centre for Research and Development and the General Directorate of Public Roads and Motorways within the Grant No. OT4- 4B/AGH-PG-WSTKT.]

2pSA3. Dynamic response sensitivity and variability of a buckling-restrained braced frame under earthquake ground motion. Max D. Magalhaes (Structural Eng., Federal Univ. of Minas Gerais, R. Sao Sebastiao do Paraiso, 305/101 - Itapoa, Belo Horizonte 31710080, Brazil, maxdcm@gmail.com) and Tony Yang (Civil Eng., Univ. of Br. Columbia, Vancouver, BC, Canada)

The main goal of this paper is to examine the variability of deformation, stress and energy absorption of a BRBF (Buckling-Restrained Brace Frame) to some dynamic parameters via a parametric study. This study is aimed at providing not only a better understanding of the vibration transmission mechanism in itself but also to produce a useful set of data which for instance can be used by earthquake engineers as input data for a hybrid analysis. This data might be useful for optimizing vibration insulation in buildings at low frequencies, where the modal behaviour of buildings strongly influences the transmission. The results that are discussed in this paper were obtained via simulations using a modal model. The analysis is based on considering the influence of some variations in the "input" parameters, which are required in the pre-processing stage of a numerical experiment, and on the subsequent vibration transmission mechanisms of typical building configurations.

2pSA4. Eulerian motion magnification applied to structural health monitoring of wind turbines. Sebastian Cygert (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland) and Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl)

Several types of defects may occur in wind turbines, as physical damage of blades or gearbox malfunction. A wind farm monitoring and damage prediction system is built to observe abnormal vibrations of elements of wind turbine: blades, nacelle, and tower. Contactless methods are developed which do not require turbine stopping. In this work, structural health monitoring of a wind turbine is evaluated using a conversion from the captured and processed video to the acoustic signal, employing the method of Eulerian motion magnification in video. It was assumed that this task can be achieved using a stabilized high-speed video camera only, directed at the wind turbine without any additional sensors mounted on windmill blades or on its body. Moreover, the developed vector sound intensity probe was used for spatial measurements in order to recover the vibration modes of wind turbines. Finally, statistical methods were applied to the processing of computed features reflecting vibrations in order to determine wind turbine technical condition. The developed method was evaluated empirically in real wind farm. [Research was subsidized by the Polish National Centre for Research and Development within the grant "STEO—System for Technical and Economic Optimization of Distributed Renewable Energy Sources," No. POIR.01.02.00-00-0357/16.]

2pSA5. Effects of L2 proficiency on the production of voiceless stops. Ji-Hyun Jeon (Yonsei Univ., Seoul KS013, South Korea, wjswlgus6623@hanmail.net) and Seok-Chae Lee (Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

This study investigates how native speakers and L2 learners of English (L2 learners being native speakers of Korean) produce voiceless stops in English in the following phonological contexts: word-initial vs. -medial, stressed vs. unstressed, and when preceded by /s/. The study also examines the correlation between proficiency of L2 learners of English and the degree of aspiration (represented by VOT) in English voiceless stops. The speech of Korean L2 learners of English rated and categorized into 5 English proficiency levels in Genie Speech Corpus will be used. We will measure VOT of voiceless stops in the aforementioned conditions produced by 5 English speakers and each 5 Korean speakers per level. We expect that highly rated L2 speakers of English will produce longer VOT in stressed syllables than in unstressed syllables, as native speakers of English do. However, as the proficiency level gets lower, L2 speakers will produce it to a lesser degree than native speakers do, presenting a wide variety of VOT length.

2pSA6. Experimental research on fatigue performance of aero-structure under two combined loads. Zhihong Liu (Northwestern PolyTech. Univ., Youyi xilu 127#, Xi'an 710072, China, liuzh@nwpu.edu.cn), Li zhang (Aircraft Strength Res. Inst., Xi'an, China), Xing Liu (Northwestern PolyTech. Univ., Xi'an, China), Dingwen Guo, and Kai Pan (Aircraft Strength Res. Inst., Xi'an, China)

In this paper, a new engineering test method which put forward the initial stage of aircraft design, bearing both acoustic and static loads, is carried out in Lab. From currently test effect, it could accurately simulate the real situation of aircraft panel under hydrostatic load and acoustic load, and the test result shows that the structure presents plastic deformation under the combined action of acoustic load and static load, which differs from the common failure mode of the structure under the action of the single strong noise load. Furthermore, the proposed method could provide guidance to support aircraft type selection.

Session 2pSC

Speech Communication: Speech Perception (Poster Session)

Rajka Smiljanic, Chair
Linguistics, University of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198

All posters will be on display from 2:00 p.m. to 5:00 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 2:00 p.m. to 3:30 p.m. and authors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

Contributed Papers

2pSC1. Perception of vowels with missing formant information. Filip Nenadic (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada), Pamela Coulter, Michael Kiefte (School of Commun. Sci. and Disord., Dalhousie Univeristy, 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca), and Terrance M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

The dominant formant-based model of vowel perception has been challenged by several whole-spectrum approaches. Recent arguments in favor of the importance of cues other than formant frequencies come from a study by Ito *et al*. [J. Acoust. Soc. Am., 110, 1141–1149] in which suppressing either of the first two formants did not radically change the identification of Japanese vowels. The present study replicates the experiment using the larger vowel system of the English language. Visual inspection shows that even when a formant is suppressed, listener responses do not deviate as much as would be expected if formants were the sole cue for vowel identification. However, quantitative analyses indicate that participant agreement in which vowel they heard is significantly lower when they respond to stimuli with suppressed formants. Additionally, the suppressed formant value becomes a less important predictor of vowel identity. These changes in responses become even larger when stimuli (original, F1-suppressed, and F2-suppressed vowels) are presented together rather than in separate blocks. Taken together, these results show that, although formants are not the only correlate to vowel identity, they seem to be the most important.

2pSC2. Short-term, not long-term, average spectra of preceding sentences bias consonant categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Phoneme perception is influenced by spectral properties of surrounding sounds. For example, listeners perceive /g/ (lower F3 onset) more often after sentences filtered to emphasize high-F3 frequencies, and perceive /d/ (higher F3 onset) more often after sentences filtered to emphasize low-F3 frequencies. These biases are known as spectral contrast effects (SCEs). Much of this work examined differences between long-term average spectra (LTAS) of preceding sounds and target phonemes. Stilp and Assgari (2018 ASA) revealed that spectra of the last 500 ms of precursor sentences, not the entire LTAS, predicted biases in consonant categorization. Here, the influences of Early (before the last 500 ms) versus Late (last 500 ms) portions of precursor sentences on subsequent consonant categorization were compared. Sentences emphasized different frequency regions in each temporal window (Early = low-F3 emphasis, Late = high-F3 emphasis, and vice versa) naturally or via filtering. Entire-sentence LTASes predicted that SCEs would not bias consonant categorization, but instead responses were biased by the spectrum of the Late window. This was replicated when the Early window did not emphasize either frequency region but the Late window did. Results endorse closer consideration of patterns of spectral energy over time in preceding sounds, not just their LTAS.

2pSC3. CRSS-LDNN: Long-duration naturalistic noise corpus containing multi-layer noise recordings for robust speech processing. John H. L. Hansen, Harishchandra Dubey, and Abhijeet Sangwan (The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

Multi-layer noise refers to scenarios where multiple distinct noise sources are simultaneously active in an audio stream. We collected a corpus named the CRSS long-duration naturalistic noise (CRSS-LDNN) corpus. It contains noise captured from complex daily-life activities using wearable LENA units. The diversity in noise-sources include construction noise, noise like multi-speaker babble, large-crowd noise, vehicle/bus noise on the road, home environment noise, etc. Corpus contains recordings of complex mixtures of these noise types. The babble noise and bus-engine noise present along with occasional impulsive-noise over long-duration is an example of such scenario. The data were collected at 16 kHz sampling rate with 16-bit precision in .wav format. This data would be released to speech community (http://crss.utdallas.edu). During the summer semester, a CRSS student wore a LENA device that was switched ON when multiple noise-sources were present. This corpus was recorded in naturalistic scenarios with uncontrolled mixing of various noise-sources. It provides naturalistic multi-layer noise recordings for evaluation of robust speech algorithms such as speech recognition, speaker diarization and verification, sentiment analysis. It consists of approximately 19 hours noise recordings. The CRSS-LDNN noise is more challenging as compared to existing noise corpora such as NOISEX that contains only single noise.

2pSC4. The role of F2 and F3 in the perception of liquids for Hindi and Mandarin listeners. Phil Howson (Univ. of Oregon, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, phil.howson@mail.utoronto.ca), Jessamyn L. Schertz (Univ. of Toronto, Mississauga, Mississauga, ON, Canada), and Irfana M. (Netaji Subhash Chandra Bose Medical College, Jabalpur, India)

Rhotics and laterals are often described as being separated by F3; however, recent work has described lower F2 as a distinguishing quality of rhotics. The current work examines the perception of liquids (rhotics and laterals) by Mandarin and Hindi listeners. Participants completed a forced-choice identification task on a set of 30 stimuli, manipulated from a natural Malayalam production /aɻa/ to vary systematically in F2 and F3. Listeners chose the consonant they heard from a set of approximants (/r/, /l/, /j/, and /w/ or /ʋ/). The results revealed that liquid identification was mostly confined to the mid F2 range, from 1228 Hz to 1728 Hz. Within that range, stimuli with the lowest F2 and F3 values were predominantly identified as rhotics, while those with the highest F2 and F3 values were identified as laterals. The F3 range of 1885 Hz–2135 Hz was ambiguous between rhotics and laterals. The data suggest that F2, as well as F3, plays a key role in rhotic identification, and suggest overlap in the F3 perceptual space for rhotics and

laterals for Mandarin and Hindi listeners. The data suggest a complex relationship between F2 and F3 and liquid perception.

2pSC5. Aging effects on categorical perception of Mandarin tones in noise. Can Xu (Commun. Sci. and Disord., The Univ. of Texas at Austin, School of Foreign Lang., Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Minhang District, Shanghai, Shanghai 200240, China, 8023xc@sjtu.edu.cn), Yuxia Wang, Xiaohu Yang (School of Foreign Lang., Shanghai Jiao Tong Univ., Shanghai, China), and Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

The current study aimed at investigating the aging effect on the categorical perception of Mandarin lexical tones with varied fundamental frequency (F0) contours in noise. Mandarin tone identification and tone discrimination in quiet and noise were measured for younger and older listeners in a categorical perception paradigm where the stimuli continua varied their F0 contour systematically from the level to the rising/falling tones. Results reported that older listeners, in contrast with their younger counterparts, performed with less stimulus-tuned changes in both identification and discrimination functions and smaller peakedness in the discrimination function for both level-rising and level-falling tones. The aging effects on Mandarin tone categoricality were observed in both quiet and noise. Moreover, noise aggravated the aging effects, especially with the high SNR condition. Plus, older listeners' identification and discrimination functions in CP paradigm positively correlated with their performance in general speech identification in noise; such correlation was not found with younger listeners. Our study suggested that older listeners had less categoricality in both identification and discrimination functions for the level-rising and level-falling tones, probably due to the aging-related decline in temporal processing.

2pSC6. Effects of receptive language ability on the neural representation of phonetic category structure. Julia R. Drouin and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Storrs, CT 06269, julia.drouin@uconn.edu)

Speech sound categories have a graded internal structure that reflects typicality of the speech input. Neuroimaging findings reveal dissociable regions for resolving category membership and processing phonetic category structure; frontal regions show increased activation for ambiguous compared to unambiguous exemplars, and temporoparietal regions show increased activation for poor compared to prototypical exemplars, even when category membership is unambiguous. Individuals with developmental language disorder (DLD) show behavioral deficits in some categorization tasks, which may reflect difficulty in organizing auditory information into structured categories. An fMRI paradigm is used to examine whether the neural representation of phonetic category structure reflects receptive language ability. Participants were assigned to either the DLD or control group based on performance for standardized assessments of receptive and expressive language. To assess neural representation of phonetic category structure, participants completed a phonetic categorization task for tokens of *bowl* and *pole* using an event-related, sparse-sampling design; voice-onset-times (VOTs) of the *pole* tokens were manipulated to have values representing an ambiguous exemplar, a prototypical exemplar, and an extreme (i.e., long VOT) exemplar. The analyses will examine whether individuals with DLD show attenuated neural representation of phonetic category structure, and, if so, whether attenuation reflects individual differences in receptive language ability.

