Session 1aAA

Architectural Acoustics: Sustainable Acoustics in Social Space and WELL Buildings

Siu Kit Lau, Cochair
Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Drive, Singapore 117566, Singapore

Andy Chung, Cochair
HKUST, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong

Ethan Bourdeau, Cochair
Standard Development, International WELL Building Institute, 381 Park Avenue South, Suite 1101, New York, NY 10016

Chair's Introduction—8:00

Invited Papers

8:05

1aAA1. Acoustical safety as important as comfort and wellness. Fumiaki Satoh (Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino, Chiba 275-0016, Japan, fumiaki.satoh@it-chiba.ac.jp)

Needless to say, the points of comfort and wellness are very important for the evaluation of spaces. In addition, how about the point of safety? This doesn't mean structural safety of buildings or safety to prevent crimes. For example, the luck of intelligibility of evacuation announcement in an emergency, this is an unsafe situation. It's a matter of "acoustical safety." To use absorption materials in spaces is very effective on the point of comfort and it leads to be better condition on the point of safety at the same time; however, the point of "acoustical safety" should be emphasized more. In the current situation in Japan, the consideration of using absorption materials for comfortable and safe spaces doesn't seem to be enough. This is a common understanding in our Japanese researchers on architectural acoustics. Therefore, a committee in Architectural Institute of Japan published a book in order to be popular using absorption materials. In this presentation, the contents about the book and the discussions up to publication will be introduced, in addition to some works on acoustical safety.

8:25

1aAA2. Noise analysis of urban change in the structure of health care up to the present from the beginning of the Republican era in Turkey: Case studies from Ankara Numune Hospital. Filiz B. Kocyigit, Cagri Bulhaz, and Cilga Resuloglu (İnterior Architecture, Atilim Univ., incek, Ankara 06560, Turkey, filizbk@gmail.com)

This research investigates the acoustical characteristics of hospitals' building environment of the Republican period and the contemporary era. Ankara Numune Hospital has been selected as a sample for research. The acoustical effect of the evolution of urban planning on the edges of Numune hospital in Capital city Ankara is analyzed. Green areas surrounded the boundaries of Numune hospital in the 1930s, whereas nowadays these green areas have been narrowed during decades. Social, political, and economic relationships changed as a result of urbanization in Turkey. Urban planning decisions affected the development, particularly the (historic) city center of Ankara where Numune hospital, the oldest hospital in Ankara, is located. Results, at the research, include equivalent sound pressure levels (Leq) as a function of location, frequency, and time of day. It appears that there are important problems in measurements taken at the entrance and at the hospital periphery and at different days and times. Average equivalent sound levels are in the 50–60 dB (A) range for 1 min, $^1/_2$, and 24 h averaging time periods. 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.

8:45

1aAA3. Acoustics comfort in buildings in Hong Kong, Macao, and the Greater Bay Area of China. W. M. To (Macao Polytechnic Inst., Macao, Macao) and Andy Chung (Macau Instituto de Acustica, Macau, Macau SAR 853, China, ac@smartcitymaker.com)

There have been various assessment standards on green and sustainable buildings to promote the better use of resources following a holistic approach. As one of the major sources of complaints, noise is an essential element in these schemes. More and more smart use of materials and design embedded is seen. This paper reviews the role of sound and acoustics in the whole life cycle for a building—from planning, design to construction, and operation. A review on the latest trends and way forward will also be given.

9:05

1aAA4. Introduction to and implementation of the WELL sound concept. Ethan Bourdeau (Standard Development, Int., WELL Bldg. Inst., 381 Park Ave. South, Ste. 1101, New York, NY 10016, epbourdeau@gmail.com)

The latest revision of the WELL rating system relocates all previous acoustic related features from the Comfort concept into the new Sound concept. The new iteration of WELL reassesses the importance of all features in terms of substantiation, evidence, best practice, and feasibility, most notably for implementation within existing spaces where minimal mitigation can be anticipated. This paper introduces the new feature set of the Sound concept with a brief introduction of the new WELL rating system platform, provides an example of implementation through a case study of the new International WELL Building Institute (IWBI) headquarters, and serves as a continuation to the discussion of acoustics in ratings systems that address IEQ, health, and well-being.

9:25

1aAA5. Using the new ASHRAE 189.1 Standard for the Design of High-Performance Green Buildings—Acoustical Control Section. Erik Miller-Klein (A3 Acoust., LLP, 241 South Lander St., Ste. 200, Seattle, WA 98134, erik@a3acoustics.com) and Jeff Boldt (IMEG Corp, Madison, WI)

The 2017 ASHRAE 189.1 Standard for the Design of High-Performance Green Buildings includes a complete rewrite of the acoustical control section. This section of the standard provides both performance and prescriptive paths to acoustical compliance and permits either acoustic commissioning or visual inspections to ensure the materials and assemblies are built to optimize acoustical performance. We will outline the importance of this new standard and review the basic parameters of the acoustical performance requirements, and how this standard can be used for your projects. This session will discuss the basis for this updated section and how you can use this standard to help guide your clients, and satisfy the LEED v4 for BD+C. The 189.1/IgCC aligns well with LEED, WELL, and other green building and improves on the LEED certification standard, so basic compliance will often garner the acoustic performance points.

Contributed Paper

9:45

1aAA6. Evaluating acoustical performance of existing offices as part of a WELL feasibility service. Ryan Bessey (RWDI, 901 King St. West, Ste. 400, Toronto, ON M5V 3H5, Canada, ryan.bessey@rwdi.com) and Jessie Roy (RWDI, Calgary, AB, Canada)

While the WELL building standard is typically applied during the design of new offices, it can also be applied retroactively to existing offices. Before undertaking such an endeavour, some companies are interested in evaluating performance with respect to the standard. This paper describes how a WELL feasibility assessment can be performed for the acoustically focused WELL features. This allows companies to know where their existing space stands with respect to the WELL building standard and what it might take to achieve the different levels of certification.

10:00–10:15 Break

Invited Papers

10:15

1aAA7. Case study and lessons learned through achieving WELL Platinum. Ken Shook (Acoust., Longman Lindsey, 200 West 41st St., Ste. 1100, New York, NY 10018, kens@longmanlindsey.com)

This paper will present a case study of one of the first projects to achieve a Platinum rating through the WELL Certification program. We will review the acoustic requirement outlined by WELL and issues faced during the design of a WELL Platinum project. These will include conflicting features that were discovered between multiple features, the limitations that base-building can present, conflicts between WELL and LEED and how to satisfy both requirements. We well summarize our lessons learned and how we have applied these into recommendations to the WELL acoustic committee through our advisory role.

10:35

1aAA8. Soundscape evaluation of urban social spaces. A comparative study: Berlin-Granada. Antonella Radicchi (Institut für Stadt- und Regionalplanung, Technische Universität Berlin, Hardenbergstraße 40 a, Sekr. B 4, Berlin, Berlin 10623, Germany, antonella.radicchi@tu-berlin.de) and Jeronimo Vida (Dept. of Appl. Phys., Univ. of Granada, Granada, Granada, Spain)

To address the design and development of healthy built environments, noise pollution has to be taken into account, especially in Europe, where over 125 million people are affected by noise pollution from traffic every year. Since the release of the European Environmental Noise Directive in 2002, there has been a growing interest in protecting and planning quiet areas as a valid tool to reduce noise pollution. However, a common methodology to properly achieve this goal is still missing. This contribution tackles this challenge, by illustrating the implementation in Berlin and Granada of a novel participatory methodology grounded on the soundscape approach, the citizen science paradigm and a novel mobile application—the Hush City app. First, the theoretical background and methods applied are described; second, preliminary findings of the comparative assessment of "everyday quiet areas" in Berlin and Granada are illustrated. In conclusion, recommendations are provided to integrate this methodology in advanced urban planning.

Contributed Papers

10:55

1aAA9. Two case studies of acoustical design in new construction using sustainable criteria: The living building challenge and WELL building design at The Georgia Institute of Technology. Jessica S. Clements (Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

The Georgia Institute of Technology is updating their campus through a series of sustainably designed buildings to create an educational eco-commons. The ongoing renewal includes the Kendeda Building for Innovated Design and the Dalney Building. These buildings are designed to meet the International Living Future Institute's Living Building Challenge Design Standard and the International Well Building Institute's WELL certification criteria, respectively. The Kendeda building consists of classrooms, commons, offices, an auditorium, and open collaborative areas. The Dalney building is primarily office and conference spaces but includes some unique challenges for adaptable open and enclosed office areas. Substantial construction for the Kendeda building is scheduled for spring of 2019 and for the Dalney building in fall of 2019. The case studies cover the design phases of the two projects and discuss the acoustical challenges in meeting these specific designs, including credit compliance, unforeseen ramifications across disciplines, and areas of problematic interpretation.

11:10

1aAA10. Just noticeable difference of masker to enhance privacy in an open-plan office. Ainun Nadiroh and Dhany Arifianto (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, ainun12@ep.its.ac.id)

If a background conversation (masker) is easy to follow, then a worker at a workstation in an open-plan office will be distracted and annoyed. The aim of this study was if the mixture of the background conversations statistical distribution close to Gaussianity, then regardless the loudness level, the intelligibility will be decreased. On the other hand, the privacy at the workstation will be increased due to the loudness level of the background conversation. To assess the privacy level in a simulated open-plan office, we propose Just Noticeable Difference (JND) of the masker. The measurement was conducted in two workstations laboratory with 64 square meters for each workstation in which three conditions (male-female, all male, and all female speakers) of babble noise was built in one of them. The other workstation was simulated with single speech sound to observe speech privacy in the presence of the masker. We used objective measures to assess the intelligibility and the privacy of the workstation. The results suggest that the saliency of the masker depends on the fundamental frequency difference (dominant speaker). The higher saliency of the masker will cause the lower of the privacy of the other workstation.

11:25

1aAA11. Continued Summer Sound Lab Academic Research: Developing thin shell precast systems for acoustical environments. Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

This paper presentation continues progress of an active multiphase acoustical research project emphasizing implementation of adaptive interactive acoustical responses in multiuse spaces to transform unhealthy learning environments. Building upon the December 2017 paper presentation, slotted and pocketed thin shell precast systems are being explored as a viable option to decrease excessive reverberation and flutter echo while increasing clarity and speech intelligibility within a multiuse space on campus. The project, launched Spring 2017, is spanning multiple phases and disciplines as faculty and students are analyzing and validating findings in both computer and physical form. Collected data will lead researchers to design applications for adaptive and resilient learning environments with minimized acoustical distractions and improved speech intelligibility. The work presented here defines prototypical concepts to inform final full-scale built form, providing scholarly merit applicable to similar spaces and/or functions.

11:40–12:00 Panel Discussion

Session 1aAB

Animal Bioacoustics: Fish and Marine Invertebrate Bioacoustics I

Bruce Martin, Cochair
JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada

Xavier Mouy, Cochair
JASCO Applied Sciences, 2305–4464 Markham Street, Victoria, BC V8Z7X8, Canada

Chair's Introduction—8:45

Invited Papers

8:50

1aAB1. Assessing vessel slowdown as an option for reducing acoustic masking for Arctic cod in the western Canadian Arctic. Matthew Pine (Dept. of Biology, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, matt.k.pine@gmail.com), David E. Hannay (Jasco Appl. Sci., Victoria, BC, Canada), Stephen J. Insley (Wildlife Conservation Society Canada, Whitehorse, Br. Columbia, Canada), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), and Francis Juanes (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada)

Noise from this shipping traffic can lead to acoustic masking, reducing the ability of marine animals to detect and use biologically important sounds. Vessel-slow down may be an alternative mitigation option in regions where re-routing shipping corridors to avoid important habitat for fish and marine mammals is not possible. We investigated the potential relief in masking from a 10 knot speed reduction of container and cruise ships. Based on ambient sound measurements and real shipping data, the percentage reduction in the available listening space for fish as a container or cruise ship passes under varying speeds and ambient sound conditions was shown. The mitigation effects from slower vessels (travelling at 15 knots compared to 25 knots), in terms of auditory masking, was equal between ambient sound conditions, but not equal between the type of vessel. Slowing vessels led to a substantial decrease in the listening space reductions, with the amount of reduction varying by distance away from vessels. Vessel slowdown through sensitive habitat could be an effective mitigation strategy for reducing the extent of auditory masking.

9:10

1aAB2. Auditory threshold in fishes: Towards international standard measurement procedures. Michele B. Halvorsen (CSA Ocean Sci. Inc., 8502 SW Kansas Hwy, Stuart, FL 34997, mhalvorsen@conshelf.com) and Michael A. Ainslie (JASCO Appl. Sci., Eschborn, Germany)

Fish rely on their auditory system for survival to detect and interpret sounds in their surroundings, especially those originating from predators, prey, and conspecifics. The sensitivity of fish hearing can be impaired, for example, by masking or temporary threshold shifts. To understand and quantify the impact of noise in the oceans, we need a quantitative and comparable understanding of fishes' impaired and unimpaired hearing sensitivities to sound pressure and sound particle motion. Historically, hearing sensitivities were measured to generate an audiogram for a fish species of interest. Therefore, audiograms are only available for a few individuals representing a few species. Crucially, differences between measurement protocols leave existing audiograms mostly incomparable. The goldfish (*Carassius auratus*) is a species for which many audiograms have been measured and the reported hearing thresholds exceed a 40 dB spread, largely attributed to differences in methodology rather than in hearing sensitivity [Ladich & Fay, 2013; Maruska & Sisneros, 2016]. These large differences indicate a need to harmonize measurement procedures. Standardization of hearing sensitivity measurements is therefore essential if we are to achieve environmental management goals in the United States, Europe, and worldwide. We propose a path that leads to the development of suitable international standards.

Contributed Papers

9:30

1aAB3. Listening in the dark—Auditory evoked potential measurements and novel imaging techniques elucidate the role of hearing in deep sea fishes. Klaus Lucke (Australia, JASCO Appl. Sci., 1/14 Hook St., Capalaba, QLD 4157, Australia, klaus.lucke@jasco.com), Shaun P. Collin, Caroline Kerr, Victoria Camilieri-Asch (Oceans Graduate School and the Oceans Inst., The Univ. of Western Australia, Crawley, WA, Australia), Joanna Krieg (School of Biological Sci. and the Oceans Inst., The Univ. of Western Australia, Crawley, WA, Australia), Robert McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, WA, Australia), and Lucille Chapuis (Oceans Graduate School and the Oceans Inst., The Univ. of Western Australia, Crawley, WA, Australia)

In the aphotic zone of the deep ocean, mechanisms for conveying information to conspecifics or gathering information about the environment and potential predators are subject to the same physical and ecological limitations as near to the sea surface with the addition of no available light. Passive or active communication often utilises bioluminescent visual, olfactory, or acoustic signals. The sensitivity of deep sea fish to sound has never been tested. In a multi-disciplinary study, deep sea fishes were sampled in the Indian Ocean and examined for acoustic sensitivity. Acoustic signals were generated in a tubular steel tank using a calibrated sound source with pressure and particle motion of generated signals measured. The auditory sensitivity was tested using the auditory brainstem responses to amplitude modulated stimuli. To determine if the pressure or particle motion component of the acoustic signals was most relevant, specimens of all species tested were imaged in a micro-CT using a modified contrast enhancing method. This method provides increased image quality of *in-situ* morphological features, including the swim bladder and ancillary structures possibly related to for sound perception. Correlating these physiological and morphological results provides the key for an ecologically meaningful assessment of hearing capabilities of deep sea fishes.