2pSC7. Structured phonetic variation facilitates talker identification. Divya Ganugapati and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

Individual talkers systematically differ in their phonetic implementation of speech sounds, and listeners use this structure to facilitate language comprehension. Here we test the hypothesis that listeners' sensitivity to talker-specific phonetic variation also facilitates voice processing. Listeners completed training and test phases. During training, listeners learned to associate talkers' voices with cartoon avatars (with feedback). Training stimuli consisted of single-word utterances from two minimal pairs (*gain*, *cane*, *goal*, and *coal*). In one condition, voice-onset-times (VOTs) of the voiceless-initial words were structured such that each talker had a characteristic VOT. In the other condition, listeners heard the same VOTs, but no talker-specific phonetic structure was present. At test, listeners heard all VOT variants for trained and novel words, and they performed the same voice-avatar identification task (without feedback). The results showed that given limited exposure (one training block), learning of the talkers' voices was equivalent between the two conditions. However, given extended exposure (three training blocks), talker identification was improved with respect to both accuracy and processing time for listeners who received structured phonetic variation compared to those who did not. These findings suggest that given sufficient time to learn talkers' idiolects, listeners use structured phonetic variation to facilitate voice processing.

2pSC8. He said, she said: Talker-specific influences on memory for spoken language. Elizabeth O'Brien and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, elizabeth.obrien@uconn.edu)

Memory for spoken language is not a veridical representation of experience. Instead, memory reflects integration across our interlocutors' messages, resulting in robust memory for meaning with relatively poor memory for specific form. Here we test the hypothesis that talker identity influences how spoken language is integrated in memory. Listeners completed encoding and recognition phases. During encoding, listeners heard sentences that contained constituents of four-part semantic units (i.e., "idea sets"). During recognition, listeners heard novel sentences that contained constituents of each idea set in addition to full idea sets and four-part sentences formed by combining constituents across idea sets (i.e., "noncases"). Across experiments, talker was manipulated in terms of (1) the match between encoding and recognition and (2) the number of talkers during encoding. The results to date replicate evidence of integration during encoding; when talker is held constant, listeners show more false memories at recognition for idea sets compared to constituents, and no false memories for noncases. However, false memories decrease when the talker differs between encoding and recognition, and when listeners hear multiple talkers compared to a single talker during encoding. These findings suggest that talker identity provides a critical structure for the integration of spoken language in memory.

2pSC9. The stability of visual–aerotactile effects across multiple presentations of a single token. Sharon Kwan, Megan Keough, Ryan C. Taylor, Terrina Chan, Murray Schellenberg, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, sksharonkwan@gmail.com)

Previous research has shown that the sensation of airflow causes bilabial stop closures to be perceived as aspirated even when paired with silent articulations rather than an acoustic signal [Bicevskis *et al.* 2016, JASA 140(5): 3531–3539]. However, some evidence suggests that perceivers integrate this cue differently if the silent articulations come from an animated face [Keough *et al.* 2017, Canadian Acoustics 45(3):176–177] rather than a human one. Participants shifted from a strong initial /ba/ bias to a strong /pa/ bias by the second half of the experiment, suggesting the participants learned to associate the video with the aspirated articulation through experience with the airflow. One explanation for the above findings is methodological: participants saw a single video clip while previous work exposed participants to multiple videos. The current study reports two experiments using a single clip with a human face (originally from Bicevskis *et al.* 2016). We found no evidence of a bias shift, indicating that the findings reported by Keough *et al.* are not attributable to the use of a single video. Instead, our findings suggest that aero-tactile cues shift consonant perception regardless of the number of recordings presented as long as the speaking face is human.

2pSC10. Perceiving prosodic prominence via unnatural visual information in avatar communication. Ryan C. Taylor, Dimitri Prica, Megan Keough (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, taylryan@gmail.com), and Bryan Gick (Haskins Labs., Vancouver, Br. Columbia, Canada)

Listeners integrate information from simulated faces in multimodal perception [Cohen, & Massaro 1990, Behav. Res. Meth. Instr. Comp. 22(2), 260–263], but not always in the same way as real faces [Keough *et al.* 2017, Can. Acoust. 45(3):176–177]. This is increasingly relevant with the dramatic increase in avatar communication in virtual spaces [https://www.bloomberg.com/professional/blog/computings-next-big-thing-virtual-world-may-reality-2020/]. Prosody is especially relevant, because compared to segmental speech sounds, the visual factors indicating prosodic prominence (e.g., eyebrow raises and hand gestures) frequently bear no biomechanical relation to the production of acoustic features of prominence, but are nonetheless highly reliable [Krahmer & Swerts 2007, JML 57(3): 396–414], and avatar virtual communication systems may convey prosodic information through unnatural means, e.g., by expressing amplitude via oral aperture (louder sound = larger opening); the present study examines whether this unnatural but reliable indicator of speech amplitude is integrated in prominence perception. We report an experiment describing whether and how perceivers take into account this reliable but unnatural visual information in the detection of prosodic prominence. Preliminary evidence suggests that oral aperture increases prominence with differences by sentence position.

2pSC11. Perceiving audiovisual speech articulation in virtual reality. Megan Keough, Ryan C. Taylor, Dimitri Prica, Esther Y. Wong, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mkeough@alumni.ubc.ca)

Listeners incorporate visual speech information produced by computer-simulated faces when the articulations are precise and pre-programmed [e.g., Cohen, & Massaro 1990, Behav. Res. Meth. Instr. Comp. 22(2), 260–263]. Advances in virtual reality (VR) and avatar technologies have created new platforms for face-to-face communication in which visual speech information is presented through avatars. The avatars' articulatory movements may be generated in real time based on an algorithmic response to acoustic parameters. While the communicative experience in VR has become increasingly realistic, the visual speech articulations remain intentionally imperfect and focused on synchrony to avoid uncanny valley effects [https://developers.facebook.com/videos/f8-2017/the-making-of-facebook-spaces/]. Depending on the VR platform, vowel rounding may be represented reasonably faithfully while mouth opening size may convey gross variation in amplitude. It is unknown whether and how perceivers make use of such underspecified and at times misleading visual cues to speech. The current study investigates whether reliable segmental information can be extracted from visual speech algorithmically generated through a popular VR platform. We report on an experiment using a speech in noise task with audiovisual stimuli in two conditions (with articulatory movement and without) to see whether the visual information improves or degrades identification.

2pSC12. Single-channel vibrotactile feedback for voicing enhancement in trained and untrained perceivers. David G. Marino (Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), Hannah Elbaggari, Tzu Hsu Chu (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada, h.elbaggari@alumn.ubc.ca), Karon MacLean (Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Auditory speech intelligibility can be enhanced by integrating information from other modalities, e.g., vision [Sumby & Pollack 1954, *J. Acoust. Soc. Am.* 26: 212] or direct manual touch [Gick *et al.* 2008. *J. Acoust. Soc. Am.* 123: EL72]. There are, nonetheless, many circumstances where shared visual attention may be hard to establish, or where in-person contact may be infeasible (e.g., in a noisy collaborative environment). To test the feasibility of using vibrotactile feedback to enhance intelligibility under noisy conditions, we use a portable voice-coil-based transducer that provides vibrotactile stimulation similar to laryngeal vibrations. Participants were asked to discriminate between minimal pairs in noise. These were distinguished in voicing and vowel height. Participants were asked to wear a vibrator on their fingers, or on their suprasternal notch. We contrasted vibrator placement with different vibration styles, such as a constant vibration on voicing, or vibrations driven by the amplitude envelope of the speech signal. In untrained perceivers we found that vibrotactile feedback increased accuracy regardless of placement. This effect, though significant, was not strong enough to be useful for everyday speech enhancement. These results, and those of a follow-up study with trained perceivers, will be reported. [Funding from NSERC.]

2pSC13. Testing symmetry of temporal window of integration between vibrotactile and auditory speech information in voiced phoneme perception. Tzu Hsu Chu (Dept. of Linguist, Univ. of Br. Columbia, 2329 West Mall, Vancouver, BC V6T 1Z4, Canada, chusophy@alumni.ubc.ca), David G. Marino (Dept. of Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), Hannah Elbaggari (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), Karon MacLean (Dept. of Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), and Bryan Gick (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

The temporal window for enhancement in cross-modal integration is asymmetrical, and does not require synchrony of stimuli for integration; this temporal offset has been linked to differences in relative signal speed as can be seen in audio-visual integration [Munhall *et al.* 1996. *Perc. Psychophys.* 58: 351] and audio-aerotactile integration [Gick *et al.* 2010. *Journ. Acoust. Soc. Am.* 128: EL342] of speech. However, as vibrotactile cues normally accompany acoustic cues, no difference in signal speed—and no concomitant perceptual asymmetry—is experienced between these two modalities. Contrary to previous research, we predict a symmetrical window for integration. A portable voice-coil transducer is used to produce vibrotactile stimuli, similar to the laryngeal vibrations normally felt in voiced speech. Results of an experiment will be presented in which participants are asked to discriminate between minimal pairs in noise with the device between their thumb and index finger, and in which audio and vibrotactile stimuli are presented in different orders and at different temporal offsets. Implications will be discussed for theories of cross-modal integration. [Funding from NSERC.]

2pSC14. The role of speaker identification in auditory and visual speech recognition: Evidence from twin speech. Madeline Petrich, Macie Petrich, Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover W225, Athens, OH 45701, leec1@ohio.edu), Seth Wiener (Modern Lang., Carnegie Mellon Univ., Pittsburgh, PA), Margaret Harrison, and Sarah Greenlee (Commun. Sci. and Disord., Ohio Univ., Athens, OH)

Understanding the content of speech does not require identifying speaker voices. Similarly, identifying speaker voices does not require understanding the content of speech. However, ample evidence has shown that speaker voices affect auditory word recognition. Familiarity with speaker voices also affects audiovisual speech recognition. This study examines whether familiarity with twin speakers, who have highly similar voices, affects the magnitude of form priming and that of the McGurk effect. In Experiment 1, participants who were familiar or unfamiliar with the twin speakers performed lexical decision and voice discrimination on pairs of auditory words that were repeated or unrelated. In Experiment 2, the same participants were asked to identify syllables in which auditory and visual information was congruent or incongruent (e.g., auditory /ba/ with visual /ga/). The auditory and visual information was also mixed between the speakers. Experiment 1 showed that familiarity resulted in reduced priming in voice discrimination but not lexical decision. Experiment 2 showed that familiarity did not result in reduced McGurk effect. It appears that the benefit of familiarity in processing auditory and visual speech is compromised when processing highly similar voices. [Work supported by Ohio University PURF and CHSP Student Research Grant.]

2pSC15. Auditory lexical decision in the wild. Benjamin V. Tucker, Filip Nenadic, and Matthew C. Kelley (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

The present report describes a version of the MALD (Massive Auditory Lexical Decision) database which investigates listener performance in the often crowded TELUS World of Science—Edmonton (TWOSE) on tablets with headphones, with most participants experiencing varying levels of uncontrolled distraction. A total of 533 listeners participated in the experiment (ages 4–86; 51% female; 81.8% native English). Stimuli were a randomly selected subset of 2000 words and 2000 pseudowords extracted from the MALD project (Tucker *et al*., 2018), recorded by a single speaker. The stimuli were further randomly divided into 20 lists, each containing 100 words and 100 pseudowords, with no practice stimuli. Each participant was presented with a single list, with the entire experimental session usually lasting between five and ten minutes. TWOSE-MALD participants, unsurprisingly, perform worse than listeners in a laboratory setting in terms of both accuracy and response latencies. The relationship between the standard predictors and response latency remains the same. We also find age related effects indicating that accuracy increases rapidly (from ages 4 to 14) and slowly plateaus, while response latencies rapidly decrease until participants reach their early twenties, after which a steady increase is noted as participants' age increases.

2pSC16. The time course of recognition of reduced disyllabic Japanese words: Evidence from pupillometry with a Go-NoGo task. Yoichi Mukai, Benjamin V. Tucker, and Juhani Järvikivi (Dept. of Linguist, Univ. of AB, 3-26 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, mukai@ualberta.ca)

While much attention has been paid to the importance of reduction in spoken word recognition, fewer studies have investigated the effect of reduction over time. Thirty-eight participants' pupillary responses were measured during the perception of Japanese disyllabic words as they performed a Go-NoGo task. We used 226 lexical items, each of which contained both reduced and citation forms of the words. All stimuli consisted of a word-medial nasal or voiced stop. Results demonstrate that the overall amount of cognitive effort required to process reduced forms was higher than that of canonical forms. That is, greater pupil dilation was observed for reduced forms than for citation forms. This result is in line with previous research (e.g., Tucker, 2011). Specifically, pupil dilation was greater with reduced forms in the time window of 436 ms to 2000 ms after the onset of stimuli. Our results also indicate that the pattern of pupil dilation over time with reduced forms differs from citation forms, indicating that reduced forms show a later onset and offset of peak dilation and more gradual constriction of pupil compared to citation forms.

2pSC17. Multimodal recognition of interrupted speech: Benefit from text and visual speech cues. Rachel E. Miller, Courtney Strickland, and Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1229 Marion St., Columbia, SC 29201, remiller@email.sc.edu)

Presenting degraded speech with visual cues facilitates speech recognition. This benefit is observed for visual speech cues that are perceptually correlated with the auditory signal, as well as for text cues that delay integration until a later cognitive-linguistic processing stage. However, it is not clear how the benefit compares between these two types of degraded multimodal presentations. The current study examined how listeners integrate visually interrupted text or visual speech cues with acoustically interrupted speech. In Experiment 1, text was periodically interrupted by white space at visual interruption rates that were associated with the auditory interruption rate of speech. In Experiment 2, videos were visually interrupted by grey frames. The synchrony of audio-visual interruption was also manipulated by presenting visual cues in-phase or 180^o out-of-phase with speech interruptions. For both experiments, speech was low-pass filtered at 2000 Hz. Preliminary results indicate that listeners obtain a benefit from both visual speech and text cues. In addition, performance is affected by the interruption rate of speech, with minimal performance obtained around an interruption rate of 2 Hz. Supplementing speech with incomplete visual cues can improve sentence intelligibility and compensate for degraded speech in adverse listening conditions. [Work supported, in part, by NIH/NIDCD.]

2pSC18. Phonetic recalibration of visual and auditory speech by visual sentences. Josh Dorsi (Psych., UC Riverside, 34875, Winchester, NY 92596, jdors002@ucr.edu), Lawrence Rosenblum, and Sharon Chee (Psych., UC Riverside, Riverside, CA)

Phonetic recalibration is a form of perceptual learning in which experience with an ambiguous speech segment accompanied by disambiguating context biases subsequent perception of that segment. For example, presenting an auditory "f" final word (i.e., "Witlof"; Dutch for "Chicory") with the final fricative replaced with a sound between /s/ and /f/ results in participants subsequently categorizing more items from a /f/-/s/ continuum as /f/ (Norris, McQueen, Cutler 2003). A similar effect is found for ambiguous visual speech that is accompanied by clear auditory speech (Baart & Vroomen 2010), and for ambiguous auditory speech that is accompanied by clear visual speech (e.g., Vroomen & Baart, 2009). It is unknown whether visual speech alone can recalibrate visual speech segments or whether recalibration in one modality can be transferred to another modality (but see, Dias & Rosenblum 2016). In Experiment 1 of this study we found that sentence context disambiguated a silent visual ambiguous /s/f/ fricative and resulted in visual recalibration. Experiment 2 used these same silent sentences but measured recalibration on an auditory continuum. Preliminary results suggest that recalibration effects can be transferred across modalities in this way. These experiments have implications for theories of amodal perceptual learning.

2pSC19. Lexical access in the face of degraded speech: The effects of cognitive adaptation. Francis Smith and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, W311 Seashore Hall, Iowa City, IA 52245, francis-smith@uiowa.edu)

Hearing impaired listeners use cognitive adaptations to cope with degraded input. Here, we ask if they adapt processes that normal hearing listeners use to cope with the fact that speech unfolds over time, creating brief periods of ambiguity. Normal listeners cope with this ambiguity by activating multiple lexical candidates which compete for recognition (McClelland & Elman, 1986). These competition dynamics change when processing degraded input (Brouwer & Bradlow, 2016; McMurray, Farris-Trimble, & Rigler, 2017; McQueen & Huettig, 2012), but it is unclear whether this reflects the degraded input itself or cognitive adaptation. In two visual world paradigm experiments, listeners heard different levels of degraded (noise-vocoded) speech. Experiment 1 manipulated the level of degradation either in blocks or randomly interleaved across trials. Interleaving led to processing delays beyond that of the level of degradation: we found switch-costs when degradation levels differed between trials. This suggests differences in lexical dynamics are not solely due to degradation in the input. In Experiment 2, a visual cue indicated the level of degradation before each trial. This reduced the processing delays and switch costs, suggesting participants adapted before the auditory input. These experiments support a role for central processing in dealing with degraded speech input.

2pSC20. Integration of speech information (or not) across electric and acoustic modes in hearing impaired listeners. Michael Seedorff (Biostatistics, Univ. of Iowa, Iowa City, IA) and Bob McMurray (Psych., Univ. of Iowa, E11 SSH, Iowa City, IA 52242, bob-mcmurray@uiowa.edu)

Many people with hearing loss use two cochlear implants (bilateral CIs), or combine a CI and hearing aid (bimodal). While the addition of a second CI or acoustic input improves speech perception (particularly in noise), it is not well understood how listeners fuse these disparate inputs. We addressed this with a duplex perception experiment. Listeners heard the four cardinal vowels from six talkers. In the duplex condition, the first formant was presented to the CI, and the second to the other CI or hearing aid (and vice versa). In normal listeners ($N = 14$), accuracy under duplex presentation (across ears; $M = 0.80$) did not differ from combined presentation (both formants, both ears, $M = 0.76$, $p = 0.14$), but performance dropped to near chance with isolated formants ($M = 0.40$, $p < 0.001$). CI users (Bilateral: $N = 7$; bimodal: $N = 11$; single-side-of-deafness [SSD]: $N = 7$) performed well in combined presentation (Bilateral: $M = 0.72$, Bimodal: $M = 0.70$, SSD: $M = 0.83$), and poorly for isolated formants (Bilateral: $M = 0.42$,

Bimodal: M = 0.39, SSD: M = 0.41). Duplex presentation was significantly poorer than combined (Bilateral: M = 0.54, p = 0.0078; Bimodal: M = 0.55, $p<0.001$; SSD: Duplex: 0.48, $p<0.001$), though better than isolated formants ($p<0.05$). Thus, CI users may not fuse inputs well at an auditory level, and the benefits of acoustic + electric hearing may derive from central integration.

2pSC21. The effect of perceptual similarity, frequency, and phonotactic restriction in loanword adaptation. Yang-Yu Chen (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, Hsinchu, Taiwan, Taiwan) and Yu-an Lu (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, F319, Humanities Bldg. 2, 1001 University Rd., Hsinchu 30010, Taiwan, yuanlu@nctu.edu.tw)

Mandarin speakers tend to adapt an intervocalic nasal as either an onset of the following syllable (e.g., *Bruno*à *bù.**lǔ.nu**ò*) or as a nasal geminate (e.g., *Daniel* à *d**ān.n**í.ěr*) (Huang & Lin 2013, 2014). Two forced-choice identification experiments were conducted to test the effects of nasalization (whether the pre-nasal vowel bears stress or not) and duration (whether the pre-nasal vowel is lax or tense): Would stronger nasalization and shorter duration encourage higher nasal gemination rate? The results showed that Mandarin speakers' choice of repairs was indeed biased by the different phonetic manipulations, suggesting an effect of perceptual similarity. Moreover, the overall preference for the V.NV form over the VN.NV form suggests an influence from the native syllable type frequency (open syllables being more frequent than closed syllables). The across-the-board higher VN.NV responses for lax than for tense vowels regardless of the phonetic manipulations are attributed to the possibility that Mandarin speakers might have perceived the tense vowels as diphthongs (i.e., English /e/ to [ej], /o/ to [ow]) and inserting a nasal coda is illegal in this contexts (*CVGN). That is, the findings suggest that the variations in loanword adaptation were guided by perception, frequency, as well as phonotactics.

2pSC22. Effects of delay time and intensity difference on recognition of overlapped speech sound. Shigeaki Amano (Faculty of Human Informatics, Aichi Shukutoku Univ., 9 Katahira, Nagakute, Aichi 480-1197, Japan, psy@asu.aasa.ac.jp), Kimiko Yamakawa (Shokei Univ., Kikuchi-gun, Kumamoto, Japan), and Katuhiro Maki (Faculty of Human Informatics, Aichi Shukutoku Univ., Aichi, Japan)

Speech sound transmitted from an outdoor loudspeaker is sometimes difficult to recognize because it is overlapped by a time-delayed speech sound from other loudspeakers. This difficulty is assumed to depend on a delay time and an intensity difference between the original and overlapping time-delayed sounds. To clarify the effects of a delay time and an intensity difference on speech recognition, a listening experiment was conducted with 21 Japanese adults using 105 Japanese spoken words in a carrier sentence, with delay times and intensity differences ranging between 0–250 ms and 0–9 dB, respectively. The experiment revealed that recognition ratios are significantly lower for delay times of 100–250 ms than for 0 ms. Similarly, the ratios are significantly lower for intensity differences of 0–6 dB than for 9 dB. These results suggest that speech sound from an outdoor loudspeaker is difficult to recognize in a large area where the difference of distances from two loudspeakers ranges between 34 and 85 m and the intensity difference between the original and overlapping time-delayed sound is less than 6 dB. [This study was supported by JSPS KAKENHI Grant Numbers JP15K12494, JP15H03207, JP17K02705, and by Aichi-Shukutoku University Cooperative Research Grant 2017-2018.]

2pSC23. Different musical experiences differentially modulate the auditory-motor processing of feedback errors during vocal production. Wenda Wang and Hanjun Liu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

The present study investigated the effects of musical experiences on auditory-motor control of pitch feedback errors during vocal production. Thirty-four female musicians who were assigned to a group of professional signers (n = 17) and a group of instrument players (n = 17) and seventeen female non-musicians aged 18–29 years participated in the present study. All participants vocalized a vowel sound /u/ for about 5-6 seconds while hearing their vocal pitch unexpectedly shifted -50 or -200 cents, and their vocal compensations for pitch perturbations were measured and compared. The results showed significantly larger compensatory vocal responses produced by non-musicians when compared to professional singers ($p<0.001$) and instrument players ($p<0.008$). Moreover, instrument players produced significantly larger vocal compensations than professional singers (p = 0.026). The observed effects of musical experiences were independent of the size of pitch perturbation. These findings provide the first behavioral evidence for the differential modulation of vocal compensations for pitch perturbations by different musical experiences, which may be related to the difference between singers and players in extensive vocal training.

2pSC24. Influence of word size and tonal sequence probabilities on Mandarin segmentation errors. Amy LaCross (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., PO Box 870102, Tempe, AZ 85287, amy.lacross@asu.edu), Jordan Sandoval (Linguist, Western Washington Univ., Bellingham, WA), and Julie Liss (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

The largely effortless process of segmenting a continuous speech stream into words has been shown cross-linguistically to be influenced by implicit knowledge about the distributional probabilities of phonotactics, stress assignment, and word size. Work in tonal languages provides compelling evidence that tones are also likely to be a source of probabilistic information for speakers of these languages. It has been shown that probability of syllable + tone combinations play a role in speech processing in Mandarin (Wiener & Ito, 2016; 2015; Wiener & Turnbull, 2013). Work in Cantonese has also demonstrated that transitional probabilities of lexical tones paired with vowels aid listeners in segmentation (Gomez, *et al*., 2018). Here we conducted two perception experiments with fifty native-Mandarin speakers and manipulated two potential segmentation cues: word size and tonal sequence probability. Contrary to our hypothesis that participants would make segmentation errors which reflect the most probable word size in Mandarin (two-syllable words) with highly probable tonal sequences, participants' errors were overwhelmingly three-syllable words with low probability tonal sequences. This suggests that biases towards segmentation errors are not predictable based on straightforward probabilities.