9:45

1aAB4. Assessing ontogenetic change in acoustic pressure sensitivity in larval fishes through modelling. Andria K. Salas (Integrative Biology, Univ. of Texas at Austin, 205 W. 24th St. A6700, Austin, TX 78712, aksalas@utexas.edu), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Lee A. Fuiman (Marine Sci. Inst., Univ. of Texas at Austin, Port Aransas, TX)

Detecting acoustic pressure can improve a fish's survival and fitness through increased sensitivity to environmental sounds. Pressure detection results from the interactions between the swim bladder and otoliths. In larval fishes, those interactions change rapidly as growth and development alter bladder dimensions and otolith-bladder distances. We used computed tomography imagery of lab-reared larval red drum (*Sciaenops ocellatus*) in a finite-element model to assess ontogenetic change in acoustic pressure sensitivity in response to a plane wave at frequencies within the range of detection by fishes. We compared the acceleration at points on the sagitta, asteriscus, and lapillus when the bladder was air-filled to results from models using a water-filled bladder. For larvae of 8.5–18 mm, the air-filled bladder amplified otolith motion by a factor of 54–3485 times that of a water-filled bladder at 100 Hz. Otolith-bladder distances increased with standard length, which decreased amplification. The concomitant rapid increase in bladder volume partially compensated for the effect of increasing otolith-bladder distances. Using idealized geometry, we found that the backbone and ribs have a negligible influence on bladder motion. Our results help clarify the auditory consequences of ontogenetic changes in swim bladder morphology and otolith-bladder relationships during larval stages.

10:00–10:15 Break

10:15

1aAB5. Experimental investigation of a robust algorithm for unambiguous source directionalization by sharks. Peter H. Rogers, James S. Martin (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, peter.rogers@gatech.edu), John C. Montgomery (Inst. of Marine Sci., Univ. of Auckland, Auckland, New Zealand), and Craig A. Radford (Inst. of Marine Sci., Univ. of Auckland, Warkworth, New Zealand)

All fish can detect the three vector components of particle motion, enabling them to determine source bearing, but particle motion alone cannot resolve which direction is towards the source and which direction is away (*i.e.,* there is a 180° ambiguity). However, fish with swim bladders can also detect acoustic pressure which enables such fish to resolve this 180° ambiguity. Proposed processing algorithms, though, often employ unrealistic restrictions such as sinusoidal time dependence and free-field conditions. We propose a robust algorithm for determining the unambiguous source direction by a fish that can detect acoustic pressure. The method is based on the time-averaged acoustic intensity vector (the time-integral of the product of pressure and particle velocity). This non-oscillatory quantity points in the direction of energy flow and hence directly away from the source. The algorithm has the advantage of being applicable to signals with any time dependence, in any propagation environment. For sharks, which do not have swim bladders, we hypothesize that the vertical component of particle acceleration can be substituted for pressure, provided that the shark is sufficiently close to the water surface. We test the validity of this hypothesis by determining the direction to the source for reef noise, a biologically relevant signal, and boat noise in a range-varying shallow water environment.

10:30

1aAB6. Assessing impacts of offshore pile driving noise on the antipredator defense and shoaling behaviors of squid (*Doryteuthis pealeii*). Ian T. Jones (Biology, Woods Hole Oceanographic Institution/ Massachusetts Inst. of Technol., Woods Hole Oceanographic Inst., 266 Woods Hole Rd. MS #50, Woods Hole, MA 02543, ijones@whoi.edu), Jenni Stanley (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), and T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA)

Impulsive pile driving occurs during construction of marine platforms, producing intense sounds that may adversely impact animals' physiology and behavior. Little is known regarding how this noise impacts sound-sensitive invertebrates such as squid, which is surprising given squids' relative abundance and key ecological role. We quantified how a commercially important squid species (*Doryteuthis pealeii*) behaviorally responded (in a controlled environment) to pile driving sounds recorded from an offshore windfarm installation within this species' habitat. Both sound pressure and particle motion components of the sound were quantified. Fifteen-minute portions of the recordings were played to individual squid. Body pattern changes, inking, jetting, and startle responses were observed during sound exposure and all squid exhibited at least one response. These responses occurred primarily during the first few noise impulses and diminished quickly over the first minute of playback, indicating short-term habituation. Responses returned after a 24-hr rest, indicating re-sensitization. Separate experiments investigated changes in shoaling behaviors by quantifying shoal cohesion and polarity in groups of squid during ten-minute noise exposures. Rapid habituation of antipredator alarm responses and changes in shoaling dynamics may alter squids' susceptibility to predation. Noise exposure may also disrupt normal intraspecific communication and ecologically relevant behavioral responses to sounds.

10:45

1aAB7. Resilience exemplified-drum fish spawning doesn't miss a beat in the eye of a hurricane. Christopher Biggs and Brad Erisman (Marine Sci. Inst., Univ. of Texas, 750 Channelview Dr, Port Aransas, TX 78373, cbiggs@utexas.edu)

Spawning locations, timing, and periodicity are important aspects of productivity and resilience in fish because they are directly related to

reproductive success. Passive acoustic monitoring can be used to study these aspects of spawning for species that produce spawning associated vocalizations. Drumming sounds produced by spotted seatrout *Cynoscion nebulosus* during spawning have been extensively described and the sound pressure level (SPL) is correlated to the intensity of spawning. We monitored spawning in seatrout April-September 2017, which coincided with a category 4 hurricane and caused a major acute disturbance. Spawning sounds within the peak frequency bandwidth of seatrout chorusing (251-500 Hz) were observed on every day of the study with an average SPL of 121.8 (CI_{95} 121.6-122.0) dB_{rms} re: 1 μPa, which was significantly higher than the background noise level of 103.9 (CI_{95} 103.8-104.0) dB_{rms} re 1 μPa ($w = 0.894$, $p<0.01$). Spawning was also confirmed at two sites within the eye of the hurricane. The onset of spawning shifted 2.2 hours earlier ($t = 13.91$, df = 36, $p < 0.01$) for five days after the hurricane. These results illustrate that spotted seatrout are extremely resilient, which indicates a primary reason they are able to thrive in highly disturbed estuarine environments.

11:00

1aAB8. Discovery of sound in the sea: Webinars as a means of communicating current underwater acoustics research to decision makers. Kathleen J. Vigness-Raposa (Marine Acoust., Inc., 2 Corporate Pl., Ste. 105, Middletown, RI 02842, kathleen.vigness@marineacoustics.com), Gail Scowcroft, Christopher Knowlton, Holly Morin (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI), Darlene R. Ketten (Woods Hole Oceanographic Inst., Perth, Western Australia, Australia), Arthur N. Popper (Univ. of Maryland, College Park, MD), and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Underwater sound used for anthropogenic activities is reviewed and restricted under a variety of environmental regulations. Decision makers must often synthesize rapidly new scientific research results to inform their assessments of potential impacts of proposed projects. To assist this need, the University of Rhode Island Graduate School of Oceanography has teamed with Marine Acoustics, Inc., in the Discovery of Sound in the Sea (DOSITS) project to provide accurate scientific information on underwater sound through a diversity of resources and digital platforms, including webinars. Building on the foundation of the successful 2015-2016 DOSITS webinar series and informed by the results of three international regulatory community needs assessments, the DOSITS project is hosting throughout 2018 a four-part webinar series on the fundamentals of underwater hearing and potential impacts of underwater sound on marine animals, particularly marine mammals (April–May) and fishes (November). Evaluation results from the first two webinars on marine mammals showed that 90% of survey respondents were very satisfied or satisfied with the content coverage, and 97% were extremely or very likely to attend future DOSITS webinars. The webinar approach has provided much needed on-the-job training for decision makers to effectively incorporate new scientific research into their evaluation processes.

MONDAY MORNING, 5 NOVEMBER 2018 — ESQUIMALT (VCC), 8:00 A.M. TO 12:00 NOON

Session 1aAO

Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Arctic Acoustical Oceanography I

Peter F. Worcester, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225

Mohsen Badiey, Cochair
University of Delaware, Newark, DE 19716

Hanne Sagen, Cochair
Polar Acoustic and Oceanography Group, Nansen Environmental and Remote Sensing Center, Thormøhlensgt 47, Bergen 5006, Norway

Invited Paper

8:00

1aAO1. Toward predictive understanding of the rapidly evolving Arctic Ocean—An overview. Wieslaw Maslowski, Younjoo Lee, Jaclyn Kinney (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, maslowsk@nps.edu), Ronbert Osinski (Inst. of Oceanology Polish Acad. of Sci., Sopot, Poland), and Samy Kamal (Oceanogr., Naval Postgrad. School, Monterey, CA)

Some of the most rapid climate changes on the planet are experienced in the Arctic. In particular, the Arctic has been warming at a quicker pace than any other place on Earth, what is recognized as Arctic Amplification (AA). This warming has been most visibly manifested through a declining perennial sea ice cover, increasing the potential for its transition from the permanent toward a seasonal coverage. Those changes also affect air-sea heat fluxes and amplify ice-albedo feedback, which strongly influences ocean's absorption of solar radiation. In addition, they also alter the Arctic Ocean acoustical regime, as the thinning sea ice moves faster and deforms easier,

while its reduced coverage allows increased momentum transfer from the atmosphere to the upper ocean. This talk will provide an updated overview of the recent changes and trends in the Arctic Ocean of relevance to acoustical oceanography. We will focus on the evolution of the upper ocean stratification and water masses, mesoscale processes, and their linkages to the changing regime of the sea ice cover from multi-year to first-year sea ice. Also, the latest advancements and outstanding challenges in modeling and prediction of Arctic climate change at sub-seasonal to interannual time scales will be discussed.

Contributed Papers

9:00

1aAO2. The 2016–2017 deep-water Canada Basin Acoustic Propagation Experiment (CANAPE): An overview. Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), Andrey Y. Proshutinsky, Richard A. Krishfield (Dept. of Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA), Jonathan D. Nash (College of Earth, Atmospheric, and Ocean Sci., Oregon State Univ., Corvallis, OR), and John N. Kemp (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Arctic Ocean is undergoing dramatic changes in the ice cover and ocean structure. The 2016–2017 deep-water Canada Basin Acoustic Propagation Experiment (CANAPE) was designed to understand the effects of changing Arctic conditions on low-frequency, long-range propagation and ambient noise. Five acoustic transceivers were deployed in a pentagon with a sixth transceiver at the center, forming an ocean acoustic tomography array with a radius of 150 km in the central Beaufort Sea. The transceivers had broadband sources centered at approximately 250 Hz located at 175-m depth in the Beaufort Duct and 15 Hydrophone Modules spanning 135 m located above the sources. A Distributed Vertical Line Array receiver with 60 Hydrophone Modules spanning 540 m was embedded within the tomographic array to provide measurements of acoustic time fronts and their fluctuations. The tomographic array was largely in open water during summer, in the marginal ice zone as it transitioned across the array during the spring and autumn, and under complete ice cover during winter. The tomographic data, together with moored data from Sea-Bird MicroCATs, Acoustic Doppler Current Profilers, ice-profiling sonars, and precision temperature sensors, will help characterize the large-scale oceanographic variability throughout the year, aiding in the interpretation the acoustic data.

9:15

1aAO3. Low–frequency acoustic transmissions under sea ice as measured in the Beaufort Sea. Matthew Dzieciuch, Peter F. Worcester (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu), John A. Colosi (Naval Postgrad. School, Monterey, CA), Andrey Y. Proshutinsky, Richard A. Krishfield (WHOI, Woods Hole, MA), Jonathan D. Nash (Oregon State Univ., Corvallis, OR), and John N. Kemp (WHOI, Woods Hole, MA)

A tomography array was deployed in the Beaufort Sea for a year beginning in late summer 2016 during the Office of Naval Research sponsored project called the *Canada Basin Acoustic Propagation Experiment* (CANAPE.) This talk will look in detail at the propagation characteristics of broadband sound transmitted at 250 Hz over 108 km to 285 km ranges as measured on a 60 element vertical line array. Over the year, the heat content and vertical stratification of the ocean change as heat is gained and lost to the atmosphere and as the momentum transfer from the wind mixes the upper layers. The changes in heat content and stratification over the year affects the travel-time of the sound as well as its fluctuations. Examples will be shown of each. Sea-ice roughness adds scattering loss as the ice thickness increases. Thus, the seasonal ice-cover modulates the transmission loss of the acoustic paths. The losses on deep turning paths are different from the sound trapped in the surface duct. The ambient noise level also changes as the seasons progress.

9:30

1aAO4. An overview of Beaufort Sea eddies, internal waves, and spice from several recent field efforts and implications for acoustic propagation. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu), Murat Kucukosmanoglu (Ocean Sci., Univ. of California, Santa Cruz, Santa Cruz, CA), Peter F. Worcester, Matthew Dzieciuch (IGPP, Scripps Inst. of Oceanogr., La Jolla, CA), Andrey Y. Proshutinsky, Richard A. Krishfield (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA), and Jonathan D. Nash (Oceanogr., Oregon State Univ., Corvallis, OR)

Several recent field efforts have revealed surprisingly complex and dynamic thermohaline structure in the upper ocean of the Beaufort Sea. Solitary and compact eddies with strong temperature contrasts and currents have been observed in multiple locations and are associated with vigorous mixing, staircase structure and intrusive feature formation. While many of the eddies are primarily found in the upper 300-m of the water column, rare deep eddies with cores near 500 to 1000-m depth have also been observed. Internal waves are generally weak with energies an order of magnitude less than mid-latitude values and they show marked dominance by near inertial waves, intermittency, spatial inhomogeneity, and deviations from the Garrett-Munk model. Strong intrusive structure, termed spice, is observed in the upper 150-m of the water column and is associated with the mixed layer and eddy activity. Acoustic implications of the associated sound speed structure will be discussed.

9:45

1aAO5. Effects of range-dependent sound speed on acoustic seaglider arrivals in the Canada Basin. Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu), Sarah E. Webster (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)

Two seagliders equipped with temperature, conductivity and pressure sensors, acoustic recorders, and acoustic Doppler current profilers were deployed in the Canada Basin as part of a large scale acoustic tomography experiment in the summer of 2016 and 2017. The seagliders received transmissions from moored acoustic sources transmitting linear frequency modulated sweeps with 100 Hz bandwidth and center frequencies on the order of 250 Hz. Sources were moored at approximately 175 m depth within an acoustic duct (sometimes referred to at the Beaufort Duct), which is present in the Canada Basin due to Pacific Winter Water, enabling acoustic transmission to long ranges. Acoustic Seagliders recorded transmissions at ranges up to 480 km. The variability and stability of the acoustic duct with range was measured by temperature and salinity profiles recorded by the gliders as they transited between moored source locations. The range dependence of this duct will be explored through oceanographic measurements made in the summer of 2016 and 2017, with particular attention paid to glider transects which roughly overlapped between the two years. The effect of the range dependent sound-speed environment on the acoustic arrivals received on the gliders will be explored through parabolic equation models and received acoustic data.