2pSC25. Concurrent aero-tactile stimulation does not bias perception of VOT for non-initial stops. Dolly Goldenberg (Linguist, Yale Univ., 192 Foster St., Apt 1, New Haven, CT 06511, dolly.goldenberg@yale.edu), Mark Tiede, and D. H. Whalen (Haskins Labs., New Haven, CT)

Previous work has established that puffs of air applied to the skin and timed with listening tasks bias the perception of voicing in onset stops by naive listeners (Gick and Derrick, 2009; Goldenberg *et al*., 2015). While the primary cue for the voicing contrast in stops is VOT (Lisker and Abramson, 1964), in English aspiration typically functions as a cue foot initially. This study tests the effect of air puffs on perception of voicing for English stops in a non-foot-initial context ("apa/aba") using VOT continua. Goldenberg *et al*. (2015) have shown that listeners are sensitive to aero-tactile effects only when these are congruent with the expected contrast (i.e., in VOT but not vowel quality distinctions). Since VOT is generally non-contrastive for English stops that are not foot-initial, air puffs were not expected to affect perception in the current case, and indeed, of 22 participants (11 females; mean age 34.2) tested, 20 showed no effect. Comparison of this null result to the significant bias observed in the earlier (foot-initial context) study extends the finding that, for aero-tactile stimulation to bias perception, the cues must be consistent with those expected in production of the perceived sounds.

2pSC26. Bimodal classification of English allophones employing acoustic speech signal and facial motion capture. Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl), Szymon Zaporowski, and Bozena Kostek (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland)

A method for automatic transcription of English speech into International Phonetic Alphabet (IPA) system is developed and studied. The principal objective of the study is to evaluate to what extent the visual data related

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to lip reading can enhance recognition accuracy of the transcription of English consonantal and vocalic allophones. To this end, motion capture markers were placed on the faces of seven speakers to obtain lip tracking data synchronized with the audio signal. 32 markers were used, 20 of which were placed on the speaker's inner lips and 4 on a special cap, which served as the point of reference and stabilized the FMC image while post-processing. Speech samples were simultaneously recorded as a list of approximately 300 words in which all English consonantal and vocalic allophones were represented. Different parameterization strategies were tested and the accuracy of vocalic segments recognition in different experiments was analyzed. The process of multimodal feature extraction is explained, the applied feature selection methods are presented and the obtained results are discussed. Further challenges related to the bi-modal feature extraction process and neural network-based decision systems employment are discussed. [Research sponsored by the Polish National Science Centre, Dec. No. 2015/17/B/ST6/01874.]

2pSC27. Effects of spatiotemporal manipulation of audiovisual speech on the perception of prosodic structure. Robert Fuhrman (Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, robert.fuhrman@alumni.ubc.ca)

Work in audiovisual speech processing (AVSP) has established that the availability of visual speech signals can influence auditory perception by improving the intelligibility of speech in noise (Sumby and Pollack, 1954). However, exactly which aspects of visible signals are most responsible for this enhancement remains an open question, although convergent evidence along several lines suggests that visible information may reflect a common articulatory-acoustic temporal signature, and that the multi-modal availability of this temporal signature is at the root of this effect. We evaluated this hypothesis in a perceptual study using simple talking face animations whose motion is driven by a signal derived from the collective motion of perioral structures of an actual talker. We applied spatial and temporal manipulations to the structure of this driving signal using a biologically plausible model that preserves the smoothness of the manipulated trajectory, and tested whether these kinematic manipulations influenced the perception of linguistic prominence, an important component of the timing and rhythm (prosody) of speech. The data suggest that perceivers are sensitive to these manipulations, and that the cross-correlation between the acoustic amplitude envelope and the manipulated visible signal was a strong predictor of the perception of prominence.

2pSC28. Integration of literal meaning of emotional phrases with vocal emotion: Comparison between Japanese and North Americans. Sumi Shigeno (Psych., Aoyama Gakuin Univ., 4-4-25 Shibuya, Shibuya-ku, Tokyo 150-8366, Japan, sshigeno@ephs.aoyama.ac.jp)

This study investigated how the display rules of emotional expression apply in vocal-only communication depending on speakers' and listeners' cultural backgrounds and their similarities and differences. A comparison was made between Japanese and North American participants, who listened to emotional phrases spoken in their native language and a non-native language (English and Japanese, respectively). The participants were instructed to listen carefully to the recordings and judge the speakers' true emotions. The speakers had been asked to express an emotion with their voice that was either congruent or incongruent with the emotion contained in the literal meaning of speech (ELMS), which, in English, corresponded to: "eleven-thirty," "good afternoon," "congratulations," "I love it," "I'm going to cry," and "my heart is breaking." The speakers spoke these utterances in neutral, happy, and sad voices. The results indicated that the Japanese listeners integrated the ELMS with the vocal emotions when the speakers were Japanese but judged the speakers' emotions based on voice alone when the speakers were North American. This implied that Japanese participants could infer the speakers' true emotions even when the ELMS was incongruent with vocal emotion. The results obtained from the Japanese and North American speakers and listeners were compared.

2pSC29. Acoustic features of intelligible speech produced under reverberant environments. Rieko Kubo and Masato Akagi (JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, rkubo@jaist.ac.jp)

Clear speech has been shown to have an intelligibility advantage over conversational speech in reverberant environments. Speech produced in reverberant environments is also reported to be more intelligible than ones produced in normal environments. Speakers adapt to the environments by modifying their articulation, and acoustics of speech. The speech produced under reverberant conditions should be modified similarly with the clear speech. Identifying the distinctive features which are shared between these speech types gives insights into which acoustic features are robust cues even in reverberant environments. In the present study, we investigated voices uttered by speakers who produced Japanese words under reverberant conditions (T60 = 1 s–5 s). Speech intelligibility tests, and acoustic analyses were performed. The results showed that, first, speech produced under long T60 is more intelligible in reverberant environments. Second, the intelligible speech has expanded vowel space with high F2 for front vowels and low F2 for back vowels, and high F1 for open vowels and low F1 for close vowels. Brief formant transitions were also found. These articulatory/acoustic modifications are likely to be enhancements of significant perceptual cues. The acoustic/articulatory features of each vowel become clearer (hyper-articulated speech; i.e., clear speech.) Moreover, perceptual compensation to reduce co-articulation effects should become easier.

2pSC30. Speech intelligibility testing of general service respirators. Christoph Hoeller, Markus Müller-Trapet, and John S. Bradley (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

The communication performance of three general service respirators was evaluated. The evaluation procedure followed a NIOSH standard test procedure which is based on a modified rhyme test (MRT), wherein human participants are placed in a controlled acoustic environment and are asked to read out lists of test words. The number of words correctly identified by a listener panel when the talker is wearing the respirator is compared to the number of words correctly identified without the respirator, giving an overall performance rating. During the implementation of the experimental procedure, it was noted that the NIOSH test procedure is ambiguous on a number of potentially significant acoustical issues, e.g., the reverberation time of the test environment, the background noise spectrum at the listener position, and the talker voice level. This talk will present details of the study and discuss the relevant issues related to speech intelligibility testing of respirators that were evaluated as part of this project.

2pSC31. Measuring the effect of speaker ethnicity on online perception: Evidence from a response time study. Noortje de Weers (Linguist, Simon Fraser Univ., 316- 4453 Main St., Vancouver, BC V5V0A2, Canada, ndeweers@sfu.ca)

The effect of speaker ethnicity on speech perception remains unclear. Proponents of the bias hypothesis maintain that presenting an Asian or Mexican face to American participants triggers a certain kind of bias that could result in worse comprehension and even hearing a non-existent 'foreign accent.' Exemplar-based studies, on the other hand, have proposed that these findings merely reflect a mismatch between listeners' expectations and the actual speech signal. While previous studies all used post-perceptual, offline tasks to examine the effect of speaker ethnicity on speech perception, this study made use of an online task instead. Thirty-two native English participants completed a speeded audio-visual sentence verification task, for which they had to classify statements as true or false. The utterances were paired with a photograph of an Asian face, a White face, or a fixation cross, and were presented in a mixed design. Both correctness scores and response times for all the different face-voice pairings were recorded. Results suggest that online processing was not affected by speaker ethnicity, as response times did not differ as a function of the various face-voice pairings. Additional findings showed that the foreign-accented voices took significantly longer to process than the native voices, and that false statements took longer to answer than true statements.

2pSC32. Perception of sexual orientation through speech: Generational differences. Lily Obeda, Melanie Putman, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Previous research has shown that listeners can perceive talkers' sexual orientation at a greater than chance levels from the acoustic characteristics of their speech (Munson *et al*., J. Phonetics [2006]). Recently, Obeda and Munson (2018, J. Acoust. Soc. Am. 143, 1924) compared the perception of 44 gay, lesbian, and heterosexual people's sexual orientation through speech by the young adults reported in Munson *et al.* 2006 (whose data were collected in 2003) to a group of young adults tested in 2018. The new listeners were matched in age to the 2003 listeners. A number of differences between the two cohorts were found. In particular, the 2018 listeners were less willing to judge heterosexual women's speech as 'sounding heterosexual'. In this presentation, we compare both sets of data to the performance of a new set of listeners who are matched in birth year to the 2003 listeners, and who are currently 33–48 years old. Collection of those data is ongoing. Comparing this new group to the two other groups will allow us to examine whether the apparent generational change observed by Obeda and Munson has led listeners like those in 2003 to reevaluate their perception of sexual orientation through speech in 2018.

2pSC33. Regional dialect and talker sex information in high-frequency region. Robert A. Fox, Ewa Jacewicz, and Magan McClurg (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

High-pass filtering, in which the higher frequencies are retained and the lower frequencies are eliminated, has not been used in speech perception research as often as low-pass filtering because most intelligibility cues have been assumed to reside in the low-frequency region, up to 4 kHz (French and Steinberg, 1947). Recent research provided new evidence that listeners can also utilize high-frequency energy in the perception of voice, talker sex, and speech characteristics such as naturalness, pleasantness, and clarity, which improves speaker and word recognition in noise. Given the potential for the existence of accessible linguistic and paralinguistic information, the current study asks whether perception of regional dialect can be influenced by spectral content in the high-frequency region, and how robust this information is—in males and females—when low-frequency cues are unavailable. Listeners from Ohio heard phrases produced by males and females from Ohio and Western North Carolina high-pass filtered at 700, 1175, 1973, 3312, and 5560 Hz. Each higher filter provided increasingly fewer cues about dialect whereas sex identification remained relatively high. Female speech provided significantly more dialect cues than male speech when more spectral information was available (filter cut-offs at 1175 and 1973 Hz, but not at 700 Hz).

2pSC34. Distinguishing Dick from Jane: Children's voices are more difficult to identify than adults' voices. Natalie Fecher, Angela Cooper, and Elizabeth Johnson (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5G4K2, Canada, angela.cooper@utoronto.ca)

Previous research on talker recognition has focused on the processing of talker identity in adult voices; however, little is known about how listeners identify child voices. This study examined adult listeners' ability to differentiate and identify child voices (2.5-year-olds) and adult voices. In Experiment 1, native English-speaking listeners completed an AX voice discrimination task with both child and adult voices. Results revealed that listeners were significantly worse at discriminating between child voices relative to adult voices, regardless of the fact that child voices were mixed gender and adult voices were all female. In Experiment 2, we examined whether listeners could learn to identify child and adult voices. Adult listeners completed an adaptive training paradigm, where they learned to identify 4 child voices on one day and 4 adult voices on a separate day. Preliminary results indicate that with training, listeners can learn to identify voices from both age groups above chance. However, listeners were still faster at learning and more accurate at identifying adult relative to child voices. Taken together, these findings are an initial step towards understanding how talker recognition is influenced by the acoustic-phonetic characteristics and articulatory capabilities that vary with talker age.