10:00

1aAO6. Overview of one year acoustical oceanography measurements on Chukchi shelf during Shallow Water Canada Basin Acoustic Propagation Experiment 2016–2017. Mohsen Badiey, Justin Eickmeier (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Altan Turgut (Naval Res. Lab, Washington, DC), Sean Pecknold (DRDC Atlantic, Dartmouth, NS, Canada), Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs, Austin, TX), Peter F. Worcester, Mathew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA), and Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada)

A multi-institutional, acoustical oceanography experiment was conducted from October 2016 through November 2017 on the Chukchi continental shelf covering 100–700 m isobaths. Parallel to a deep-water experiment conducted during the same period, the Shallow Water Canada Basin Acoustic Propagation Experiment (SW CANAPE) was designed to assess basin scale acoustic signals on the shelf region while detailed oceanographic dynamic of the shelf break region, particularly the upwelling and other dynamic of the upper 500 m water column, was measured simultaneously. Multiple arrays of oceanographic sensors including upward looking ice profiler, current profiler, temperature, conductivity, and pressure profiles measured temporal and spatial dynamics of 500 m upper ocean in connection with acoustic measurements. Distributed in a 30 km^2 area north of Barrow Alaska, vertical line arrays including an L-shaped array, covered upper 200 m of the water column. Two acoustic sources placed at 148 m and 193 m depths on the shelf emitted broadband acoustic signals in frequency bands (700–1100 Hz, and 1400–4000 Hz) along and across the shelf while the sound speed and current profile and surface ice were being measured continuously. Deep water low frequency signals were also recorded. This talk provides an overview of the SW CANAPE experiment and highlights some of the detailed measurements. [Work supported by ONR.]

10:15–10:30 Break

10:30

1aAO7. Arctic Beaufort Gyre duct transmission measurements and simulations. Timothy F. Duda, Ying-Tsong Lin, Weifeng G. Zhang (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), John A. Colosi (Naval Postgrad. School, Monterey, CA), and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Transmissions of 100–300 Hz sound over distances of 220–505 km in the Beaufort Sea area for 10 months revealed strong long-term trends and short-term variability. The sound was emitted by Canada Basin Acoustic Propagation Experiment (CANAPE) Deep Water sources, and received by CANAPE Shallow Water receivers close to the Chukchi Sea. Much of the sound that arrives at these distances is trapped in the Pacific Water duct. A dynamical model of the area driven by representative forcing, which includes eddies that have propagation consequences, is used to examine changes to the water column at small eddy scales and at broader scales that can influence the sound. Processes such as variable excitation of ducted normal modes, time-variable duct sound-speed mean profile, and coupling of ducted modes to other modes by range dependent eddy features are examined with the model. The effects of these processes are quantified and compared to field observations.

10:45

1aAO8. A year-long record of low-frequency, long-range, ducted sound propagation from the Canada Basin to the Chukchi Shelf. Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), Altan Turgut (U.S. Naval Res. Lab., Washington, DC), Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada), Ying-Tsong Lin, Andrey Y. Proshutinsky, and Richard A. Krishfield (Woods Hole Oceanographic Inst., Woods Hole, MA)

The Pacific Arctic Region, has experienced decadal changes in atmospheric conditions, seasonal sea-ice coverage, and seawater temperature. From the October 2016 to September 2017, the Canada Basin Acoustic Propagation Experiment (CANAPE) was conducted to understand the changing soundscape and to explore the use of acoustic signals as a remote sensing tool in the modern Arctic. During the experiment, low-frequency signals from five tomographic sources located in the Canada Basin were recorded by an array of hydrophones with both horizontal and vertical apertures located on the Chukchi Shelf at the 150 m isobath. The propagation distances ranged from 240 km to 520 km, and the propagation conditions changed from persistently ducted in the basin to seasonally upward refracting on the continental shelf. The water column properties and ice draft were measured by oceanographic sensors on the basin tomography moorings and by an array of oceanographic moorings on the continental shelf and slope to characterize the temporal and spatial variability of the environment. This talk examines the range-dependent measurements and explains the observed variability in the received signals through propagation modeling using the oceanographic measurements. [Work sponsored by ONR.]

11:00

1aAO9. Seabed properties at the 150 m isobath as observed during the 2016–2017 Canada Basin Acoustic Propagation Experiment. Jason D. Sagers and Megan S. Ballard (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

Seabed layering and sediment properties impact sound propagation in ocean waveguides, particularly in environments where sound propagation paths repeatedly interact with the seafloor. As part of the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE), experiments were designed to investigate seabed layering and sediment properties on the Chukchi Shelf. First, the shallow water experimental region was surveyed with a subbottom profiler to provide information about the overall sediment layering. Second, ship-radiated noise from a research vessel sailing specifically designed tracks was received on the Persistent Acoustic Observation System (PECOS). These recordings provide an opportunity for short- to mid-range geoacoustic inversion for sediment properties. Third, *in-situ* acoustic sound speed measurements were made with the Acoustic Coring System (ACS) while two to five meter long core samples were simultaneously collected. This talk presents initial findings of the seabed layering and sediment properties from these three experiments. [Work sponsored by ONR.]

11:15

1aAO10. The effects of ice cover and oceanography on medium-frequency acoustic propagation on the Chukchi Shelf. Sean Pecknold (DRDC Atlantic Res. Ctr., PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca), Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada), Ildar R. Urazghildiiev (JASCO Appl. Sci., Ithaca, New York, NY), Michelle Weirathmueller (JASCO Appl. Sci., Seattle, Washington), Mohsen Badiey (Univ. of Delaware, Newark, DE), Altan Turgut (Naval Res. Lab., Washington, DC), Jason D. Sagers, and Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

The Canada Basin Acoustic Propagation Experiment (CANAPE) was a year-long experiment exploring the changing nature of sound propagation and ambient noise in the Arctic ocean. As part of this experiment, medium-frequency signals at 0.7–1 kHz and 1–4 kHz were transmitted by two sources on the Chukchi Shelf. One of these sources was located in an area of 150 m of water depth, approximately 350 m from a directional receiver array and 50 km from an 8-element vertical line array in a water depth of about 125 m. Oceanographic sensors were located both on the arrays and on a set of moorings on the shelf, and an ice-profiling sonar was located between the arrays about 15 km from the source. In this talk, we will focus on using the measured environmental data and propagation modeling to characterize the variability observed in the short-range and long-range received acoustic signals over the course of CANAPE.

11:30

1aAO11. Measured acoustic intensity fluctuations on the Chukchi continental shelf during Canada Basin Acoustic Propagation Experiment 2017. Mohsen Badiey, Justin Eickmeier, Lin Wan (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Altan Turgut (Naval Res. Lab, Washington, DC), Sean Pecknold (DRDC Atlantic Res. Centere, Dartmouth, NS, Canada), and Megan S. Ballard (Appl. Res. Labs., Austin, TX)

One of the main objectives of the Shallow Water (SW) CANAPE experiment was to gain a thorough understanding of a yearlong propagation of broadband signals from deep to shallow water with simultaneous oceanographic and acoustic measurements together along the and across the shelf break region. Using more than eleven acoustic arrays and seven oceanographic moorings in a 30 km^2 region on the Chukchi shelf this task is being done by assessing both deep water sound signatures and shallow water source transmissions. In this paper with present analysis of acoustic signals from both shallow and deep water sources on the Chukchi continental shelf for a specific time period between June and August 2017 where a 20 dB intensity drop from along-the-shelf source (S2) at 150 m water depth was observed for more than few weeks. This intensity drop is strongly correlated with occurrence of a large oceanographic event spanning the top 150 m water column due to Pacific Water outflow from Bering sea and retreat of Marginal Ice Zone (MIZ). During the same period, cross-the-shelf source (S1) was not transmitting signal but the reception from the deep water acoustic transmitters also show variability that could be correlated with the basin scale water column variability and the ice coverage. [Work supported by ONR.]

11:45

1aAO12. Beamforming shallow water propagation of mid-frequency acoustics on the Chukchi Shelf. Justin Eickmeier, Mohsen Badiey (Univ. of Delaware, 17 McCord Dr., Newark, DE 19713, jeickmei@udel.edu), Sean Pecknold (DRDC Atlantic, Dartmouth, NS, Canada), and Altan Turgut (Acoust. Div., Naval Res. Lab, Washington, DC)

The shallow water Canadian Basin Acoustic Propagation Experiment (CANAPE 2016–2017) was designed to study the effect of oceanographic variability on the acoustic field in the Arctic. The physics of the acoustic waveguide on the Northeastern edge of the Chukchi Shelf are influenced by dynamic boundary conditions and spatio-temporal fluctuations in temperature/salinity profiles, including the upwelling of Atlantic bottom water, sinking Bering Sea surface water and sub-mesoscale eddy formation. These fluctuations influence the acoustic waveguide characteristics of a persistent sound speed channel centered at 150 m depth. The University of Delaware (UDel) deployed seven oceanographic moorings (OM) perpendicular to the isobaths on the Chukchi Shelf (145–700 m depth). A Naval Research Lab source (S1) was deployed in-line with the OMs, generating Linear Frequency Modulated (LFM) signals in alternating 700–1100 Hz and 1400–4000 Hz bands. A Vertical Line Array (VLA) was deployed in-line and within the span of the OMs. The array aperture spanned the depth of the sound speed channel. Beamforming measurements from the VLA in combination with environmental measurements and 2D PE model output will determine the degree of influence of individual physical oceanography processes on the spatio-temporal structure of the sound channel and internal acoustic wave propagation.

MONDAY MORNING, 5 NOVEMBER 2018 SIDNEY (VCC), 8:45 A.M. TO 12:00 NOON

Session 1aBA

Biomedical Acoustics and Physical Acoustics: State of the Art in Lung Ultrasound: Past, Present, and Future

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via Sommarive, 9, Trento 38123, Italy

Martin D. Verweij, Cochair
Acoustical Wavefield Imaging, Delft University of Technology, Lorentzweg 1, Delft 2628CJ, Netherlands

Chair's Introduction—8:45

Invited Papers

8:50

1aBA1. Sonographic interstitial syndrome: Clinical aspects. Gino Soldati (Clinical and Interventional Ultrasonography Dept., Ospedale Valle del Serchio, via dell'Ospedale, 3, Castelnuovo Garfagnana, Lucca 55032, Italy, gino.soldati@uslnordovest.toscana.it), Andrea Smargiassi, and Riccardo Inchingolo (Pulmonary Medicine Dept., Fondazione Policlinico Universitario A. Gemelli IRCCS, Rome, Italy)

Lung ultrasonography is gaining ever more consideration among physicians for detecting interstitial diseases. When the tissue is aerated, ultrasound may explore only the first peripheral layer of lung. Alveolar geometry and topology interact with ultrasound like air bubbles in a foam (wet or dry). In clinical ultrasound the normal lung appears like a specular reflector. Hyperdense pre-consolidated lung is physically similar to a wet foam and is sonographically diagnosed by artifacts generically called B-Lines (Sonographyc interstitial syndrome—SIS). Many pathologic conditions are involved. In order to give more specificity to SIS and to differentiate clinical patterns, it is crucial to analyse histologic patterns between cardiogenic and pneumogenic pathologies. Cardiogenic involvement develops through the thickening of interlobular septa, then the alveolar flooding occurs. Thickened interlobular septa can represent a specific "deterministic" acoustic trap that could generate specific artifacts. In a later stage, fluid overflows from interstitium and small free intra-alveolar bubbles are generated without structural tissue subversion. In case of pneumogenic SIS there is a structural deformation with folded and collapsed alveoli, thickened interstitium and damaged alveoli. In conclusion, specific artifacts, namely, radiofrequency signals, could be representative of specific patterns of SIS.

9:10

1aBA2. Possible role of research platforms in lung ultrasound. Piero Tortoli, Enrico Boni (Information Eng. Dept, Università di Firenze, via Santa marta 3, Firenze 50136, Italy, piero.tortoli@unifi.it), and Alessandro Ramalli (Cardiovascular Sci., KU Leuven, Leuven, Belgium)

In a few years, ultrasound research platforms, also known as open scanners, have become a great tool for facilitating the experimental activities of ultrasound labs. Ideal platforms should be easily programmed to permit visualization of the region of interest, transmission of arbitrary sequences of arbitrary waveforms, acquisition of massive amounts of raw echo-data, and, possibly, real-time implementation of innovative processing methods. Such characteristics may be particularly relevant in lung ultrasound (LUS) applications, where quantitative methods designed around the lung properties are needed. In fact, since standard US- imaging is designed to handle impedance mismatches that are typically much lower than those found in lung tissue, LUS mostly relies on the qualitative interpretation of imaging artefacts. In particular, full control of all parameters influencing the transmit/receive modalities may allow the test of original transmit/receive strategies, while the access to raw-data may permit the off-line test of novel approaches. In this talk, the main characteristics of advanced open scanners will be reviewed, and sample applications not feasible with standard clinical scanners illustrated. Emphasis will then be given to the specific contributions that such platforms can provide to LUS. Phantom and *in-vivo* results obtained with the ULA-OP research platforms will be presented.

Contributed Papers

9:30

1aBA3. Deep learning for automated detection of B-lines in lung ultrasonography. Ruud J. G. van Sloun (Eindhoven Univ. of Technol., Eindhoven, Netherlands) and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Trento 38123, Italy, libertario.demi@unitn.it)

The application of ultrasound imaging to the diagnosis of lung diseases is gaining attention. Of particular interest are several imaging-artifacts, e.g., A and B line artifacts. A-lines are hyperechoic horizontal lines, which are substantially visualized across the entire image and parallel to pleural-line. They represent the normal pattern of the lung if pneumothorax is excluded. Differently, B-line artifacts correlate with pathology and are defined as hyperechoic vertical artifacts, which originate from a point along the pleura-line and lie perpendicular to the latter. Their presence has been linked to an increase in extravascular lung water, interstitial lung diseases, non-cardiogenic lung edema, interstitial pneumonia and lung contusion. In this work, we describe a method aimed to support the clinicians by automatically identifying the frames of an ultrasound video where B-lines are found. To this end, we employ modern deep learning strategies and train a fully convolutional neural network to perform this task on b-mode images of dedicated ultrasound phantoms (Demi *et al.*, Sci. Rep. 2017). We moreover calculate neural attention maps that visualize which components in the image triggered the network, thereby offering simultaneous localization. Future work includes characterization of the detected B-lines to enable adequate phenotyping of various lung pathologies.

9:45

1aBA4. Ultrasound lung spectroscopy: Preliminary *in-vivo* results. Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive 9, Trento 38123, Italy, libertario.demi@unitn.it), Marcello Demi (Medical Image Processing, Fondazione Toscana Gabriele Monasterio, Pisa, Italy), and Gino Soldati (Emergency Medicine Unit, Valle del Serchio General Hospital, Lucca, Italy)

Nowadays, lung pathologies are mainly diagnosed by X-ray imaging, with the golden-standard being computed tomography (CT). However, CT is not available at patients' bed, not always accessible, and expensive. Additionally, CT delivers a relatively high radiation-dose, with the risk of an increased possibility to develop cancer. Recently, attention is shifting toward ultrasound imaging, which would offer a more portable, more accessible, cheaper, and safer alternative. However, despite promising clinical findings, technical development is lacking. In fact, ultrasound imaging is not optimal for lung investigation being designed for low acoustical impedance mismatches. As a consequence, clinicians base their decisions on qualitative interpretations and imaging artifacts, such as the well-known B-line artifact. In this work, following a recently published *in-vitro* study (L. Demi *et al.*, "Determination of a potential quantitative measure of the state of the lung using lung ultrasound spectroscopy," Scientific Reports, 2017), we report on the first preliminary *in-vivo* results. Standard B-mode, multi-frequency imaging-sequences, and RF-data, were acquired. Consistently with the *in-vitro* study, clinical data show that B-lines can be characterized on their frequency content. These results further support the hypothesis that the analysis of ultrasound spectral features can be used to develop a quantitative method dedicated to the lung.

10:00

1aBA5. Feature detection and pneumonia diagnosis based on clinical lung ultrasound imagery using deep learning. Xinliang Zheng, Sourabh Kulhare, Courosh Mehanian (Intellectual Ventures Lab., 14360 SE Eastgate Way, Bellevue, WA 98007, lzheng@intven.com), Zhijie Chen (Shenzhen Mindray Bio-medical Electronics Co, Shenzhen, China), and Ben Wilson (Intellectual Ventures Lab., Bellevue, WA)

Pneumonia is a common disease with both high morbidity and mortality. The diagnosis of pneumonia remains a clinical challenge, especially in low resource settings where prevalence is high, diagnostic devices are limited, and doctors are scarce. Lung ultrasound has been identified as a useful and low-cost tool for pneumonia diagnosis in many studies. In the present work, we first developed a convolutional neural network (CNN)-based deep learning algorithm to automatically identify four key features linked to lung conditions: pleural line, B-line, consolidation, and pleural effusion. The algorithm was trained using ultrasound data collected from over 150 pediatric and adult patients, with features annotated by expert sonographers. A single shot detection (SSD) framework was developed to detect those features in each video frame image. We then explored the accuracy of diagnosing pneumonia based on one or more lung ultrasound features, using CT as a gold standard. Our results indicate that deep learning algorithms can successfully detect abnormal lung features in ultrasound imagery, and those features can be used to diagnose pneumonia at a high sensitivity level. Computer-assisted ultrasound interpretation has the potential to place point of care, expert-level diagnostic accuracy in the hands of low-resource health care providers for pneumonia diagnosis.

10:15–10:30 Break

Invited Papers

10:30

1aBA6. Ultrasound assessment of pulmonary edema using multiple scattering. Kaustav Mohanty (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell, Behrooz Masuodi, Mir H. Ali, Thomas Egan (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., 911 oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

Ultrasound imaging in the lungs is challenging because of air-filled alveoli which induce multiple scattering. Pulmonary edema is characterized by increased extravascular lung water, which causes acute dyspnea. There is an urgent need from pulmonologists to non-invasively quantify pulmonary edema, in order to monitor response to treatment. The present methodology takes advantage of multiple scattering and calculates the scattering mean free path (L*) for the quantitative assessment of pulmonary edema. L* is a semi-local property, which could be correlated to the amount of fluid buildup in the lung. Pulmonary edema was induced in 4 Sprague-Dawley rats using ischemia reperfusion injury. Following a sternotomy, L* was calculated from the time evolution of the backscattered incoherent intensity using an L11-4v linear array connected to a Verasonics scanner. Ex-vivo data were also acquired for 2 edematous pig lungs. For measurements, the left lung was edematous whereas the right lung was treated as control. Significant differences ($p<0.001$) were found between control ($L^* = 330\pm89\mu m$) and edema ($L^* = 876\pm179\ \mu m$) for *in-vivo* rats. For *ex-vivo* pig lungs, L* was found to be $190\pm84\ \mu m$ for control and $1074\pm361\ \mu m$ for edema. This suggests the potential of L* for the assessment of pulmonary edema.

10:50

1aBA7. Lung ultrasound surface wave elastography. Xiaoming Zhang (Radiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Lung ultrasound surface wave elastography (LUSWE) was developed for noninvasively measuring superficial lung tissue stiffness. In LUSWE, a 0.1 second harmonic vibration is generated on the chest wall of a subject using a handheld vibrator. An ultrasound probe is aligned with the vibration excitation in the same intercostal space to measure the generated surface wave propagation on the lung. A human subject is examined in a sitting position. The lung is tested at the total lung volume and through three intercostal spaces on each lung. In a prospective clinical study, we measure the lung surface wave speeds at 100 Hz, 150 Hz, and 200 Hz on 91 patients with interstitial lung disease (ILD) and 30 healthy control subjects. Significant differences of wave speed between patients and controls were found in all 6 lung regions and for 3 frequencies. We also found positive correlation between LUSWE and clinical assessments including CT and pulmonary function tests. LUSWE is used to track the changes of surface wave speed for patients. LUSWE is a safe and noninvasive technique for generating and measuring surface wave propagation on the lung. LUSWE may complement the clinical standard high-resolution computed tomography for assessing ILD.

Contributed Paper

11:10

1aBA8. On the artefactual information of ultrasound lung images. Marcello Demi (Medical Imaging, Fondazione Toscana Gabriele Monasterio, Via G. Moruzzi 1, Pisa 56124, Italy, demi@ftgm.it), Gino Soldati (Emergency Medicine Unit, Valle del Serchio General Hospital, Lucca, Italy), and Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

In standard B mode imaging, a probe transmits brief pulses and receives the echoes reflected by every structure of the explored medium. A set of consecutive pulses are used to reconstruct a two-dimensional image even though some of the assumptions needed to do this are not exactly satisfied. Consequently, numerous visual artefacts are present in ultrasound images and physicians are aware that significant discrepancies between the ultrasound images and the anatomy of the examined medium may exist. Nonetheless, such artefacts are usually analyzed accurately since they provide clinical information and understanding of the physical mechanisms which are at the basis of their formation is important. In the case of lung images, everything we see beyond the chest wall represents artefactual information since the aerated spaces of the lung reflect most of the energy of the ultrasound pulses. Herein, we will describe the genesis of the most important lung artefacts by means of mathematical models. We will discuss how the so called A lines, B lines, and White Lung may be generated by the reverberation effects between the probe and the pleura, by the trapped and subsequently re-radiated acoustic energy and by the multiple reflections between randomly distributed aerated spaces.

Invited Paper

11:25

1aBA9. A review of pulmonary injury by diagnostic ultrasound. Douglas Miller, Zhihong Dong, Chunyan Dou (Radiology, Univ. of Michigan, 3240A Medical Science I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), and Krishnan Raghavendran (Surgery, Univ. Michigan, Ann Arbor, MI)

Diagnostic ultrasound (DUS) induction of pulmonary capillary hemorrhage (PCH) presents a unique clinical safety issue. Pulmonary DUS has become routine in clinical and point-of-care settings worldwide for diagnosis of edema, effusion, pneumonia, etc. Our experiments in rats have shown that DUS-induced PCH increased rapidly above exposure thresholds, which depended on specific conditions. Using a linear array at 6.6 MHz, peak rarefactional pressure amplitude thresholds were 1.5 MPa for B mode, 1.1 MPa for Angio Doppler, 1.1 MPa for M mode, and 0.6 MPa for pulsed Doppler. Thresholds were nearly constant from 1.5 MHz to 12 MHz, suggesting involvement of a frequency-independent mechanism, such as radiation pressure. Physiological conditions were found to be as important as physical exposure parameters. Infusion of saline reduced the effect. Xylazine in the normal ketamine/xylazine anesthesia enhanced PCH relative to ketamine only, and the clinical sedative dexmedetomidine also lowered thresholds. Rats with positive-pressure ventilation had DUS-induced PCH inhibited by only +4 cm H_2O, or enhanced by only -4 cm H_2O end expiratory pressure acting on the blood-air barrier. With subsequent studies and better understanding of this phenomenon, DUS-induced PCH will be avoidable by use of sub-threshold output by sonographers using DUS machines with output control.

Contributed Paper

11:45

1aBA10. Observation of acoustic fountain generation by diagnostic ultrasound shear wave elastography. Brandon Patterson and Douglas Miller (Radiology, Univ. of Michigan, 1301 Catherine St., Ann Arbor, MI 48109, awesome@umich.edu)

High-intensity focused ultrasound has been shown to drive fountains and atomization at liquid-gas and tissue-gas interfaces. Though these phenomena are not well studied for diagnostic ultrasound, they have been hypothesized to play a role in diagnostic ultrasound-induced pulmonary capillary hemorrhage (PCH). We demonstrate that push pulses used in diagnostic shear wave elastography (SWE) also cause fountaining and atomization at water-air and blood-air interfaces. A focused ultrasound transducer (SSI SL15-4), was aimed upward, through water, at an air interface. An SWE pulse sequence, including four push pulses (amplitude $\leq$ 8.6 MPa, pulse duration $\cong$ 650 ms, and center frequency $\cong$ 5.0 MHz), was initiated. The interface was photographed at 20k fps and four successive fountains were observed. Fountains heights up to 12 and 11 mm were observed for water and blood respectively and ejected water droplets traveled up to 30 cm above the surface. The spacing of the four fountains was measured to be the same as the spacing of four PCH areas observed on rat lungs exposed to the same pulse sequence. Concurrent studies should reveal the relative efficacy of PCH induction from SWE and normal pulse echo and Doppler imaging.

MONDAY MORNING, 5 NOVEMBER 2018 SALON C (VCC), 7:50 A.M. TO 12:00 NOON

Session 1aNS

Noise, Physical Acoustics, Signal Processing in Acoustics, and ASA Committee on Standards: Supersonic Jet Aeroacoustics I

Allan C. Aubert, Cochair
Propulsion & Power, NAVAIR, 2344 Centennial Loop, Round Rock, TX 78665

Caroline P. Lubert, Cochair
Mathematics & Statistics, James Madison University, 301 Dixie Ave., Harrisonburg, VA 22801

Chair's Introduction—7:50

Invited Papers

7:55

1aNS1. Jet aircraft noise—Past, present, and future issues. Richard L. McKinley (Oak Ridge Inst. for Sci. and Education, 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, rich3audio@aol.com) and Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Aircraft noise has been an issue since for the US Air Force since World-War I. The early days of flight had noise issues that were specific to reciprocating engines and propellers. The advent of the jet engine in the 1940s changed the noise issues and focus dramatically. The jet noise problem was first addressed in the Air Force Research Laboratory (AFRL) by Henning von Gierke and has continued

until today with the development of more powerful fifth-generation fighter jet engines. This presentation will focus on the role of AFRL in noise issues with jet aircraft from the early days until now, and how that noise impacted people and communities; the research that fostered noise reductions; noise from current aircraft; the problems and issues that have persisted with jet noise; and some thoughts for future research.

8:15

1aNS2. Full-spectrum near-field acoustical holography for fighter jet noise imaging. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Kent L. Gee, Kevin M. Leete, Mylan R. Cook (Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

The development of acoustic imaging technologies over the previous decade has proven useful for increasing our understanding of the noise generation mechanisms inside the turbulent flows of full-scale tactical aircraft engines. In particular, advancements in near-field acoustical holography have allowed for jet noise source imaging from measurements made on tied-down aircraft over hard reflecting ground surfaces. These images have been limited in bandwidth due to the spatial resolution and extent of the holography arrays. However, improved aperture extension methods have allowed for representation of the lowest frequencies, and a new tool has been developed to produce accurate sound field images at above-Nyquist frequencies for broadband sources. These two technologies are implemented together to extend the imaging of fighter jet noise near-field toward the full spectrum required for an aircraft noise model.

8:35

1aNS3. Localization of directional noise sources from high-performance military aircraft through subarray beamforming analysis. David F. Van Komen, Tracianne B. Neilsen, Blaine M. Harker, Kent L. Gee, S. Hales Swift (Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, david.vankomen@gmail.com), Micah Downing, Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

High-performance military aircraft noise is created by multiple sound generation mechanisms that need to be understood to guide noise reduction efforts and for adequate sound field predictions. Phased-array methods can be used to produce frequency-dependent equivalent acoustic source models. The Hybrid (beamforming) method [Padois *et al.,* J. Sound Vib. **333** (2014)] is applied to an acoustical measurement along a 71-microphone ground-based array, spanning 32 m, placed in the vicinity of a high-performance military aircraft as the engine was operated at different powers. Application of the Hybrid method to the full-array creates an overall equivalent source model that is sufficient for predicting overall field radiation but fails to separate the different noise sources. Applying the Hybrid method to subarrays separates broadband shock-associated noise from the main radiation lobes of turbulent mixing noise. Results show that the subarray-based equivalent source distributions for the different types of noise originate from overlapping source locations. Further analysis of the subarray-based equivalent noise sources using coherence and directionality from the unwrapped phase of the cross-spectral source reconstructions identifies overlapping, frequency-dependent source regions with characteristics unique to broadband shock-associated noise and turbulent mixing noise. [Work supported by an Air Force Research Laboratory SBIR.]

Contributed Papers

8:55

1aNS4. Beamforming of crackle-related events in supersonic jet noise. Aaron Vaughn, S. Hales Swift, Kent L. Gee (Brigham Young Univ., C110 ESC, Provo, UT 84602, aaron.burton.vaughn@gmail.com), Micah Downing, Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC), and Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH)

Crackle is an annoying component of supersonic jet noise. In the far field, crackle is related to the presence of acoustic shocks that develop due to nonlinear propagation; however, the intermittent source events that drive crackle generation are not well understood. This study investigates the apparent source locations of events related to crackle, which could include high-amplitude or steepened, shock-like waveforms. The measured data were obtained through ground-array measurements near a high-performance military aircraft, which was run at different engine powers. The apparent source regions corresponding to different event triggers, such as pressure amplitude, derivative amplitude, and spectral characteristics, are compared. The crackle-related event beamforming is also compared against the results of time-averaged, frequency-domain source localization. [Funded by an AFRL SBIR.]

9:10

1aNS5. Comparing two inverse array methods for source reconstructions of noise radiated from a high-performance jet aircraft. Jacob A. Ward, Kevin M. Leete, Kent L. Gee, David F. Van Komen, Tracianne B. Neilsen (Phys., Brigham Young Univ., N243 ESC, Provo, UT 84602, jacob.ward@live.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

Because direct flow measurements of tactical aircraft jet engines are not currently possible, acoustic source characteristics are instead inferred from array processing. This paper compares two array processing methods using the same array data from a high-performance military aircraft. Hybrid beamforming (HBF) and multisource statistically optimized near-field acoustical holography (M-SONAH) have both been used previously for frequency-dependent jet noise source characterization, but are compared here for the same input data. A 71-element linear array of equally spaced microphones was placed approximately parallel to the shear layer covering a distance of 32 m. Complex pressures obtained from this array served as the input to both methods. Favorable agreement in terms of maximum source location, source shape, and source extent was seen between the two methods' respective results. While the methods continue to have their relative strengths and reasons for use, this favorable agreement indicates that an improved understanding of military jet noise sources has been achieved. [Work supported by an AFRL SBIR.]

Invited Papers

9:25

1aNS6. Meteorological effects on long-range nonlinear propagation of jet noise from a static, high-performance military aircraft. Brent O. Reichman, Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), and Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

The impact of nonlinearity on the propagation of noise from military jet aircraft has been fairly well documented, but only within a few hundred meters from the aircraft. This paper describes analysis of nonlinear propagation for morning static runups of F-35 aircraft at greater distances, out to 1220 m near the direction of maximum radiation and at heights ranging from 0 m up to 30.5 m. A comparison of overall levels with distance and height reveals evidence of significant atmospheric refraction effects, and a general trend of decreasing level with height. Examination of nonlinearity metrics reveals opposite behavior, however. At these distances, nonlinear propagation effects are often strongest in waveforms with lower sound levels, which is counterintuitive. One important finding, however, is that acoustic shock strength can vary greatly from runup to runup, even for seemingly small changes in atmospheric conditions. This analysis demonstrates the need for further research into long-range nonlinear propagation of jet noise through realistic atmospheric conditions.