2pSC35. Visual gender biases in English stop voicing perception. Connie Ting, Rachel Soo (Univ. of Toronto, 100 St. George St., Toronto, ON, Canada, connie.ting.ct@gmail.com), and Jessamyn L. Schertz (Univ. of Toronto, Mississauga, ON, Canada)

Listeners leverage visual information, such as perceived speaker gender, during speech perception. While work has shown visual gender biases on fricative (Johnson & Strand, 1996) and vowel perception (Johnson *et al*., 1999), its effects on stop voicing perception are understudied. We present an identification task where visual gender primes (male/female portraits) were used to investigate perceived speaker gender effects on English stop voicing. Subjects (n = 22) identified 'pa'/'ba' syllables from an 11-step VOT continuum (0–50 ms) in 5 pitch levels (high: 250 Hz, 230 Hz, mid: 170 Hz, and low: 130 Hz, 110 Hz). High and low pitch tokens were always preceded by a female and male portrait, respectively. Mid pitch tokens were preceded by female or male images in separate blocks. Given that f0 is higher following voiceless obstruents (Lehiste & Peterson 1961; Mohr 1971), if listeners utilize visual gender cues, acoustically ambiguous tokens paired with a male portrait should elicit more voiceless "pa" responses than those paired with a female portrait. Preliminary results pattern in the expected direction; mid tokens are perceived as "pa" more when paired with a male (15–25 ms = 79% "pa" response) than a female face (73% "pa" response); however, this effect was not statistically significant.

2pSC36. Effects of emotional tones of voice on the acoustic and perceptual characteristics of Mandarin tones. Huishan Chang (Dept. of Audiol. and Speech-Lang. Pathol., Asia Univ., No. 500, Liufeng Rd., Wufeng Dist., Taichung 41354, Taiwan, sandychang0217@gmail.com), Shuenn-Tsong Young (Holistic Education Ctr., Mackay Medical College, New Taipei, Taiwan), Pei-Chun Li (Dept. of Audiol. and Speech-Lang. Pathol., Mackay Medical College, New Taipei, Taiwan), Woei-Chyn Chu (Dept. of Biomedical Eng., School of Biomedical Sci. and Eng., National Yang-Ming Univ., Taipei, Taiwan), and Cheng-Yu Ho (Holistic Education Ctr., Mackay Medical College, New Taipei, Taiwan)

Emotional tones of a speaker's voice and lexical tones involve similar acoustic correlates, but only lexical tones could change the meaning of a word in tonal languages. The purpose of this study is to investigate the interaction between these two types of tonal variations by examining the acoustic and perceptual characteristics of the four Mandarin tones across different emotional tones of voice. In experiment 1, acoustic analyses of fundamental frequency, mean amplitude, and duration was conducted on a syllable with the four tones produced in a carrier phrase with four different emotional tones of voice (anger, fear, happiness, and sadness). The same acoustic measures were also taken on the Mandarin neutral tone produced with the four emotional tones. In experiment 2, speech materials from experiment 1 were used to investigate the effects of the emotional tones on the perception of Mandarin tones. The results showed that all four emotional tones had significant effects on the acoustics and perception of Mandarin tones. These findings suggest that emotional tones of voice impact both acoustic and perceptual characteristics of lexical tones.

2pSC37. Acoustic emotion recognition using spectral and temporal features. Tejal Udhan (Dept. of Elec. and Comput. Eng., Florida State Univ., 2525 Pottsdamer St., Tallahassee, FL 32310, tu13b@my.fsu.edu) and Shonda Bernadin (Dept. of Elec. and Comput. Eng., FAMU- FSU College of Eng., Tallahassee, FL)

In this paper, utility of different low- level, spectral and temporal features is evaluated for the task of emotion recognition. The aim of an ideal speech emotion recognition system is to extract features that are representative of the emotional state of speaker. Pitch, intensity, frequency formants, jitter, and zero crossing rate are five features proposed for characterizing four different emotions, anger, happy, sadness, and neutral. Low- level spectral and temporal features have ease of calculation and limit the complexity of emotion recognition systems since they are commonly single dimensional features. A decision-tree based algorithm is designed for characterizing emotions using these acoustic features. It has been proven that various aspects of a speaker's physical and emotional state can be identified by speech alone. However, the accuracy of such analyses has not been optimized due to acoustic variabilities such as length and complexity of human

speech utterance, gender, speaking styles, and speech rate. Since speech emotion recognition is a developing and challenging field, most powerful features for emotion recognition are not yet defined; hence, investigating the utility of selected features for emotion recognition is an important task.

2pSC38. A methodological framework to derive the mental code of emotional and social inferences in sound. Emmanuel Ponsot (ENS, 1 Pl. Igor Stravinsky, Paris 75004, France, ponsot@ircam.fr)

Most research carried out so far in auditory emotion has been based on theory-driven experiments. Such experiments, which involve a limited number of conditions, help in refining existing models or theories but cannot expose elements or processes that were not expected and targeted initially. Here, I will present a methodological framework based on a data-driven approach that allows to probe the mechanisms of auditory cognition in a space that is not constrained by a priori hypotheses. This framework uses signal-processing algorithms for manipulating the acoustical characteristics of an input sound and create a huge number of parametrically-manipulated, natural expressive variations of this sound, which are then used as stimuli in psychophysical experiments employing a reverse-correlation technique. I will present different contexts in which this framework has been used, in particular to explore the processing of speech prosodic dimensions in the formation of social impressions. Because this approach offers a principled way to reverse-engineer any high-level judgment in any individual, it should be helpful to understand the algorithms that the brain uses to build high-level emotional or social impressions from the acoustical characteristics of sound in their complexity.

2pSC39. Study on the relationship between modulation spectral features and the perception of vocal emotion with noise-vocoded speech. Zhi Zhu, Ryota Miyauchi (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., 5-201 Student House, 1-1 Asahidai, Nomi, Ishikawa 9231211, Japan, zhuzhi@jaist.ac.jp), Yukiko Araki (Kanazawa Univ., Kanazawa, Ishikawa, Japan), and Masashi Unoki (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi-shi, Japan)

Previous studies about vocal-emotion recognition with noise-vocoded speech showed that temporal modulation cues provided by the temporal envelope play an important role in the perception of vocal emotion. To clarify the exact feature of temporal envelope that contributes to the perception of vocal emotion, a method based on the mechanism of modulation frequency analysis in the auditory system is necessary. In this study, auditory-based modulation spectral features were used to account for the perceptual data collected from vocal-emotion recognition experiments using noise-vocoded speech. At first, the modulation spectrogram of the emotional noise-vocoded speech was calculated by using an auditory-based modulation filterbank. Then, ten types of modulation spectral features were extracted from the modulation spectrograms. Finally, modulation spectral features and the perceptual data were compared to investigate the contribution of temporal envelope to the perception of vocal emotion with noise-vocoded speech. The results showed that there were high correlations between modulation spectral features and the perceptual data. Therefore, the modulation spectral features should be useful for accounting for the perceptual processing of vocal emotion with noise-vocoded speech. [Work supported by JSPS KAKENHI Grant Number JP. 17J08312, and Grant in Aid for Scientific Research Innovative Areas (No. 18H05004) from MEXT, Japan.]

2pSC40. Proposals of noise-robust spoken words for a broadcast via outdoor loudspeakers. Kimiko Yamakawa (Cultural Commun., Shokei Univ., 2-8-1 Musashigaoka-kita, Kikuyo, Kikuchi-gun, Kumamoto 8618538, Japan, jin@shokei-gakuen.ac.jp) and Shigeaki Amano (Aichi Shukutoku Univ., Nagakute, Aichi, Japan)

Spoken words used in a broadcast via outdoor loudspeakers were analyzed in terms of word frequency, word familiarity, and phoneme composition to clarify their characteristics. Based on the analysis, we identified 13 original words that are frequently used but probably difficult to listen to and proposed alternative words having higher word familiarity and a lesser number of noise-intolerant phonemes. To verify the proposal, a listening experiment was conducted. Thirteen each of the original and proposed words were presented with and without pink noise (SNR = 0 dB) to 21 participants with normal hearing ability. Error ratios of word identification were obtained from participants' responses. In a with-noise condition, the error ratio of the proposed words (7.1%) was significantly lower than that of the original words (18.3%), but no difference in error ratio was observed between the original and proposed words in a without-noise condition. These results indicate that the proposed words have higher noise robustness and less listening difficulty than the original words, and that they are useful for transmitting accurate information in a loudspeaker broadcast.

2pSC41. Optimal use of the visual analog scale: Observations from ratings of emotional speech. Shae D. Morgan (Univ. of Louisville, 390 South 1530 East, Ste. 1201, Salt Lake City, Utah 84112, shae.morgan@utah.edu)

Researchers use many tools to measure participant performance during experiments. The visual analog scale has been used in many areas of research as a valid way to measure listener perception of various aspects of a signal. Ratings using the visual analog scale are commonly averaged over a number of trials to estimate listeners' perception. This presentation delves into issues which may be encountered when using a visual analog scale for measuring perception of voice (in this case, vocal emotion). An experiment was performed which gathered ratings of emotional activation and pleasantness for many stimuli of four emotion categories (angry, sad, happy, and calm) from 10 listeners. These data will be examined for patterns of variance in a time-course analysis to identify listener certainty or biases created when using the visual analog scale during a long experimental task. Implications of bias and the impact on research results will be discussed along with additional and alternative measurement options for researchers to consider.

2pSC42. It's time to collaborate: What human linguists can learn from machine linguists. Michael D. Fry (Linguist, Univ. of Br. Columbia, 2329 West Mall, Vancouver, BC V6T 1Z4, Canada, mdfry20@gmail.com)

For decades, one of the primary goals of machine learning has been to emulate human performance by learning from human behavioural data. However, machine learning has now matured to a point where humans are learning from machines—for example, consider playstyles of Go (Silver, D., *et al.,* "Mastering the game of go without human knowledge," *Nature* 550.7676(2017):354). Drawing on this idea, this project considers what speech scientists might be able to learn from considering how machines analyze the speech signal. One goal in phonetics is to break down the speech signal into units such as phonemes, syllables, and tones. Machines are also able to break down the speech signal in either a supervised or unsupervised manner. In supervised learning, the machine is trained to classify phonemes, tones, etc., using previously known labels. In unsupervised learning, the machine learns a lower-dimensional, latent representation of the speech signal and then identifies meaningful clusters in the latent space. This project takes an unsupervised approach to lexical tone identification in Mandarin. Specifically, an adversarial autoencoder and density-based clustering are used to identify the tones of Mandarin. Results contrast Mandarin lexical tones as determined by linguists and by machine learning.

Session 2pUWa

Underwater Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Unmanned Vehicles and Acoustics

Erin Fischell, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., WHOI, MS 11, Woods Hole, MA 02543

Chair's Introduction—1:00

Invited Papers

1:05

2pUWa1. Bearing and range tracking with a two-hydrophone ocean glider. Elizabeth T. Küsel and Martin Siderius (Portland State Univ., Ste. 160, 1900 SW 4th Ave., Portland, OR 97201, ekusel@pdx.edu)

A Slocum glider fitted with two hydrophones separated by a distance of approximately 3 feet recorded sperm whale sounds during an experiment off the west coast of the island of Sardinia, Italy, in the summer of 2014. Time difference of arrival analysis of their echolocation clicks provided bearing tracks. It was also found that the left-right ambiguity inherent of the bearing estimation process could be broken by the natural oscillation in glider heading. More recently, a controlled experiment on a quiet lake with a known fixed acoustic source was devised to investigate unambiguous bearing estimation and possible ranging. Simple target motion analysis (TMA) using the glider's position data resulted in range estimates of only a few hundred meters short of the true source location. Analysis of the acoustic data on the other hand, presented challenges to satisfactory TMA performance. Overall, it was found that simple bearing-only TMA analysis is possible with two-hydrophone acoustic data and small angular change of a few degrees in the heading of the observing platform. Results will be discussed with implications to marine mammal population density estimation studies, since the ability to estimate direction and range of sound-producing animals could improve estimation of detection probabilities.

1:25

2pUWa2. Relative navigation for command and control of multiple low-cost autonomous underwater vehicles. Nicholas R. Rypkema (Elec. Eng. and Comput. Sci., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 5-223, Cambridge, MA 02139, rypkema@mit.edu), Erin Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Henrik Schmidt (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

A long-held goal of autonomous underwater vehicle (AUV) research has been the coordinated use of multiple vehicles for novel applications. Unfortunately, the underwater domain poses significant challenges for navigation and communication—sophisticated navigational sensors increase vehicle price, size, and power use, and acoustic communications are limited in bandwidth and channel capacity. These factors make multi-vehicle deployments rare and limited in scale, with operators commanding each AUV individually and with limited coordination. In this work, we describe an operating paradigm that enables user-friendly command and control of multiple AUVs. Each vehicle uses a custom low-power, low-cost acoustic system to navigate and receive operator commands, consisting of a passive hydrophone array and timed acquisition and processing boards which enable the vehicle to self-localize relative to a synchronized acoustic beacon. By selecting between various beacon-transmitted signals, all vehicles can be simultaneously commanded to switch between behaviors. Additionally, a consequence of beacon-relative navigation is that movement of the beacon results in the concurrent movement of all AUVs. This system is ideal for multi-vehicle operations using inexpensive, miniature AUVs, as it does not require conventional high-cost navigational sensors or acoustic modems, and its passive nature and associated operating scheme enable the deployment of an arbitrarily large number of AUVs. [Work supported by Battelle, ONR, Lincoln Laboratory, DARPA.]

1:45

2pUWa3. Incorporating real-time acoustic ranging and glider-based Doppler measurements to aid vehicle navigation. Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, swebster@apl.washington.edu), Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Andrey Shcherbina (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Aleksandr Aravkin (Dept. of Appl. Mathematics, Univ. of Washington, Seattle, WA), Craig M. Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

In the summer of 2017, two Seagliders equipped with low frequency acoustic recorders and 1 MHz acoustic Doppler current profilers (ADCPs) were deployed in the Canada Basin as part of a large-scale acoustic tomography experiment. Acoustic tomography sources were moored at approximately 175 m depth within an acoustic duct enabling acoustic transmission to be received at long ranges. The sources transmitted at a center frequency of approximately 250 Hz every four hours, and ranges between the sources and Seaglider were

estimated in real time using a WHOI MicroModem. These ranges can be incorporated into an extended Kalman filter for navigation. In addition, glider-mounted upward-looking ADCPs recorded ~26.5 m shear profiles every 15 seconds. These overlapping profiles can be used to estimate (in post-processing) both the local current profile on a per-dive basis and the glider's relative velocity through the water. In real time, the close-range ADCP velocity measurements can be used to estimate the glider's relative (Through The Water, TTW) velocity to improve the glider's subsea position estimate. We will describe our recent developments in using these Doppler measurements to aid glider navigation, comparing the results to previous developments in range-aided navigation.

2:05

2pUWa4. Wavenumber domain acceleration for stripmap fast factorized backprojection beamforming. Timothy Marston and Daniel Plotnick (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu)

Synthetic aperture sonar (SAS) imaging is frequently used in conjunction with autonomous underwater vehicles (AUV's) to perform high resolution wide-area surveys. Wave action and currents perturb the trajectories of AUV's performing these surveys and these perturbations have important ramifications for the SAS beamforming process. Generalized beamforming algorithms operating in the time domain are very flexible with regard to trajectory non-linearities, but they also tend to have a higher computational cost than alternative frequency domain algorithms. Of the generalized beamformers one of the more efficient is fast factorized backprojection (FFBP), which leverages sub-aperture processing and a series of coordinate transformations to improve computational efficiency. FFBP can be used to beamform SAS data, however many SAS systems make use of a real aperture to boost range potential. The existence of a real aperture implies that a frequency domain beamformer could be used to bypass the first few stages of FFBP without loss of generality; however, standard frequency domain SAS beamforming algorithms are not compatible with the FFBP framework. A wavenumber domain beamformer compatible with the early-stage FFBP framework is derived and beamforming results for the modified FFBP algorithm are compared with the standard approach that operates entirely in the time domain.

2:25

2pUWa5. Monitoring of macroalgae (kelp) farms with autonomous underwater vehicle-based split-beam sonar. Erin Fischell, Timothy K. Stanton, Amy Kukulya, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., WHOI, MS 11, Woods Hole, MA 02543, efischell@whoi.edu)

The long-term goal of ARPA-E's MARINER (Macroalgae Research Inspiring Novel Energy Resources) program is to increase the scale of offshore kelp aquaculture so that biofuel may be derived from macroalgae. An autonomous underwater vehicle (AUV) system including acoustic, optic, and environmental sensors has been developed for the purposes of monitoring these large-scale kelp farms. The primary sensor for observing farm infrastructure such as horizontal longlines (from which the kelp grows) as well as kelp growth is a broadband split-beam sonar system. Structural information, such as droop in the longlines, is available from time-of-flight of the acoustic echoes. Growth of kelp is quantified from both time-of-flight and volume backscattering echo data. Experimental results from ocean measurements taken with the REMUS 100 AUV on longlines are presented, along with preliminary processing techniques for estimating longline position and macroalgae extent for vehicle control and farm mapping purposes [work supported by ARPA-E.]

2:45–3:00 Break

3:00

2pUWa6. A hybrid time and wavenumber domain algorithm for generalized beamforming with circular synthetic aperture data. Timothy Marston and Daniel Plotnick (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu)

Synthetic aperture sonar systems typically scan along a linear trajectory, however autonomous underwater vehicles (AUV's) provide a useful platform for performing scans along more complicated trajectories with specific advantages. One example is the circular scan, which provides superior wavenumber coverage and resolution for images reconstructed near the circle center. A variety of approaches have been put forward for beamforming synthetic aperture data collected along a circular trajectory, but for images that span a significant fraction of the circle radius, accelerated time-domain approaches such as fast-factorized backprojection (FFBP) have been suggested. The computational requirements for generating a fully sampled image collected around a circular aperture are extremely large, however, even for FFBP. This is primarily because of the extremely small pixel size required to maintain adequate spatial sampling as larger portions of the aperture are coherently fused in the later stages of the FFBP algorithm. A method is presented for replacing the later stages of the algorithm with a series of spectral mappings, retaining the generality of the beamformer while improving computational efficiency.

Contributed Papers

3:20

2pUWa7. Underwater source localization using a fleet of three gliders. Yong-Min Jiang (Res. Dept., NATO - STO - Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, yong-min.jiang@cmre.nato.int)

The NATO Centre for Maritime Research and Experimentation has been exploring the potentials of passive acoustic payload equipped underwater gliders for maritime intelligence, surveillance and reconnaissance missions. During the CMRE Glider Sensors and Payloads for Tactical Characterization of the Environment 2015 sea trial, it was successfully demonstrated that a fleet of three passive acoustic payload equipped gliders simultaneously detected the 'anomalous' sound in shallow water and reported back to the glider control center. This work studies the feasibility of localizing the source (the coordinates and the depth) using the information provided by the glider fleet. The results of three different source localization approaches are compared. The impact of the errors in conventional glider time and location data, as well as insufficient/incorrect knowledge of the ocean environment on underwater source localization is discussed. (*Work funded by NATO-Allied Command Transformation.*)

3:35

2pUWa8. A method for passive localization of three-dimensional position. Mingyu Song, JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Beijing CuiWei St., Beijing, China, 56608812@qq.com), and Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang, China)

The movement operation of the machine installed on the ship and the ship itself will disturb the surrounding medium, and the vibration's transmission forms the sound field of a ship. Analyzing the sound field of the target source has profound significance. The target three-dimensional position can be known according to the source of the noise radiated sonar-gram, this can be used for positioning the moving target source and tracking its trajectory A method using focused beam-forming to measure the distribution image (under-water image) of ship's noise source which depth is unknown is presented in this paper. This is for the past open literature mentioned using a one-dimensional horizontal array for two-dimensional plane of the target source localization defects. The arrays cloth on the seafloor, when basing on the principles of geometrical acoustic and the superposition of important early reflected sound energy on direct sound energy, the three-dimensional coordinates of the target would be calculated. For the mentioned principle, this paper carried out the related analyzes simulation and the acoustic image verifies the feasibility of the algorithm. The distance would bring in focused-peak broadening, positioning-error increasing, by dividing frequency band, the phenomenon could be restrained and the problem could be improved. The results show that the method can locate the target quickly and accurately, and it can be used in the fixed arrays offshore areas.

3:50

2pUWa9. Passive detection parameter measurement of azimuth and frequency based on single vector sensor. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), Mingyu Song, and JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Beijing, China)

This paper mainly studies the measurement technology of passive detection parameters based on single vector sensor. Ship radiated noise contains a lot of target information, so the extraction of the line noise of the ship's radiated noise is of great significance in the measurement of passive detection parameters. The radiation noise of the ship is a broadband spectrum modulated by the spectral signal envelope. In this paper, the mathematical model of the radiation noise of the ship is simulated and the demodulation of the ship's radiated noise is carried out by using the square demodulation and the absolute value demodulation. The average azimuth and complex intensifier are used to estimate and analyze the target azimuth, and the estimated angle is statistically processed by histogram and weighted histogram. For the frequency estimation technique, the adaptive frequency estimator based on the adaptive notch filter and the adaptive LMS (least mean square) algorithm is chosen to study the target signal frequency.Through the verification of the simulation experiment, the azimuth estimation technique and the adaptive frequency estimation technique are feasible. By estimating the azimuth sequence and frequency sequence of the target radiation signal, the relevant information of the target can be obtained, so it is of great significance in the research of underwater acoustic detection technology.

4:05

2pUWa10. Mapping underwater noise, detection of ships and cetaceans using a *SeaExplorer* glider at a basin level: Feedback from the first 1000 km-long acoustic exploration of the Western French Mediterranean Sea. Laurent Beguery (ALSEAMAR, Rousset, France), Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Romain Tricarico (ALSEAMAR, Rousset, France), Cédric Gervaise, Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France), cathy anna valentini poirier, and Pierre Boissery (Agence de l'Eau RMC, Marseille, France)

In response to concerns about the impact of man-made noise on marine ecosystems, researchers and environmental managers are currently collecting *in situ* measurements of oceanic noise levels. The objectives of *in situ* measurements are to provide the acoustic signatures of individual ships, with the use of AIS databases, to feed the models; to calibrate the model for mapping of shipping noise, and to assess marine biodiversity through the sounds emitted by marine animals (invertebrates, fishes, and cetaceans). The usefulness of the data collected depends on the duration of acquisition and the diversity of the measurements (e.g., shipping density and water depth). Gliders are ideal vehicles to collect noise level data across oceanic basins and over long time periods. Here, we show results from a *SeaExplorer* glider equipped with a high quality acoustic payload travelling for 30 days (09/15/2017-10/15/2017) along a 1000 km-long transect of the Western French Mediterranean Sea. The trajectory of the glider was chosen to sample the highest and lowest shipping densities. We here report on the statistical distribution of oceanic noise levels in the bandwidths assessed by the European Marine Framework Strategy Directive as well as on the detection of cetacean and ship sounds along the transect.

Session 2pUWb

Underwater Acoustics: Signals and Systems II

Anthony P. Lyons, Chair
Center for Coastal and Ocean Mapping, University of New Hampshire, Durham, NH 03824

Contributed Papers

1:30

2pUWb1. Effects of reverberation on estimates of synthetic aperture sonar multi-look coherence. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu), Jonathan L. King (Naval Surface Warfare Ctr., Panama City, FL), and Daniel C. Brown (Appl. Res. Lab., Penn State Univ., State College, PA)

Multi-look coherence makes use of phase information for target detection and classification by splitting the angle and frequency spectral bandwidth contained in a complex synthetic aperture sonar image into sub-bands and then estimating the coherence between the images formed from these sub-bands. Based on experimental results, it appears to be possible to separate man-made targets from interfering reverberation and clutter using coherence, as targets have features that scatter coherently as a function of angle or frequency. Characteristics of a target object may also be inferred, as sub-band coherence is a sensitive function of both angle and frequency. The expectation operation performed to obtain estimates of the complex correlation coefficient is evaluated using a spatial average, which lowers the spatial resolution of the resulting coherence map. The final resolution of a coherence image will be a function of the original image resolution, the number of sub-look images formed from the original image and the number of samples used in the expectation. In this work, we explore the effects of signal, reverberation, and noise levels on estimates of coherence using experimental data. Trade-offs in terms of signal-to-reverberation, expectation window size, and the number of sub-looks will be presented.