9:45–10:00 Break

10:00

1aNS7. Large-eddy simulation of supersonic jet noise emanating from an F404 nozzle at model scale. Junhui Liu and Ravi Ramamurti (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, junhui.liu@nrl.navy.mil)

Large-eddy simulations of supersonic jet noise emanating from an F404 nozzle at the model scale have been carried out using the JENRE flow solver. A wall-model method that was previously validated for a high subsonic flow over a flat plate is used to model the boundary layer effect inside the nozzle. The nozzle geometry is a faceted bi-conic convergent-divergent nozzle with a design Mach number equal to 1.65 and the nozzle-exit diameter equal to 5.07 inches. Both mildly and highly overexpanded conditions are tested for heated jets. The time averaged flow field, turbulence intensities, and the far-field noise are compared with available experimental data. The effects of both the boundary-layer thickness and turbulence intensity at the nozzle exit are investigated to assess their impact on jet noise generation and the noise source characteristics.

10:20

1aNS8. Sensitivity of acoustic predictions on the mesh size of large-eddy simulation of supersonic jet. Inman Jang, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, ohm@yonsei.ac.kr), H. S. Joo (Dept. of Mech. and Aerosp. Eng., Seoul National Univ., Seoul, South Korea), S.-J. Shin (Dept. of Mech. and Aerosp. Eng., Inst. of Adv. Aerosp. Technol., Seoul National Univ., Seoul, South Korea), and J. W. Park (Agency for Defense Development, Daejeon, South Korea)

Intense noise produced by the supersonic jet plume from a rocket can cause damage to its structural system and even a failure during launch. In order to predict the acoustic loading by rocket jet noise, large-eddy simulation (LES) is commonly employed for analysis of the jet flow field that is subsequently used as a source condition for predicting the acoustic field. Although desirable from the accuracy standpoint, LES of a supersonic jet is often burdened by an overwhelming computational cost in both runtime and memory. In this talk, we discuss the sensitivity of acoustic predictions on the spatial resolution of LES, and propose a guideline for reducing the mesh requirement. . Here, the flow field of a supersonic jet is calculated using a hybrid RANS/LES scheme on meshes of different sizes, and is fed into the Helmholtz-Kirchhoff integral for prediction of the acoustic farfield. Changes in overall sound pressure level (OASPL) and directivity of the Mach radiation are monitored as the LES mesh varies in size. The study suggests that (a) the accuracy of the acoustic prediction is not much affected by the use of a coarser mesh to a certain extent and (b) the resulting improvement in computation speed and economy thus outweighs the loss of details in LES, given the prediction of acoustic loading as the ultimate goal. (This work was conducted at High-Speed Vehicle Research Center of KAIST with the support of the Defense Acquisition Program Administration and the Agency for Defense Development under Contract UD170018CD.)

10:40

1aNS9. Improvement of aero-vibro acoustic simulation technique for prediction of acoustic loading at lift-off. Seiji Tsutsumi, Shinichi Maruyama (JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Wataru Sarae, Keita Terashima (JAXA, Ibaraki, Tsukuba, Japan), Tetsuo Hiraiwa (JAXA, Kakuda, Japan), and Tatsuya Ishii (JAXA, Tokyo, Japan)

Aero-vibro acoustic simulation for the prediction of harmful acoustic loading at lift-off of launch vehicle is developed. In this simulation technique, high-fidelity large-eddy simulation with computational aeroacoutics based on full-Euler equations is employed for computing jet aeroacoustics and their propagation to the outside of payload fairing. Acoustic field inside the payload fairing is computed by the coupled vibro-acoustic simulation based on finite element method. A simplified fairing model is used for the validation of the present method. An impact hammer test and acoustic vibration test using a loudspeaker in an anechoic chamber are conducted for validating the structural model. Then, the accuracy of this method is validated by using the acoustic vibration test result with a subscale rocket engine.

11:00

1aNS10. Spectral analysis of jet turbulence and radiated sound. Oliver T. Schmidt (MAE, Univ. of California, San Diego, 1200 E California Blvd., MC 104-44, Pasadena, CA 91125, oschmidt@caltech.edu), Tim Colonius (MCE, California Inst. of Technol., Pasadena, CA), Aaron Towne (CTR, Stanford Univ., Pasadena, CA), Georgios Rigas (MCE, California Inst. of Technol., Pasadena, CA), and Guillaume A. Brès (Cascade Technologies Inc., Palo Alto, CA)

Informed by LES data and resolvent analysis of the mean flow, we examine the structure of turbulence in jets in the subsonic, transonic, and supersonic regimes. Spectral (frequency-space) proper orthogonal decomposition is used to extract energy spectra and decompose the flow into energy-ranked coherent structures. We demonstrate that two distinct mechanisms, which can be distinguished by their characteristic frequency scaling and spatial support, lead to the formation of wavepackets—coherent structures that are known for their acoustic importance in the aft-angle radiation of high subsonic and supersonic jets. We compare these characteristics to acoustic source features extracted from hologram sound pressure measurements in a recent publication. The evidence strongly suggests that both mechanisms are active in full-scale jets and comprise the experimentally educed sources of sound.

11:20

1aNS11. Launch vehicle acoustic measurements for community noise model development and validation. Alexandria R. Salton, Michael M. James (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, Alex.Salton@blueridgeresearch.com), Matthew F. Calton (Blue Ridge Res. and Consulting, Provo, UT), Kent L. Gee, Reese D. Rasband, Daniel J. Novakovich, and Brent O. Reichman (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A measurement campaign is being conducted under ACRP Project 02-81 to compile a database of high-fidelity modern rocket propulsion noise measurements for facilitating community noise model development and validation. As part of this measurement campaign, acoustic measurements of the Orbital ATK Antares rocket, launched as part of NASA's eighth cargo resupply mission (OA-8E) to the International Space Station, were collected. The OA-8E acoustic measurement test plan was based on guidelines from the community noise measurement protocol being developed under the same effort. Pressure measurements were collected from 46 microphones at 19 unique geographic locations ranging from 0.2 to 19.1 km of the launch pad. The acoustic measurements and the resultant initial data products related to the OA-8E Antares launch will be presented. Comparisons of the multiple recording systems and microphone heights/orientations will be discussed in relation to their relevance in informing equipment recommendations and site layout related to the protocol design.

11:40

1aNS12. Experimental study of acoustic reduction technique for H3 launch vehicle. Wataru Sarae, Keita Terashima (JAXA, 2-1-1 Sengen, Ibaraki, Tsukuba 305-8505, Japan, sarae.wataru@jaxa.jp), Seiji Tsutsumi (JAXA, Sagamihara, Kanagawa, Japan), Masao Takegoshi (JAXA, Kakuda, Japan), and Hiroaki Kobayashi (JAXA, Kanagawa, Japan)

A subscale Acoustic test, the H3-scaled Acoustic Reduction Experiments (HARE), was conducted to predict liftoff acoustic environments of the H3 launch vehicle currently being developed in Japan. The HARE is based on 2.5% scale H3 vehicle models, which is composed with a GOX/GH2 engine and solid rocket motors, Movable Launcher (ML) models with upper deck water injection system and Launch Pad (LP) models with deflector and lower deck water injection systems. Approximately 20 instruments measured far/near field acoustic and pressure data. Last year the results of the first campaign of the HARE, which aims at understanding the effects of elevation, the shape of ML, were presented. This year, the results of the second campaign which aims at studying the effects of acoustic reduction techniques such as acoustic shields and water injection will be presented.

Session 1aPAa

Physical Acoustics and Signal Processing in Acoustics: Acoustic Metamaterials and Super-Resolution Imaging

Matthew D. Guild, Cochair
Acosutics Div., Naval Research Lab., Code 7165, 4555 Overlook Avenue, SW, Washington, DC 20375

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

Chair's Introduction—8:15

Invited Papers

8:20

1aPAa1. Acoustic imaging with a metamaterial Luneburg lens. Steven Cummer, Yangbo Xie (Dept. of Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, cummer@duke.edu), Yangyang Fu (College of Phys., Optoelectronics and Energy, Soochow Univ., Suzhou, China), Junfei Li, Zhetao Jia, Chen Shen (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC), Yadong Xu (College of Phys., Optoelectronics and Energy, Soochow Univ., Suzhou, China), and Huanyang Chen (Dept. of Electron. Sci., Xiamen Univ., Xiamen, China)

The Luneburg lens is a spherically symmetrical gradient refractive index device with unique imaging properties. Its wide field-of-view and minimal aberration have led it to be successfully applied in microwave antennas. However, only limited realizations of this type of image formation have been demonstrated in acoustics. Here, we apply a new design method for scalable and self-supporting metamaterials to experimentally demonstrate Luneburg lenses for airborne ultrasound. Experimental imaging results are demonstrated.

8:40

1aPAa2. Super-resolution using a microscale membrane array metasurface. Shane W. Lani (Johns Hopkins Appl. Phys. Lab, 11100 Johns Hopkins Rd., M.S. 8-220, Laurel, MD 20723, lani.shane@gatech.edu), Karim G. Sabra, and F. Levent Degertekin (Georgia Inst. of Technol., Atlanta, GA)

A metasurface or 2D metamaterial composed of a membrane array can support an interesting acoustic wave field. These waves are evanescent in the direction normal to the array and propagate in the immersion fluid along the metasurface. These waves are a result of the resonant membranes coupling to the fluid medium and propagate with a group and phase speed lower than that of the bulk waves in the surrounding fluid. Additionally these membranes can generate and detect membrane displacement capacitively such as in an array of Capacitive Micromachined Ultrasonic Transducers (CMUT) as a specific example. A model is developed that can solve for the modes of the membrane array in addition to transiently modeling the behavior of the array to help show the performance of the array in relation to wave propagation. Potential applications of this wave field are examined in the context of subwavelength focusing and imaging. Several methods of acoustic focusing are used on an array consisting of dense grid of membranes and several membranes spatially removed from the structure. Subwavelength acoustic focusing to a resolution of $\lambda/5$, limited by the size of one membrane, is shown in simulations and verified with experiments. This fundamental work in characterizing the waves above the membrane metasurfaces is expected to have impact and implications for transducer design, resonant sensors, 2D acoustic lenses, and subwavelength focusing and imaging.

9:00

1aPAa3. Superresolution through regular sampling: Theory and applications. Annie Cuyt and Wen-shin Lee (Dept. of Mathematics and Comput. Sci., Univ. of Antwerp, Middelheimlaan 1, Antwerp, Antwerpen 2020, Belgium, annie.cuyt@uantwerpen.be)

Estimating fine-scale spectral information from coarse-scale measurements is an important issue in many signal processing applications. The problem of superresolution is therefore receiving considerable attention. We offer a technique that allows to overcome the Shannon-Nyquist sampling rate limitation and at the same time may improve the conditioning of the numerical linear algebra problems involved. The technique is exploiting aliasing rather than avoiding it and maintains a regular sampling scheme [3]. It relies on the concept of what we call an identification shift [1, 2], which is the additional sampling at locations shifted with respect to the original locations in order to overcome any ambiguity in the analysis. Neither the original sampling nor the identification shift need to respect the Shannon-Nyquist sampling theorem. So far the technique shows great potential in magnetic resonance spectroscopy, vibration analysis, echolocation, music signal processing, ISAR radar problems, and DOA or direction of arrival. [1] Annie Cuyt and Wen-shin Lee, "Smart

data sampling and data reconstruction," US patent 9,690,749. [2] Annie Cuyt and Wen-shin Lee, "Smart data sampling and data reconstruction," EP2745404B1. [3] Annie Cuyt and Wen-shin Lee, "How to get high resolution results from sparse and coarsely sampled data," ArXiv e-print 1710.09694 [math.NA].

9:20

1aPAa4. Potential applications of high-index acoustic metamaterials. Farzad Zangeneh Nejad and Romain Fleury (EPFL, EPFL - STI - LWE - ELB 030, Station 11, Lausanne 1015, Switzerland, farzad.zangenehnejad@epfl.ch)

In general, sound propagates in solid materials with a speed that is larger than that of air. Here, we discuss and demonstrate a simple metamaterial design that provides a sound speed slower than that of air. We highlight the potential of high-index acoustic metamaterials for open waveguiding, real-time analog acoustic signal processing, acoustic birefringence, and beam splitting.

9:40

1aPAa5. Far-field superresolution imaging using shaped acoustic vortices. Matthew D. Guild, Jeffrey S. Rogers, Charles Rohde, and Gregory Orris (Acoust. Div., Naval Res. Lab., Code 7165, 4555 Overlook Ave., SW, Washington, DC 20375, matthew.guild@nrl.navy.mil)

The ability to overcome the limitations on resolution due to the effects of diffraction has attracted significant attention in recent years. Previously proposed methods to overcome this limit, and therefore achieve superresolution, have largely been restricted to operating within the near-field region of the aperture. In this work, we will describe how acoustic helicoidal waves can create acoustic vortices that are well below the resolution limit, and how this can enable far-field superresolution acoustic imaging. The acoustic vortices generated in this manner propagate from the near-field into the far-field through an arrangement of stable integer mode vortices, thereby enabling the generation of far-field superresolved features in the acoustic pressure field. In this paper, theoretical and numerical results will be presented for an acoustic aperture which is capable of generating superresolved far-field features in the radiated acoustic pressure, and results will be shown illustrating the superresolution capability of this novel technique. [Work supported by the Office of Naval Research.]

Contributed Paper

10:00

1aPAa6. Noise correlation in a metamaterial: From laboratory to field data. Aida Hejazi Nooghabi (Sorbonne Univ., 4, Pl. Jussieu Case 129, T.46-00, Et.2, Paris 75252, France, aida.hejazi@gmail.com), Julien de Rosny (ESPCI Paris, PSL Res. Univ., Institut Langevin, Paris, France), Lapo Boschi (Sorbonne Univ., Paris, France), and Philippe Roux (ISTERRE, Grenoble Alpes Univ., Grenoble, France)

In the METAFORET experiment, a seismic survey is conducted in a 120 m×120 m flat area, partly occupied by a relatively regular grid of tall pine trees, and partly by a canola field. We study the scattering effects of trees on cross correlations of ambient signal. A wave field is generated by several arrays of sources and recorded at a dense array of receivers within the area of interest. In parallel, we conduct a lab experiment, where the Earth's subsurface and the trees (resonators) are idealized by a thin elastic (aluminum) plate and an array of rods, respectively. The Lamb waves propagating in plate are inherently 2-D and dispersive; thus, a good analogue of seismic surface waves observed in field data. Based on the reciprocity theorem, we can treat receivers as sources and vice-versa, resulting in a virtually uniform source distribution. We auto-correlate recordings corresponding to each virtual source-receiver pair, and visualize the auto-correlation maximum as a function of virtual-source location: this provides a map of energy contributed by virtual sources, inside and outside the region of resonators. From that, we identify the metamaterial position as well as bandgap and propagation band. Results from field and laboratory data are compared.

10:15–10:30 Break

Invited Paper

10:30

1aPAa7. Ultra-compact reflective metasurface for wave manipulation. Jian Chen and Zheng Fan (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Block N3, Singapore 639798, Singapore, chen_jian@ntu.edu.sg)

Recently, electromagnetic metasurfaces have witnessed a promising future for useful devices with extremely compact footprint. However, their mapping to acoustic regimes remains challenging because the acoustic wavelength is considerably larger than the optics, especially at low frequencies. We propose an ultra-compact metasurface that can manipulate the reflected waves upon plane waves incidences. Remarkably, the metasurface simply consists of a rigid plate perforated with deep-subwavelength grooves of varying depths. A theoretical formulation based on multiple-scattering and dynamical diffraction is established to address the underlying mechanism, and used to optimize metasurfaces for wave manipulation. Numerical simulations and experimental measurements are conducted on the designed sample, which are in good agreement and well predicted by the theory. This study provides a comprehensive guideline for the design of ultra-compact and planar acoustic reflective metasurface; moreover, the study may promise an additional avenue to integrate acoustic devices in practical applications.

10:50

1aPAa8. Wavefront shaping of acoustic waves in scattering media. Yuning Guo (Dept. of Mech. Eng., Univ. of Colorado Boulder, 427 UCB, 1111 Eng. Dr., Boulder, CO 80309, yuning.guo@colorado.edu)

As an essential necessity for fundamental study and a broad variety of technological applications, it is important to control the propagation of acoustic waves and select their particular wavefronts. This work is to achieve effective wavefront shaping of acoustic waves along specific paths in strongly scattering media including phononic periodic structures and random media. In the phononic crystal, a kind of artificial phononic structures, wavefront shaping, and super-resolution imaging have been achieved by designing the geometry and material parameters. Furthermore, the confocal technique of the optical beam and acoustic beam has been improved to achieve sub-wavelength imaging in random media with strongly scattering. The presented results of wavefront shaping have implications for tailoring phonon dynamics in scattering media, which offers further possibilities of controllable acoustic waves in complex materials and enlightens the interaction of multi-physics fields.

11:05

1aPAa9. Inverse design method in acoustic wave manipulation. He Gao and Jie Zhu (Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong SAR, China, gaohe077@gmail.com)

We present here the involvement of inverse design method with the acoustic wave modulation, to simplify how we get the required material parameters and design novel acoustic devices that can perform certain prescribed operation. As an example, the acoustic cloaking will be discussed. The existing acoustic invisibility design based on coordinate transformation method needs negative index media which can be realized through complicate structures. With inverse design method, we have designed a circular cloak device with positive index materials, that successfully make a circular scatterer acoustically invisible. This inverse design method takes a significant step towards realizing practical cloaking devices and would open up new possibilities for manipulating acoustic wavefront with simplified parameter distribution and optimized properties.

11:20

1aPAa10. A kind of polar coordinate system adaptable metasurface and its broadband wavefront manipulation. Shanjun Liang (Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong PolyU CF306, Hung Hom, Hong Kong, s.j.liang@connect.polyu.hk), Tuo Liu (Mech. Eng., The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), He Gao (Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Fei CHEN, and Jie Zhu (Mech. Eng., The Hong Kong Polytechnic Univ., Kowloon, Hong Kong)

We report a type of polar coordinate system adaptable metasurface by which a three-dimensional Bessel beam in broadband of 2–6 kHz is realized. For the common design of metasurface, many kinds elements are for a square lattice in the Cartesian coordinate system while sometimes a cylindrical symmetric source can be more suitable for practical working conditions, the 3D Bessel beam for example. Here, we present a unit cell for metasurface by embedding helical blades into an element of required shape. Thus, the lattice structure can be adjusted easily to the distinct coordinate systems. Hence, the arrangement of the elements can be more flexible, which means a larger freedom of design for the requirement of bi-anisotropic metasurfaces can be obtained. With the advantage of helical blades, an ideal control over energy transmission efficiency can be achieved too. Based on this, we investigated the characteristics and performance of the elements and realized a better performed Bessel beam in three dimensional. This design thinking can also be utilized to other kinds of wavefront modulation in broadband for both transmission and reflection control.

11:35

1aPAa11. A rigid acoustic metamaterial with phase modulation and power attenuation. Nansha Gao (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, gaonansha@nwpu.edu.cn)

We introduced a rigid structure into the acoustic metasurface design, proposed labyrinth structure based on the equivalent medium theory and different media are replaced by curly labyrinth. Layered medium theory and equivalent medium theory are combined to design arbitrary acoustic metasurface structure. An acoustic metasurface studied in this paper and realised simultaneous phase modulation and power attenuation in the air, the effective range covered from 30 to 90 degrees, and power attenuation is over 40%. According to layered medium theory which could modulate acoustic wave direction, the metasurface with same function can also be applied to underwater. Its related simulation results are calculated by FEA method. Finally, by introducing the curly labyrinth theory, the underwater acoustic metasurface with simultaneous phase modulation and power attenuation is designed and verification. This paper potential applications in the rigid underwater acoustic metasurface in low frequency, adjustable direction, and sound attenuation.

Session 1aPAb

Physical Acoustics, Noise, and ASA Committee on Standards: Outdoor Sound Propagation I

Vladimir E. Ostashev, Cochair
U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

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

Chair's Introduction—9:00

Invited Papers

9:05

1aPAb1. On the origin of thunder: Reconstruction of lightning flashes, statistical analysis, and modeling. Arthur Lacroix (CEA, Paris, France), Thomas Farges (CEA, Arpajon, France), Régis Marchiano (Institut Jean Le Rond d'Alembert, Sorbonne Université, Paris, France), and François Coulouvrat (Institut Jean Le Rond d'Alembert, CNRS, Université Pierre et Marie Curie, 4 Pl. Jussieu, Paris 75005, France, francois.coulouvrat@upmc.fr)

Thunder from 27 natural lightning flashes of three thunderstorms has been recorded in 2012 in Southern France in the 0.1–180 Hz frequency bandwidth, using a 50 m-wide triangular array of 4 recalibrated microphones in the 0.3–20 km distance range from lightning. Source reconstruction allows to separate, within the acoustical signal, Cloud-to-Ground (CG) from Intra-Cloud (IC) parts of the discharge. The possibility to separate nearby CG events is shown. A total of 36 CG signals and associated spectra is obtained, along with some IC signals. The combination of reconstruction, separation, and frequency analysis provides new insights on the origin of thunder. Thunder infrasound is shown unambiguously to originate dominantly from return strokes. Spectra of CGs and ICs are similar, but of higher amplitude for CGs. No sharp frequency peaks can be clearly evidenced. The influence of distance, therefore of propagation effects, is pointed out. Best fits of energy and frequency gravity center dependence with distance are in agreement with a nonlinear line source propagation. A link between acoustic energy and impulse Charge Moment Change (iCMC) is also indicated. Lightning is modeled as a randomly tortuous line source, and the resulting spectra are compared to observations.

9:25

1aPAb2. Nonlinear reflection of weak shock waves from a rough surface in air. Maria M. Karzova (Phys. Faculty, Moscow State University, Moscow, Russian Federation; Univ. Lyon, CeLyA, Ecole Centrale de Lyon, LMFA - UMR CNRS 5509, Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Thomas Lechat (Univ. Lyon, Ecole Centrale de Lyon and LMFA UMR CNRS 5509, F-69134, Ecully, France), Sébastien Ollivier (Univ. Lyon, Université Lyon 1 and LMFA UMR CNRS 5509, F-69622, Ecully, France), Didier Dragna (Univ. Lyon, Ecole Centrale de Lyon and LMFA UMR CNRS 5509, F-69134, Ecully, France), Petr V. Yuldashev, Vera Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (Univ. Lyon, Ecole Centrale de Lyon and LMFA UMR CNRS 5509, F-69134, Ecully, France)

Irregular reflection of weak acoustic shock waves occurs under the framework of the von Neumann paradox. In this study, the influence of the surface roughness on the reflection pattern was studied experimentally using spark-generated spherically divergent *N*-waves of 1.4 cm length reflecting from rigid rough surfaces in air. Dimensions of the roughness were varied from 20 up to 500 μm for different surfaces. A Mach-Zehnder interferometry method was used to reconstruct the pressure waveforms near the surface. The reconstruction was performed by applying the inverse Abel transform to the phase of the signal measured by the interferometer. It was shown that the height of the Mach stem became shorter for surfaces with larger dimensions of the roughness and disappeared when the surface roughness was large enough. Such tendency was also observed in simulations based on the Euler equations where the acoustic source was introduced as a Gaussian-envelope energy injection and the roughness was either sinusoidal or random and described by a Gaussian correlation function. [Work supported by RSF-17-72-10277 and by the Labex CeLyA of Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ ANR-11-IDEX-0007).]

9:45

1aPAb3. Modeling of acoustic pulse propagation and beamforming in forests. Michael B. Muhlestein, Vladimir E. Ostashev, and D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Acoustic pulse propagation through forests is important for many applications, including noise attenuation by stands of trees, and localization of sound sources. Due to the highly complicated distribution of trees in natural forests, it is appropriate to consider a forest as a continuous distribution of scatterers of various shapes and sizes. A propagation model based on radiative transfer theory under a modified Born approximation may be developed for this situation to describe both the coherent and diffuse sound propagation . The simple case of an impulse in an infinite homogeneous forest of diffuse scatterers is first considered, and then the effects of successively including non-diffuse scatterers, ground reflections in a forest of finite height, and, finally, a realistic forest model are analyzed. These theoretical findings are then compared with experimental results. Lastly, a numerical example describing the effect of a forest on a simple beamforming array is considered.

10:05

1aPAb4. Correspondence between sound propagation in discrete and continuous random media. Vladimir E. Ostashev, D. K. Wilson, Michael Muhlestein (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd, Hanover, NH 03755, vladimir.ostashev@colorado.edu), and Keith Attenborough (Open Univ., Leighton Buzzard, Bedfordshire, United Kingdom)

Wave propagation in random media is important in many applications such as sound propagation in a turbulent atmosphere and scattering by bubbles and microparticles in the ocean. Formulations for the statistical moments of the sound pressure field in continuous and discrete random media are usually done independently. In this presentation, it is demonstrated that the equations for the first two statistical moments in a continuous random medium have the same form as those for a discrete random medium if the scattering properties of the media are expressed in terms of the differential scattering cross section and total cross section. This analogy enables us to apply methods developed in wave propagation in continuous random media to discrete media, and vice versa. As an example, the existing theory of the interference of the direct and ground reflected waves in a turbulent atmosphere is used to study the effect of trees on the interference of these waves in a forest. The results obtained are compared with experimental data. The correspondence between wave propagation in discrete and continuous random media can also be used in other fields of physics.

10:25–10:40 Break

10:40

1aPAb5. Bayesian estimation of mean transmission loss along multiple paths with randomly scattered signals. D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Vladimir E. Ostashev (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A general problem is considered in which a source of unknown power transmits to multiple receiver locations. The signal is randomly scattered along each transmission path, for example, by turbulence, a forest, or buildings. At each receiver location, one or more observations of the signal power are collected. It is assumed that the functional form of the probability density function (pdf) of the received signals (an exponential pdf for strong scattering, or a gamma pdf for weak scattering) is known based on an understanding of the scattering process, although the mean transmission losses (TLs) from the source to receivers are uncertain. From the signal observations, we wish to estimate the mean power of the source and the mean TLs for each path. A Bayesian formulation for this problem is presented. An inverse gamma prior is used for the TL, which is the conjugate prior for the exponential or gamma scattered signal pdf (likelihood function). Analytical solutions are then derived for a number of limiting scenarios: (1) unscattered signals along multiple paths with dependent TLs, (2) strongly scattered signals along multiple paths with the same TL, and (3) weakly or strongly scattered signals along multiple paths with independent TLs.

11:00

1aPAb6. An improved numerical scheme for predicting sound propagation in a vertically stratified medium. Yiming Wang and Kai Ming Li (Mech. Eng., Purdue Univ., 140 South Russell St., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

Propagation of sound in a stratified medium has been studied for many decades. The ray tracing method, which is one of the most popular ways to solve this kind of problems, involves identifying the sound rays connecting the source with the receiver. However, the ray method does not yield accurate solutions when either the source or the receiver is placed near a turning point. Under these circumstances, it is found necessary to use a wave-based approach leading to a more accurate solution for predicting the sound fields. Typically, the solution is written in a form of a highly oscillatory integral that can be evaluated numerically. High computational times are required to obtain accurate solutions. There have been significant developments in the numerical approximation of highly oscillatory integrals. An improved scheme, which is known as Levin's collocation method, has low computational costs but it gives accurate numerical solutions for high-frequency sound fields. In this paper, the propagation of sound in a stratified and unbounded medium is considered where there is only one turning point. The Levin collocation method is explored for calculating the sound fields efficiently. In addition, an asymptotic solution is also derived for comparison with the Levin collocation method.

11:20

1aPAb7. Nondimensional analysis of wind noise and atmospheric surface-layer properties. Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@usace.army.mil), Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Christopher M. Hocut (U.S. Army Res. Lab., White Sands Missile Range, NM)

Wind noise is a prominent limitation to the signal to noise ratio of acoustic sensors. Realistic expectations of signal detectability can be generated by predicting the noise floor prior to a sensor deployment; however, a relationship must be established between the wind noise measured by a sensor and atmospheric surface-layer properties. Under conditions of horizontal homogeneity and quasi-steadiness, Monin-Obukhov similarity theory relates friction velocity, temperature scale, and roughness length to the near-surface profiles of mean wind speed and turbulent intensity, which in turn are known to govern wind noise. It is expected that the ratio of one-third octave band root-mean-square sound pressure to the turbulent flux of momentum, Strouhal number, and dimensionless elevation have a nondimensional relationship that collapses wind noise data as a function of Monin-Obukhov parameters. In order to establish such a relationship, we analyze a dataset of wind noise recorded in Spring 2018 within the Army Research Laboratory's Meteorological Sensing Array on the Jornada Experimental Range, New Mexico. This dataset consists of continuous recordings of ambient noise at several sites on audio microphones up to 20 m above ground level, co-located with a suite of high-fidelity meteorological instruments, including sonic anemometers.

11:40

1aPAb8. Calculations of low-frequency wind noise along a low two-dimensional hill surface. Gregory W. Lyons (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, gregory.w.lyons@erdc.dren.mil) and Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Measurement of outdoor sound propagation is often limited by wind noise, i.e. pressure fluctuations from atmospheric turbulence, especially for infrasound and low audible frequencies. Over a flat ground surface, the spectral density of wind noise can be predicted by the shear-turbulence mechanism for static pressure fluctuations using a mirror flow atmospheric turbulence model for the inhomogeneous surface-blocking effect. This study moves beyond a flat ground model to consider the effects of flow distortion by weak topography on surface wind noise, specifically, flow over a low two-dimensional hill, free from separation, at large Froude number. The integral solution for the pressure Poisson equation is used as a starting point, with turbulence modeled by the mirror flow in surface-following coordinates. The perturbation analysis of Hunt *et al.* [Q. J. R. Meteorol. Soc. 114(484), 1435–1470 (1988)] is used to model the mean shear rate as a function of elevation and distance over the hill. For upwind, downwind, and crest positions along the hill surface, the pressure correlation solution is integrated numerically to evaluate one-dimensional spectra. The relative forms of these spectra are analyzed to describe the effects of the hill crest acceleration and downwind wake deficit. Implications for microphone placement are also considered.