1:45

2pUWb2. Compressive normal mode estimation in shallow water. Yongsung Park, Woojae Seong (Seoul National Univ., Gwanak-gu, Gwanak-ro 1, Seoul National University Bldg. 36 - Rm. 212, Seoul 08826, South Korea, ysparkwin@snu.ac.kr), and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Estimation of normal modes in an ocean waveguide from acoustic data on a vertical line array involves estimation of mode shapes and their wavenumbers. Since these horizontal mode wavenumbers are sparse, compressive sensing (CS) can solve the sparse signal reconstruction problem with high accuracy. Here, grid-free CS technique, atomic norm minimization (ANM) is used to estimate the horizontal mode wavenumbers and their mode shapes. These mode shapes and wavenumbers can be estimated using the grid-free CS without *a priori* knowledge of the environment. The grid-free CS successfully extracts mode shapes and wavenumbers even with a partial water column spanning array and even in the case when the mode shape functions are not orthogonal.

2:00

2pUWb3. Analysis and interpretation of underwater acoustic data originated from the vicinity of the last known location of the Argentinian submarine ARA San Juan. Peter L. Nielsen, Mario Zampolli, Ronan Le Bras, Pierrick Mialle, Paulina Bittner, Alexander Poplavskiy, Mikhail Rozhkov, Georgios Haralabus, Elena Tomuta, Randy Bell, and Patrick Grenard (Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Office E0565, VIC, P.O. Box 1200, Vienna 1400, Austria, Peter.Nielsen@ctbto.org)

The CTBT International Monitoring System (IMS) is a world-wide network of seismic, infrasound, hydroacoustic, and radionuclide stations designed and deployed to detect nuclear explosions. Two of the IMS hydrophone stations, one in the Atlantic Ocean and one in the southern Indian Ocean, recorded signals of unknown nature which originated from the vicinity of the last known location of the San Juan submarine on 15 November 2017. To verify the accuracy of this hydroacoustic event localization, the Argentinian Navy successively deployed a controlled depth charge to the North of the last known position of the submarine. The signals from this source were also detected on the same two IMS hydrophone stations. Several techniques were employed to compare hydroacoustic features in the signals from the 15 November event and the controlled depth charge, including an assessment of spectral energy levels, cepstral analysis, azimuth and arrival time estimation of direct and reflected signals. This presentation compares the received signals, examines their main characteristics, and, through the use of acoustic modelling, identifies possible propagation paths between the source and receiver, including a plausible explanation of early arrivals hypothesized to follow propagation paths through the Antarctic ice sheet.

2:15

2pUWb4. Studying underwater sound level caused by bridge traffic in Lake Washington. Shima Abadi, Derek Flett, Ryan Berge, Jeremy DeHaan, Virdie Guy, Urooj Qureshi, and Michael Cook (Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, flettd23@uw.edu)

Underwater noise pollution due to human activities has greatly increased in recent years. There are several studies on high-intensity impulsive noises such as pile driving and seismic exploration. However, less is known about the effects of long-term exposure to low-intensity noises such as those due to bridge traffic. To characterize such noises, we have designed an underwater recording package. This package is equipped with a hydrophone to measure noise, a Raspberry-Pi to control recordings, and a cellular modem to stream near real-time data to the cloud. Data is collected in Lake Washington at two locations: (1) close to the Evergreen Point floating bridge across Lake Washington that connects Seattle to its eastern suburbs and (2) far from the bridge in the middle of the lake for comparison. The Evergreen Point floating bridge, the longest floating bridge in the world, is made of 77 large water-tight concrete pontoons and capable of carrying over 70,000 vehicles per day. The noise level measured close to this bridge is correlated with the public data on traffic load and wind speed available by the Washington State Department of Transportation.

2:30

2pUWb5. Transverse horizontal spatial coherence of explosive signals in northern South China sea. Bo Zhang, Jingyan Wang, and Yanjun Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhangbo@mail.ioa.ac.cn)

Transverse horizontal spatial cross-coherence was measured experimentally by bottom-mounted receiver arrays using explosive sources in Northern South China sea, where the water depth varies from 100 m to 1000 m in about 100 km ranges with a 200 m depth at about 75 km range. Expressed in terms of wavelengths, the coherence length is shown to be larger than 150λ at an acoustic frequency of 100 Hz in shallow water of 100 m depth. It is much greater than Carey's shallow-water experimental result 30λ estimated from array signal gain (*The Journal of the Acoustical Society of America 104, 831 (1998)*), while it is consistent with Rouseff's modeling result of a coherence length larger than 200λ (*The Journal of the Acoustical Society of America 138, 2256 (2015)*). In contrast to the good coherence in shallow water, the coherence length is measured to be only $5\lambda \sim 20\lambda$ at 100Hz in the transitional area with the array bottom-mounted at 1000m depth. Besides, both Carey and Rouseff argue that the transverse horizontal spatial coherence length depends only weakly on range. In the present study, however, the coherence length is shown to depend highly on range, and it fluctuates synchronously with the sound-field intensity while range varies.

2:45–3:00 Break

3:00

2pUWb6. Empirical modelling for derived metrics as function of sound exposure level in marine pile driving. Roel A. Müller (Acoust. & Sonar, TNO, Oude Waalsdorperweg 63, Den Haag 2597 AK, Netherlands, roel.muller@tno.nl), Michael A. Ainslie (JASCO Appl. Sci. (Deutschland) GmbH, Eschborn, Germany), Michele B. Halvorsen (CSA Ocean Sci. Inc., Stuart, FL), and Tristan Lippert (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Hamburg, Germany)

To protect marine life, piling activities at sea are subject to regulations on sound emission. Different regulatory authorities base different action criteria based on different statistics of the emitted sound pressure. Modelling efforts to predict these metrics have achieved mixed success. While the sound exposure level (SEL) can be predicted relatively well, it has proven harder to model the peak sound pressure level (Lpk) accurately. It would therefore be valuable to have a reliable way of estimating Lpk, based on a prediction of SEL. Correlations between SEL and Lpk and between SEL and rms sound pressure level (Lrms) are obtained from measurements during piling of the Luchterduinen wind farm off the Dutch west coast, and are used to assess the applicability of correlations found at other wind farms. A metastudy using data from Luchterduinen as well as the German Bight gives a more robust trend line for use at other North Sea sites. Lrms is computed in two different ways, involving the 90 % energy signal duration on the one hand and the effective signal time duration (Teff) on the other. The value of Lrms based on Teff is found to be more robust. [Work sponsored by BOEM.]

3:15

2pUWb7. *In situ* demonstrator of a method to track in three dimensions cetaceans with passive acoustics in the context of close interactions impacts with marine renewable energy devices. Julie Lossent (Inst. of Eng. Univ. Grenoble Alpes, CNRS, Grenoble INP, GIPSA-Lab,, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Bouzidi Medjber (SINAY, Caen, France), Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Achraf Drira, Yanis Souami (SINAY, Caen, France), and Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France)

Collision risk and close interactions of cetacean with marine renewable energy (MRE) devices are among the main concerns and the least documented of the list of potential impacts of MRE. In this study, we developed a method to track in three dimensions click trains emitted by cetaceans in a close vicinity to MRE devices. We deployed two synchronized directional recording arrays 150 m apart from each other near Cherbourg (France). With an acoustic tag VEMCO© placed at several depths we performed transects with several trajectories simulating the potential behaviors (diving, surfacing, avoidance, and attraction) and click production of *delphinidae*. Thanks to the directional recording arrays and by crossing the direction of arrival of the same sounds calculated on each array, we build 3D maps of the synthetic clicks. We manage to reconstruct the trajectories (absolute positions x, y, and z in meters) of the emitter. In the areas located 100 m upstream and downstream of the two recording arrays, the location accuracy is maximum. This method could also be used to prove the departure and the non-return of cetacean from an exclusion zone. Moreover, this approach has been designed to be operational for the monitoring of cetaceans in environmental impact assessment studies.

3:30

2pUWb8. Controlled source level measurements of whale watch boats and other small vessels. Jennifer L. Wladichuk, David E. Hannay, Zizheng Li, Alexander O. MacGillivray (JASCO Appl. Sci., 2305 – 4464 Markham St., Victoria, BC V8Z 7X8, Canada, jennifer.wladichuk@jasco.com), and Sheila Thornton (Sci. Branch, Fisheries and Oceans Canada, Vancouver, BC, Canada)

The Vancouver Fraser Port Authority's Enhancing Cetacean Habitat and Observation (ECHO) program sponsored deployment of two autonomous marine acoustic recorders (AMAR) in Haro Strait (BC), from July to October 2017, to measure sound levels produced by commercial vessels transiting the strait. Fisheries and Oceans Canada (DFO), a partner in ECHO, supported an additional study using these same recorders to systematically measure underwater noise emissions (0.01-64 kHz) of whale watch boats and other small vessels that operate near Southern Resident Killer Whale (SRKW) summer feeding habitat. During this period, 20 different small vessels were measured operating at a range of speeds (nominally 5 knots, 9 knots, and cruising speed). The measured vessels were categorized into six different types based primarily on hull shape: ridged-hull inflatable boats (RHIBs), monohulls, catamarans, sail boats, landing craft, and one small boat (9.9 horsepower outboard). Acoustic data were analyzed using JASCO's PortListen® software system, which automatically calculates source levels from calibrated hydrophone data and vessel position logs, according to the ANSI S12.64-2009 standard for ship noise measurements. We found clear positive correlations of source levels with speed for all of these vessels; however, the speed trends were not as strong as those of large commercial vessels.

3:45

2pUWb9. Characterization and modeling of typhoon-generated noise in South China Sea. Jingyan Wang and Fenghua Li (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, 100190 Beijing, China, wangjingyan@mail.ioa.ac.cn)

Ocean noise recorded during a typhoon is dominated by wind-generated noise sources, and can be used to monitor the typhoon and investigate the mechanism of the wind-generated noise. An analytical expression for the typhoon-generated noise intensity is derived as a function of wind speed. A "bi-peak" structure was observed in an experiment in South China Sea during which typhoon-generated noise was recorded. Strong wind speed dependence and frequency dependence were also observed in the frequency range of hundreds to thousands of Hertz. The model/data comparison shows that results of the present model are in reasonable agreement with the experimental data, and the typhoon-generated noise intensity has a dependence on frequency and a power-law dependence on wind speed.

Session 2pUWc

Underwater Acoustics: Effects Due to Elasticity and Interfaces

Ahmad T. Abawi, Chair
HLS Research, 3366 North Torrey Pines Court, Suite 310, La Jolla, CA 92037

Contributed Papers

2:30

2pUWc1. Dynamic permeability and tortuosity in weakly consolidated granular media. Peyman M. Moradi and Apostolos Kantzas (Chemical and Petroleum Eng. Dept., Univ. of Calgary, ES903-2500 University Dr. NW, Calgary, AB T2N1N4, Canada, seyedpeyman.mohammad@ucalgary.ca)

Micro-scale rock heterogeneity plays an important role in modifying the seismic response of porous media and must be considered in reservoir studies such as seismic monitoring of fluid flow or interpretation of Stoneley waves. This work focuses on the prediction of dynamic permeability and dynamic tortuosity of granular media at different frequencies from porosity and average grain size data, utilizing numerical simulations. The simulations are conducted through cubic consolidated and unconsolidated digital rock samples of three different rock types, by applying a macroscopic oscillatory pressure gradient to each porous medium and combining fluid flow with solid mechanics to solve the wave propagation problem. The JKD parameter called dynamically connected pore size (Λ) is then predicted for each rock type. We investigate how pore space geometry, cement content and morphology, and matrix heterogeneity can influence the dynamic and geometrical tortuosities.

2:45

2pUWc2. Scattering from multiple elastic targets using the coupled finite element/boundary element method. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com), Ivars P. Kirsteins (NUWC, Newport, RI), Philip L. Marston, and Timothy D. Daniel (Washington State Univ., Pullman, WA)

The fluid-structure interaction technique provides a paradigm for solving scattering from elastic targets embedded in a fluid by a combination of finite and boundary element methods. In this technique, the finite element method is used to compute the target's elastic response and the boundary element method with the appropriate Green's function is used to compute the field in the exterior medium. The two methods are coupled at the surface of the target by imposing the continuity of pressure and normal displacement. This results in a self-consistent boundary element method that can be used to compute the scattered field anywhere in the surrounding environment. The method reduces a finite element problem to a boundary element one with drastic reduction in the number of unknowns, which translates to a significant reduction in numerical cost. In this talk, the method is extended to compute scattering from multiple targets by self-consistently accounting for all interactions between them. The model allows to identify block matrices responsible for the interaction between targets, which proves useful in many applications. The model is tested by comparing its results with those measured involving two aluminum cylinders one of which is excited by modulated radiation pressure.

3:00

2pUWc3. Seismic profile processing to identify sediment layering structure in shallow water. Wenbo Wang, Qunyan Ren, Jun Li, and Li Ma (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, renqunyan@mail.ioa.ac.cn)

The equivalent layering structure of surface sediment is a key parameter for sound propagation modelling and geoacoustic inversion in shallow waters. Previously, the structure identification is manually performed through identifying these horizontal lines as shown in the profiling image, which represent the interfaces with strong acoustical impedance contrast. In this paper, a hybrid image processing method (statistical equalization, multi-scale line filtering, and wavelet decomposition) is adapted to automatically identify these interfaces, afterwards, an image-binary threshold and a dislocation phase subtraction approach are successively applied to determine sediment layer number and relative thickness. The data from dozens lines of surveys as collected in 2016 are processed. Through integrating the processing results with GPS data, 2D distributions of sediment layer number and thickness of the first and second sediment layer are constructed. The spatial distributions of sediment layer number and thickness demonstrate certain trends, which are in accordance with expected sedimentary process in this region and should be carefully treated in 3D sound propagation modelling and geoacoustic inversion.

3:15

2pUWc4. Reflected seismic waves radiated by the apex of a wedge-shaped ocean. Wang Xiaohan, Shenchun Piao, Ya-Qin Liu, and Dong Yang (Haerbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, wangxiaohan@hrbeu.edu.cn)

A sound propagation experiment has been carried out near the coast in Qingdao to investigate underwater sound propagation characteristics over a sloping bottom. Hydrophones and Ocean Bottom Seismometers (OBSs) were arranged in different place to receive the signal radiated by explosive source. In the temporal signals received by the OBSs, some weak and mystical "tails" following the direction waves were observed but the hydrophones only received the direction waves without the "tails." Analyzing signal of different OBSs at different distance, it is indicated that the "tails" is the seismic waves coming from a fixed location. The temporal signal propagating in elastic wedge-shaped ocean is simulated with spectral-element method. The results of the model show that, when the acoustic waves propagates to the apex of the wedge, the boundary among the surface, the seabed and the coast, the apex will radiate the "tail" seismic waves. The reflection seismic waves can be apply in geoacoustic parameter estimation and Underwater Topographic Survey.

3:30

2pUWc5. An examination of the scaling rules for sound emissions from targets excited by point-like forces. Ivars P. Kirsteins (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, i.kirsteins@gmail.com), Ahmad T. Abawi (HLS Inc., La Jolla, CA), Philip L. Marston, and Timothy D. Daniel (Washington State Univ., Pullman, WA)

In earlier work [T. Daniel *et al*., JASA **140**, 3123 (2016)] we had showed experimentally that the modes of small elastic targets in water could be excited using modulated radiation pressure (MRP) generated by focused ultrasound to create detectable sound emissions. A potential advantage of the MRP approach is that the narrow beam of the ultrasound beam permits the generation of point-like forces on the target's surface with surgical precision. But an important question is how this technique scales with target size, e.g., the amount of force required to achieve a desired sound emission level. Here we examine how the sound emission levels and surface velocities of target modes driven by a point force scale with target size a. We will show that for a constant driving force, the far-field sound emission pressure levels scale as 1/a and the surface velocity as $1/a^2$. Specific examples are presented for circular plates, spheres, and cylindrical-like targets. Finally, results will be shown from a recent tank experiment at Washington State University where we attempted to experimentally determine the scaling rules using small targets excited by MRP.

3:45–4:00 Break

4:00

2pUWc6. The study on dispersion characteristics of Scholte wave in elastic media with variable parameters. Ya-Qin Liu, Hai-Gang Zhang, and Shi-E Yang (Harbin Eng. Univ., No.145 Nantong St., Nangang District, Heilongjiang, Harbin 150001, China, liuyaqin@hrbeu.edu.cn)

In the real ocean environment, due to the reflection of the water-sediment interface, the elastic sediment will generate inhomogeneous compressional wave and inhomogeneous shear wave, and the interaction between them will form the Scholte wave. Besides, the compressional and shear wave velocities in an elastic sediment layer are changing with depth leading to the coupling of compressional and shear waves. As the dispersion characteristics of the Scholte wave is affected by the coupling effect, in this paper, a study on the dispersion characteristics of Scholte wave in an environment with variable parameters elastic sediment layer is given. Specific environmental examples are simulated and the dispersion curves are compared with the KRAKENC program. In addition, the influence of the parameters of elastic sediment layer on the Scholte wave dispersion curves is analyzed. Comparing the dispersion curve of the Scholte wave in the uniform sediment layer and in elastic medium with variable parameters, it is found that some characteristics of the dispersion curve in the uniform layer will no longer fit the elastic layer with variable parameters.

4:15

2pUWc7. Broadband numerical simulation of scattering from rough pressure-release and fluid-fluid interfaces. Derek R. Olson (Oceanogr., Naval Postgrad. School, 201 Appl. Sci. Bldg., University Park, Pennsylvania 16802, olson.derek.r@gmail.com) and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

The scattered field from the seafloor is often measured using short, broadband pulses, whereas many models for the mean scattered intensity are in the frequency domain. This As higher resolution seafloor mapping systems, i.e., synthetic aperture sonar, become more common, it is important to understand the effects of high resolution on measurements of both the averaged scattered intensity, and the distribution of the envelope of the scattered field. To address this problem, Monte Carlo simulations of the scattered field due to rough interfaces separating two fluids are perform. The integral equation governing the total pressure, the Helmholtz-Kirchhoff integral equation, is numerically solved using the boundary element method. Simulations are performed both in the limit of pressure release, as well as for more realistic sediment sound speed and density parameters. The von-Karman spectrum is used to generate random rough surfaces. Fourier synthesis is used to to generate time domain signals, which are then analyized in terms of the mean energy and amplitude distribution. The dependence of the scattering cross section and scintillation index on signal bandwidth (resolution) is examined. It is found that there is no observable dependence of the scattering cross section on bandwidth, but that there is significant dependence of the scintillation index.

4:30

2pUWc8. Multiple scattering effects for a solid cylinder with axis oblique relative to a nearby horizontal surface. Aubrey L. Espana, Daniel Plotnick, Kevin Williams (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu), and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA)

It has been shown that when studying the acoustic scattering from a cylinder near a flat interface, there exists multiple paths by which sound can travel to, and subsequently scatter from, the cylinder. Specifically, sound can couple into a number of surface elastic waves (SEW), some of which give rise to an observed enhancement to the backscattering. Previously, the coupling conditions for these mechanisms have been derived, which are dependent on the relative angles between the target, the source/receiver and the interface [Plotnick, *et al*., JASA 140(3), 2016]. Here, a non-singular Helmholtz Kirchhoff Integral method has been implemented in a hybrid finite element/propagation model in order to predict the first and second order scattering mechanisms that exist when the cylinder is near an air-water interface, with its axis oblique with respect to the interface. This model, combined with the coupling conditions for SEWs, is used to dissect tank measurements that previously were only partially understood. Various multiple scattering mechanisms are identified and discussed. [Work supported by the Office of Naval Research.]

4:45

2pUWc9. Analysis on acoustic longitudinal horizontal correlation of air-to-water transmission. Lingshan Zhang, ZhaoHui Peng, and GuangXu Wang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhanglingshan@mail.ioa.ac.cn)

Spatial correlation is an important characteristic of underwater sound field, which is relevant to the detection of aerial targets from underwater. But compared with waterborne sources there are few researches about airborne sources. In order to perform further analysis on the air-to-water sound transmission in shallow water, State Key Laboratory of Acoustics, Institute of Acoustics, conducted an experiment in the South China Sea in March, 2013. During the experiment, multi-frequency signals transmitted by a hooter hung on a research ship were received by a horizontal line array (HLA) which laid on the seabottom, and signals far from 9.8 km were received successfully. Based on data recorded in the experiment, the longitudinal horizontal correlation coefficients of underwater sound field excited by airborne source at different distances with the frequency is 128 Hz are obtained, which have evident oscillation structure as horizontal spacing increases. The mathematical expression is derived based on the normal mode theory and gives a better physical explanation for the oscillatory behavior of the experimental longitudinal horizontal correlation.

5:00

2pUWc10. The treatment of the density discontinuity in the split-step Fourier algorithm using the hybrid approach. Mustafa Aslan (Turkish Naval Acad., National Defense Univ., 833 Dyer Rd., Monterey, CA 93943-5216, maslan1@nps.edu)

Various numerical models generate the solution for the modelling of underwater acoustic wave propagation treat the density discontinuity due to the density change at the interface by different approaches. In the split-step Fourier parabolic equation model, the smoothing function can be applied for the treatment of the density discontinuity, but the phase error is also accumulated with range in the solution. Yevick and Thomson developed the hybrid split-step/finite difference method in which density dependent terms are introduced in the additional operator to be applied by Padé approximation in the algorithm and the phase accuracy is significantly improved using the method. In this work, instead of Padé approximation, various approximations is examined for the density related operator in the split-step Fourier algorithm.

5:15

2pUWc11. Three dimensional finite element modeling of scattering by objects at rough interfaces. Aaron M. Gunderson and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78758, aaron.gunderson01@gmail.com)

Fully three-dimensional finite element models for buried objects and objects at interfaces have been extended to incorporate interface roughness. Near field results are solved using a fully scattered field formulation, in which all fields in the absence of the target are applied and the target scattering is solved. Results are extended to the far field through the Helmholtz-Kirchhoff integral, using a numerically determined Green's function approach. Interface roughness is applied using a modified power law wave-number spectrum. Previous model demonstrations evaluated the scattering in the flat interface limit, in which background fields were prescribed analytically. Now, the presence of the surface roughness forces background fields to be evaluated and prescribed numerically. Such models have strong applicability for predicting scattering by objects buried within or resting on a rough seafloor interface. Model results are compared with various experimental and other modeled results, demonstrating model validity over a wide frequency range. [Work supported by Applied Research Laboratories IR&D and ONR, Ocean Acoustics.]

OPEN MEETINGS OF TECHNICAL COMMITTEES/SPECIALTY GROUPS

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, 6 November

Committee	**Start Time**	**Room**
Engineering Acoustics	4:30 p.m.	Rattenbury A/B (FE)
Acoustical Oceanography	7:30 p.m.	Esquimalt (VCC)
Animal Bioacoustics	7:30 p.m.	Oak Bay 1/2 (VCC)
Architectural Acoustics	7:30 p.m.	Theater (VCC)
Physical Acoustics	7:30 p.m.	Colwood 1/2 (VCC)
Psychological and Physiological Acoustics	7:30 p.m.	Salon B (VCC)
Speech Communication	7:30 p.m.	Salon A (VCC)
Structural Acoustics and Vibration	7:30 p.m.	Saanich 1/2 (VCC)

Committees meeting on Wednesday, 7 November

Committee	**Start Time**	**Room**
Biomedical Acoustics	7:30 p.m.	(FE)
Signal Processing in Acoustics	7:30 p.m.	(FE)

Committees meeting on Thursday, 8 November

Committee	**Start Time**	**Room**
Computational Acoustics	4:30 p.m.	Esquimalt (VCC)
Musical Acoustics	7:30 p.m.	Crystal Ballroom (FE)
Noise	7:30 p.m.	Shaughnessy (FE)
Underwater Acoustics	7:30 p.m.	Rattenbury A/B (FE)