Session 1aSAa

Structural Acoustics and Vibration, Physical Acoustics, Underwater Acoustics, and Architectural Acoustics: Advanced Modeling Techniques for Computational Acoustics

Kuangcheng Wu, Cochair
NSWCCD, 9500 MacArthur Blvd., West Bethesda, MD 20817

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg. 1302, Newport, RI 02841

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

Invited Papers

8:00

1aSAa1. Scalar metrics for structural acoustic system analysis and differentiation. Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., P.O Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu)

As computational resources continue to grow and as researchers are better able to harness these resources, the size and complexity of numerical models for studying structural acoustic systems will only increase. However, despite these ever larger models, there remains a need to reduce the results into an easily interpreted form. In this work, a variety of scalar metrics—where *scalar metric* is used to refer to any scalar-valued function of either frequency or time—are presented and evaluated for their utility in displaying relevant features of structural acoustic models. Particular attention is paid to metrics that allow an analyst to differentiate between related models, such as when performing a design study where several different options are being considered. The metrics are ranked for several example problems according to how well they describe the features of interest as well as how readily they may be computed.

8:25

1aSAa2. Improving multiple model parameters of a complex fluid-loaded structure from acoustic measurements. Alyssa T. Liem and James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, atliem@bu.edu)

A method for correcting multiple mechanical and acoustical properties in a finite element model using the Neumann series is presented and demonstrated with numerical examples. In previous work, the authors developed a method for estimating a model parameter in a complex structure using a Neumann series as an approximation to system response. The method computed the sensitivity of the system response due to changes in the parameter and subsequently determined the change needed in the parameter to bring model response into agreement with vibration measurements. This previous work demonstrated the accuracy and computational efficiency of the method when correcting one model parameter. In the present work, the authors extend the analysis to compute sensitivities for multiple parameters of a system, allowing for the correction of more than one parameter. The limits and accuracies of the method are explored for a canonical acoustic system in which a complex structure interacts with an acoustic medium. Two uncertain model parameters of the system are corrected by bringing model response into agreement with acoustic measurements. Results of these examples will be reviewed and presented to illustrate the accuracy and robustness of the method.

8:50

1aSAa3. On the study of vibrational interactions of an internal substructure with a main structure submerged in water and its acoustic radiations using admittance approach. Pei-Tai Chen (Dept. of System Eng. and Naval Architecture, National Taiwan Ocean Univ., No.2,Pei-Ning Rd. Keelung 20224, Taiwan, ptchen@mail.ntou.edu.tw)

It is important and practical to design an internal substructure for supporting machines which generates vibration sources for a main submerged stiffened shell structure. Vibration propagates from the internal supporting structure to the wetted shell structure, thus radiating acoustic energy into water. The present study can be divided into two categories: 1. the main wetted structure, including stiffeners and bulkheads, etc., radiates acoustic energy into water subject to forces which is the junction interacting forces arisen from the vibrating machine, 2. the interaction force pertains to the coupling dynamic characteristics between wetted structure and the internal structure. An admittance approach is adopted to characterize individual structures and the coupled equation is derived by continuity of displacement variables and equaling forces with opposite sign at the junction of connected structures. Admittance matrices are computed by using junction forces between the structures whereas the responded displacements at the junctions are the elements of the admittance matrices. The coupled admittance equation is complex symmetric matrix. An eigenvalue analysis is performed to investigate the interaction junction forces, accordingly, the radiation characteristics of the coupled main wetted structure under fluid loading in connection with supporting internal structures.

Contributed Papers

9:15

1aSAa4. Deterministic and statistical parameter characterization in resonant fluid-structure interaction problems. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, timo.lahivaara@uef.fi), Peter Göransson (Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., Stockholm, Sweden), and Jacques Cuenca (Siemens Industry Software, Leuven, Belgium)

This research focuses on developing computational methods to estimate model parameters in resonant fluid-structure interaction problems over a wide frequency range by means of model inversion approaches. The considered problems are widely known to be subjected to local minima, which represent a major challenge in the field of parameter identification. In the proposed method, the frequency spectrum is divided into successive substeps allowing to efficiently guide the estimation towards the global minimum, i.e., the true model parameters. The estimation is performed through two frameworks, namely, the deterministic using gradient-based optimization and Bayesian using Markov chain Monte Carlo method. Proposed numerical examples illustrate the effectiveness and potential of the proposed stepwise scheme to find the global minimum and reduce the overall computational burden.

9:30

1aSAa5. On the use of model truncation to extend the applicability of the finite element method to higher frequencies. Anthony L. Bonomo, Joshua McWaters, and Kuangcheng Wu (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, anthony.l.bonomo@navy.mil)

Many structural acoustics problems of interest can be modeled as a vibrating elastic structure situated in and fully coupled to an infinite acoustic fluid domain. To model such problems using the finite element method, techniques have been developed to approximately enforce the Sommerfeld radiation condition at the boundary of the computational domain and prevent spurious boundary reflections from adversely affecting the calculated solution. These techniques include radiation boundary conditions, infinite elements, and perfectly matched layers. It is well known that due to computational constraints, the finite element method is often restricted to relatively low frequencies. However, many of the same techniques that have been used to enforce the Sommerfeld radiation condition can also be used to truncate the computational domain further and allow the finite element method to be used to study higher frequency problems where the structural acoustic response is relatively localized. This talk explores this model truncation application.

9:45

1aSAa6. Acoustic wave scattering from a wave-bearing cavity in a rectangular waveguide. Muhammad Afzal and Hazrat Bilal (Dept. of Mathematics, Capital Univ. of Sci. and Technol., Islamabad 44000, Pakistan, dr.mafzal@cust.edu.pk)

This article discusses the acoustic scattering in a waveguide containing flexible cavity bridged by vertical membranes. A tailored-Galerkin approach is adopted for the solution of governing boundary value problem. Unlike to the usual Mode-Matching (MM) technique, the schemes adopted here for solution modify the MM procedure by using Galerkin and Modal approaches. In Galerkin process, the displacement of vertical membrane is expanded by means of the usual orthogonal modes whilst in the later case, this displacement is found by utilizing the non-orthogonal modes of wave-bearing cavity. The results for scattering powers and transmission loss are shown against frequency and the dimensions of the chamber. A good agreement in results obtained via Galerkin and Modal approaches for different sets of edge conditions is seen. The numerical results show that the choices of edge conditions significantly affect the transmission loss and scattering powers.

10:00

1aSAa7. Multi-layer actuator array to render a vibration field on a point-excited panel speaker. Ki-Ho Lee, Jeong-Guon Ih, and Youngjin Park (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, k.h.lee@kaist.ac.kr)

Panel speakers often adopts the vibration actuator attached to a plate center to excite the whole panel. When a thin rectangular plate is excited by a point force, the generated bending wave is reflected quickly from the edges, so the plate is governed by the reverberant field. Because many vibrational modes participate even in the low frequencies, the radiated sound spectrum is involved with many peaks and troughs resulting a poor sound quality. To minimize the modal participation, the vibration is rendered to be confined and in-phase within a circular area enclosing the actuator point, and the vibration is being suppressed outside of it. An additional multi-layer actuator array surrounding the speaker zone is employed to control the vibration, thus fulfilling the rendered field. The study aim is now to obtain an appropriate gain of the actuators by solving the inverse problem consisting of the transfer matrix between field points and control actuators. The effect of the number of arrays is tested for the radius of 0.05–0.2 m. The control result reveals that the signal-to-noise ratio in the speaker and baffle zone is improved 8–11 dB by the three-layer array than by the single-layer array with the same size.

Session 1aSAb

Structural Acoustics and Vibration, Engineering Acoustics, and Signal Processing in Acoustics: Utilization of High-Speed Cameras to Measure Vibration

Micah R. Shepherd, Cochair
Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801

Trevor W. Jerome, Cochair
Department of Acoustics, The Pennsylvania State University, GTWT Water Tunnel, State College, PA 16804

Invited Paper

10:45

1aSAb1. Deflectometry, a full field slope measurement technique: General perspectives and application to loading identification using the virtual fields method. Olivier Robin, Patrick O'Donoughue, and Alain Berry (Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Olivier.Robin@USherbrooke.ca)

The ability of photographic techniques to quickly record high volumes of scientific data has been understood since the early 1900s. It is only following the advent of digital imaging and data processing systems that so-called full-field optical measurement techniques became sufficiently reliable for performing high spatial density structural vibration measurements. In particular, the deflectometry technique directly provides a spatially and temporally resolved measurement of slope fields on plane structures using a single high-speed camera. The presentation first demonstrates the principles of this technique and explores some direct applications of such full-field measurements. The advantages and drawbacks of deflectometry are compared with other optical techniques like digital image correlation or scanning laser Doppler vibrometry. A key aspect of several engineering domains consists in the identification of dynamic loads acting on structures by inverse methods. It is shown that coupling deflectometry measurements with the virtual fields method enable the reconstruction of stationary and transient excitations without any specific regularization. Experimental reconstruction results on an aluminum panel are presented for two different transient mechanical loadings: instrumented impact hammer and impacting metal marbles (multiple unknown excitations). Finally, the identification of random excitations is considered using a plate and a membrane.

Contributed Papers

11:10

1aSAb2. Estimating Poisson's ratio of a free, rectangular panel using video-based modal analysis. Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu), Olivier Robin (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada), Stephen Hambric (Appl. Res. Lab, Penn State Univ., State College, PA), and Patrick O'Donoughue (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada)

Recent work has shown that the Poisson's ratio of an isotropic material can be determined using the anticlastic curvature that exists in certain mode shapes of a free, rectangular panel of that material. The shapes must be measured experimentally in order to determine the curvature that exists. The curvature is then related to Poisson's ratio based on a relationship that depends on thickness and length-to-width ratio. For accurate determination of the anticlastic curvature, high spatial resolution is required. In this paper, high speed video is used to experimentally measure the mode shapes of a free, rectangular panel. The spatial resolution achieved is much higher than that obtained using traditional methods due to the inherent resolution of the camera. The high-speed video results are demonstrated and compared to modes using traditional modal techniques. The Poisson's ratio is then computed and found to agree well with published values.

11:25

1aSAb3. The problem of parallax when using high speed cameras for measurement. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

When using a high-speed camera as a recording device to measure the displacement or position of an object, the problem of parallax must be properly accounted for, or significant errors in measured positions can occur. In this paper, an experiment to measure the elastic properties of several golf balls of varying construction is described. The goal of the experiment was to verify whether a correlation exists between a ball's coefficient-of-restitution (COR) and the frequency of its lowest structural vibration mode. Balls were dropped from a height onto a rigid surface and the COR was obtained by taking the ratio of speeds just before and just after impact with the rigid surface. Balls were dropped and bounced in front of a scale with fine graduation markings. Video recorded with a high-speed camera was processed using tracking software to measure the ball's position and velocity as a function of time. However, parallax due to the camera field of view introduced significant error into measurement of position and velocity. This paper describes how the parallax error was accounted for in the video-recorded measurements in order to obtain data that matched theoretical expectations.

11:40

1aSAb4. A method for measuring the dynamic parameters by extracting the incident. Hong Hou, Zhengyu Wei, Nansha Gao (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an Shaanxi 710072, China, houhong@nwpu.edu.cn), and Jianhua Yang (School of Automation, School of Automation, Northwestern PolyTech. Univ., Xi'an, China)

The complex Young's modulus of viscoelastic materials can be determined by monitoring the propagation of a traveling burst in a thin bar, which is called wave-speed method. While the test samples are always too short to make the incident wave for lower-frequency signals be detected, which lead to low-frequency test is difficult using this method. To solve this problem, a method of extracting the incident wave is presented in this paper. A pulse wave is generated by the exciter and is used to force the longitudinal vibration of a viscoelastic thin bar. The velocities of the two ends of the thin bar are measured by two laser Doppler vibrometers. The incident waves at both of the two ends of the sample can be easily extracted, and the dynamic parameters of the sample can be obtained by the ratio of the vibration velocities of them. This method expands the experimental frequency range of the wave-speed method and the measured results agree well with that of commercial viscoelastic instrument.

MONDAY MORNING, 5 NOVEMBER 2018 SHAUGHNESSY (FE), 7:55 A.M. TO 12:00 NOON

Session 1aSP

Signal Processing in Acoustics, Acoustical Oceanography, Architectural Acoustics, Musical Acoustics, Underwater Acoustics, and Noise: Machine Learning for Acoustic Applications I

Peter Gerstoft, Cochair
SIO Marine Phys Lab MC0238, Univ of California San Diego, 9500 Gillman Drive, La Jolla, CA 92093-0238

Weichang Li, Cochair
Aramco Research Center - Houston, 16300 Park Row, Houston, TX 77084

Chair's Introduction—7:55

Invited Papers

8:00

1aSP1. Machine learning in acoustic applications with wave physics models, a tutorial. Weichang Li (Aramco Res. Ctr. - Houston, 16300 Park Row, Houston, TX 77084, lwc@alum.mit.edu)

Through recent decades of intensive research, machine learning has established its great potential in areas such as social networking, e-commerce, computer vision, natural language processing, and robotics. Recently, it has also started to capture the attention of the acoustic research community where the problems have wave physics basis and the research has focused on related areas such as signal/image processing. This is evident in the significantly increased number of related papers presented at recent ASA meetings. The challenge appears to concerning several aspects: i) What types of acoustic applications can be formulated as ML problems in a way that could provide interesting results and potentially better performance? ii) What would be the relationship between wave physics models and machine learning methods? iii) What is the dataset requirement of machine learning that is potentially different from say, signal processing techniques, and what implications it might have on experimental design, data collection, and annotation? ii) What are the areas that might not have much benefit from taking a machine learning approach? This tutorial will present a number of acoustics/geophysics application examples along with their analogs in canonical machine learning applications, as an attempt to illustrate possibilities and limitations.

8:40

1aSP2. InversionNet: A real-time and accurate full waveform inversion with convolutional neural network. Youzuo Lin and Yue Wu (Geophys., Los Alamos National Lab., Bikini Atoll Rd., Los Alamos, NM 87545, ylin@lanl.gov)

With data proliferation in all geosciences domains, machine learning and data analytics are emerging as important research areas in geosciences. Full-waveform inversion has been an important tool to infer the subsurface based on geophysical measurements. However, solving full-waveform inversion can be challenging. The existing computational methods for solving acoustic full-waveform inversion are not only computationally expensive but also yields low-resolution results because of the ill-posedness and cycle skipping issues of full-waveform inversion. To resolve those issues, we employ machine-learning techniques to solve full-waveform inversion in this work. In particular, we build a convolutional neural network to model the correspondence from data to velocity structures. Our numerical examples using synthetic reflection data show that our new methods much improve the accuracy of the velocity inversion. Furthermore, the computational time for inverting the seismic data is significantly reduced.

9:00

1aSP3. The machine learning aspects in building the global smartphone seismic network. Qingkai Kong and Richard Allen (Earth & Planetary Sci., UC Berkeley, 289 McCone Hall, UC Berkeley, Berkeley, CA 94720, kongqk@berkeley.edu)

MyShake is a global crowdsourcing smartphone seismic network to monitor and detect earthquakes. After it got released to the public in 2016, we arrived at more than 300,000 downloads with more than 800 detected earthquakes globally within 2 years. Machine learning plays a critical role in MyShake that makes everything happen. In this presentation, I will present the details of how we use the artificial neural network to distinguish earthquakes from the human activity movements recorded on a single phone in real-time. This includes how we do the data acquisition, pre-processing the data, addressing imbalanced datasets, feature engineering/selection, and evaluating the model. I will also talk other machine learning aspects in the MyShake network including the convolutional neural network that we built on the server to further classify the whole waveforms to find that caused by earthquakes, the adversarial machine learning for securities of the system, dealing with the dynamically changing network, training customized model for each user etc. These machine-learning applications in MyShake illustrate the power of combining data science and geophysics and provide good examples of how do we better facilitate the interactions of the two fields.

Contributed Papers

9:20

1aSP4. Total independent energy distance—A measure for choosing efficient spectrogram resolutions. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

This paper describes a quantitative method for choosing the most efficient frequency resolution and window overlap when creating spectrograms. The method calculates the Euclidean distance, d, and the coefficient of determination, R^2, between a pair of successive power spectra in the spectrogram. The quantity $d \times (1—R^2)$ is the Euclidean distance between the windows' power spectra, discounted by the information redundancy between the spectra. The Total Independent Energy Distance (TIED) is then the sum of those distances along the entire spectrogram. The effectiveness of TIED is computed for a variety of artificial signals including white noise, sawtooth waveforms that vary in their frequency resolution, pulse trains, and amplitude-modulated signals that vary in their time distribution, and frequency sweeps that vary in both. The TIED is then calculated for human speech, bird song, and insect buzzes, demonstrating that it provides quantitative support to time- and frequency-resolutions for the spectrograms that closely adhere to the traditional resolutions that have been arrived at through generations of human pattern recognition.

9:35

1aSP5. Dereverberation binaural source separation using deep learning. Dhany Arifianto and Mifta N. Farid (Dept. of Eng. Phys., Insitut Teknologi Sepuluh Nopember,Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id)

This paper reported a deep-learning based binaural separation using gammatone-frequency cepstral coefficient (GFCC) and multi-resolution cochleagram (MRCG) as spectral features and interaural time difference (ITD) and interaural level difference (ILD) as spatial features. A binary mask was estimated by deep neural network (DNN) binary classifier that used the features as a training data and ideal ratio mask (IRM) as a training target. In the experiment, a male speaker as a target speech at azimuth 0^o and a female speaker as a masker speech at azimuth 30^o, 20^o, 10^o, and 5^o in rooms with 0.32, 0.47, 0.68, and 0.89 s reverberation time (RT_{60}). As a training process, 50 mixtures were used for each condition experiment. The classifier contained two hidden layers, 200 binary neurons and 50 epoch, and Restricted Boltzmann Machine (RBM) was used as pre-training process. The RBM and the classifier learning rate was 1 up to 0.001 from epoch 1 to epoch 50. The sound quality results indicated by 88% intelligibility of STOI method which means that the separated sound is easily understood by the listener. The MOS value is 2.8 which means the sentence is clear but spectral distortion is slightly annoying.

9:50

1aSP6. Predicting transmission loss errors through machine learning. Jennifer Cooper, C. J. Della Porta, and Olivia Ott (Johns Hopkins Univ. Appl. Phys. Lab., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu)

Uncertainty in a sound speed profile can lead to uncertainty in the associated prediction of transmission loss (TL). In order to better quantify the effect of imperfect knowledge of the sound speed profile on the acoustic propagation, a study was done comparing pairs of sound speed profiles. In each pair, one profile was treated as ground truth and the second profile was a perturbed version of the first. No single metric on the sound speed profiles, such as mixed layer depth or surface layer characteristics, correlated well with the errors in resulting TL. Several attempts at creating a more complex metric on the profile that could predict errors in the TL were also unsuccessful. However, even a rather simple machine learning approach was able to reliably predict TL errors. Results of the study will be presented and implications discussed.

10:05–10:20 Break

Invited Papers

10:20

1aSP7. Deep learning for estimating porous material properties from full-waveform data. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, timo.lahivaara@uef.fi)

This research focuses on both developing computational tools to model transient wave propagation in coupled poroviscoelastic-viscoelastic-acoustic media and estimating material properties from the recorded full-waveform data. Numerical simulation of wave-dominated problems is computationally demanding. Efficient parallelization, capability to handle complex geometries, and sufficient numerical accuracy are some of the requirements for a suitable full-waveform simulation technique. On the other hand, the robustness and prediction accuracy are needed from the method used to solve the corresponding inverse problem. In this work, the discontinuous

Galerkin method is used to solve the forward model while the convolutional neural networks is used to solve the estimation problem. Two-dimensional model problems with simulated data are presented. In the numerical experiments, the primary unknowns are estimated while the remaining parameters which are of less interest are successfully marginalized.

10:40

1aSP8. Machine learning for behavior classification of passively tracked vessels. Erin M. Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu), Oscar A. Viquez, and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

Situational awareness is a constant challenge for autonomous underwater vehicles (AUVs) due to limited communication to the surface, navigation drift, and a need to operate in busy areas around other vessels. One piece of information that would be valuable to both keep AUVs safe and inform autonomous monitoring missions is the location and behavior of nearby surface vessels. Passive acoustic data collected on a hydrophone array and processed on an on-board computer can provide bearing and time-to-intercept (TTI). This information can be used to classify overall boat behavior and inform AUV response: for example, investigating an area a boat has circled or avoiding an approaching vessel for vehicle safety reasons. Simulation studies were used to characterize trajectories for simple vehicle behaviors based on the bearing and TTI data produced by an existing passive tracking system. A classifier based on K-nearest-neighbor with dynamic time warping as a distance metric was used to classify simulation data. The simulation-based classifier was applied to classify experimental tracking data on boats completing different types of behaviors using bearing/TTI from dock-based and AUV-based hydrophone arrays.

Contributed Papers

11:00

1aSP9. Proof of concept: Machine learning based filling level estimation for bulk solid silos. Paaranan Sivasothy, Matthias Andres, and Gregor Corbin (Dept. of Mathematics, Technomathematics, Gottlieb-Daimler-Straße 44, Kaiserslautern 67663, Germany, sivasothy@mv.uni-kl.de)

Rigid silos are often under pressure and filled with hazardous materials. Therefore, in very few cases their level can be checked visually. For example, the level of mobile bulk solids silos, which are often used on construction sites, is checked by a worker throwing a stone against the silo and uses the sound to estimate the invisible silo cavity. This method is subjective and shows great errors based on experience. The proposed method should implement this principle robustly by machine application. A sensor unit is to give a mechanical impulse to the outer silo wall with a percussion mechanism and record the resulting sound with a microphone. Just as a human being is able to make an experience-based statement about the filling level of the silo over the growing amount of sounds heard, different machine learning approaches are to be considered in order to imitate this procedure. Machine learning approaches usually aim to find patterns between different sizes. A strongly approximated mathematical model will be used to prove that there is a noisy but demonstrable direct correlation between the level and the acoustic impulse response. This should justify future efforts to find suitable machine learning methods for this application.

11:15

1aSP10. Grating lobe prediction and deconvolution for synthetic aperture sonar. Jeremy Dillon (Kraken Robotics, 430 Water St., Ste. 100, St. John's, ON A1C 1E2, Canada, jdillon@krakenrobotics.com)

Synthetic aperture sonar (SAS) arrays are typically undersampled in the sense that the array element spacing is much larger than the acoustic wavelength. Grating lobes are suppressed by making a judicious choice of beampattern nulls for the transmit and receive elements, such as a 3:2 ratio for the length of the transmit and receive elements. However, grating lobe artifacts can appear in SAS imagery when the target strength difference between a highly reflective object and the surrounding seabed exceeds the sidelobe level of the synthetic array. We present a theoretical model of the SAS point scattering function (PSF) that takes into account shaded element beampatterns for a multichannel SAS array. The PSF model is validated using experimental data from AquaPix, a wideband 300 kHz interferometric SAS. In practice, observed SAS images are described by the convolution of the seabed reflectivity with the PSF. Therefore, knowledge of the PSF facilitates the removal of grating lobe artifacts using deconvolution techniques such as the Richardson-Lucy algorithm. Conventional and deconvolved SAS images of a highly reflective target are presented to demonstrate the effectiveness of deconvolution based on the modeled PSF.

11:30

1aSP11. Geospatial estimation of noise levels between sparsely distributed sensor nodes using machine learning. Matthew G. Blevins and Gordon M. Ochi (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil)

Monitoring noise levels over large areas is typically limited by the number of sensor nodes. While previous studies have been able to accurately estimate noise levels using densely distributed sensors in confined spaces, such as along roads in urban areas, estimating noise levels with relatively few sensors spaced over a large area remains a challenging problem. Point sampling of noise levels due to single noise events is also limited by the models used to estimate between sensor locations and their inherent assumptions. To address these limitations we propose nonlinear, adaptive, and robust tools to estimate levels between sensor nodes based on machine learning. Random forests, support vector machines, and Gaussian processes are explored along with conventional geostatistical methods such as ordinary kriging. These methods are trained and evaluated on data from both measurement and simulation of blast noise on military testing and training ranges. The performance of the methods is evaluated using cross validation and root-mean-square-error.

11:45

1aSP12. Measurement matrix design for sparse spatial spectrum estimation in Khatri-Rao subspace. Jiajun Shen and Tian Zhou (Harbin Eng. Univ., Rm. 821, underwater Acoust. Bldg., No. 145, Nantong Ave., Nangang District, Harbin, Heilongjiang 150001, China, shenjiajun@hrbeu.edu.cn)

Compressive (or namely compressed) sensing (CS) exhibits superior performance in sparse spatial spectrum estimation from a predefined vandermonde based dictionary. The CS theory requires that dictionary is as incoherent (orthogonal) as possible since the number of atoms is generally much more than observation vectors, and, namely, the dictionary is over-complete or redundant. Previous researches focus on designing sensing matrix to reduce the mutual coherence of dictionary. However, according to Grassmannian frames, it is still a problem that the coherence of a given dictionary is hard to break through an equiangular tight frame (ETF). To address the problem, we proposed and proved a KR-KSVD method to break through the original lower bound of mutual coherence. In the Khatri-Rao subspace, measurement matrix is designed by minimizing the cost function between the Gram matrix of the equivalent dictionary and an identity matrix with the KSVD method. Simulations demonstrate that the method can produce a better performance in terms of mutual coherence property and sparse recovery accuracy.

Session 1aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Variability in Shallow Water Propagation and Reverberation I

Brian T. Hefner, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

David R. Dall'Osto, Cochair
Acoustics, Applied Physics Laboratory at University of Washington, 1013 N 40th St., Seattle, WA 98105

Invited Papers

8:00

1aUW1. Assessment of the temporal and spatial dependence of reverberation mechanisms for KOREX-17. Dajun Tang, Brian T. Hefner, and Taebo Shim (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Mid-frequency reverberation data were obtained over a 9-day period off Geoje Island, Republic of Korea, complemented by transmission loss and backscatter measurements. The reverberation data were collected using the Autonomous Reverberation Measurement System (ARMS), a benthic lander with a source and receive array mounted on a rotation stage in order to collect reverberation data as a function of bearing angle. The ARMS was repeatedly deployed in roughly the same location over the course of the experiment. The surficial sediments close to the ARMS are known to be composed primarily of mud with occasional rock outcrops. Beyond a range of 2 km, the sediment composition is uncertain and a geoacoustic measurement survey has been planned to collect additional data in this region. Using contemporaneous measurements of the sea surface directional wave spectrum and the currently limited geoacoustic data, modeling is used to estimate the relative importance of sea bottom and surface reverberation. [Work supported by of the Office of Naval Research.]

8:20

1aUW2. Vector intensity properties from direct and reverberant field from a mid-frequency sonar in shallow water. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu) and David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Intensity Vector Autonomous Recorder (IVAR) is a benthic lander system that coherently measures 3-axis acoustic particle velocity and pressure (combined sensor). Results using IVAR in a mid-frequency propagation and reverberation study conducted near Geoje Island, Republic of Korea, are presented. The measurements were made within shallow semi-circular bay (depth ~25 m) with bay opening to deeper waters, and with directional wave measurements made within the bay waters. This presentation will focus on measurements made at a fixed range (100–1000 m, depending on test) from the Autonomous Reverberation Measurement System (ARMS), a benthic lander with a source and receive array mounted on a rotation stage in order to collect reverberation data as a function of bearing angle. Several vector and scaler metrics emerge based on different combinations of second-order acoustic fields are discussed, such as rate of energy transport and active and reactive intensity that lend insight into the process of shallow water reverberation. For example, in one continuous measurement made over a 12 hour period, there is high correlation between wave slope and horizontal beam width as determined with horizontal active intensity. This and other effects related to variability in shallow water reverberation will be discussed.

8:40

1aUW3. Measurements and modeling of mid-frequency propagation loss during the Korea Reverberation Experiment 2017 (KOREX-17). Su-Uk Son (The 6th R&D Inst., Agency for Defense Development, Jinhae P.O. Box 18, Chanwon, Gyeongnam 51678, South Korea, suson@add.re.kr), Hyuckjong Kwon, Jee Woong Choi (Dept. of Marine Sci. and Convergent Technol., College of Sci. and Technol., Hanyang Univ., Sangnok-gu, Ansan-si, Gyeonggi-do, South Korea), Youngnam Na, Joung-Soo Park (The 6th R&D Inst., Agency for Defense Development, Changwon, Gyeongnam, South Korea), and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Sound propagation in shallow water is significantly influenced by fluctuation of the water medium and scattering from rough ocean boundaries. To study the effect of the intensity fluctuations caused by the various environmental effects, mid-frequency propagation loss measurements along with the ocean environmental measurements were conducted on May 25–31, 2017, in shallow water off Geoje island, as part of the Korea Reverberation Experiment 2017 (KOREX-17). Continuous wave and linear frequency modulated signal with a center frequency of 3.5 kHz were transmitted by the Autonomous Reverberation Measurement System (ARMS). The Self Recording Hydrophone (SRH) was towed by the R/V Mirae at a speed of approximately 3 knots along two different tracks. Sound speed profiles were measured using the moored CTD chain near the source location, covering the water column between 3 and 50 m. CTD casts were also conducted at the beginning and end of each track. In this talk, the fluctuations of measured propagation losses were presented for both tracks. Finally, the effects of the sound speed variations on the propagation loss will be discussed in comparison with the model predictions obtained using the measured sound speed profiles.

9:00

1aUW4. Oceanographic effects on mid-frequency acoustics during KOREX-17. Taebo Shim, Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, tbshim@apl.washington.edu), Su-Uk Son (Agency for Defense Development, Chanwon, Gyeongnam, South Korea), Youngnam Na (Agency for Defense Development, Changwon, Gyeongnam, South Korea), and Dajun Tang (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

A 9-day mid-frequency, shallow water experiment was conducted off Geoje Island, Republic of Korea, in May 2017. The experiment consisted of transmission, reverberation, and backscatter measurements. The experiment site includes a shallow bay, with water depth less than 30 m, which opens to the Korea Strait where the depth reaches 60 m at a range of a few kilometers from the bay entrance. While the bathymetry of the site is well documented, the geo-acoustic properties of the area is complex, comprised of mud with rock outcrops and regions of sand. The oceanography during the experiment was dominated by tidal forcing and this is expected to be the main source of temporal variability in propagation and reverberation at the site. This paper focus on understanding the variability of the water column in space and time by analyzing data from CTD casts and from a CTD chain, supplemented by data from a nearby oceanographic buoy. A preliminary assessment of the impact of this variability on transmission loss is also examine. [Work supported by the Office of Naval Research and the Agency for Defense Development.]

9:15

1aUW5. Geoacoustic inversion of 3.5-kHz towed receiver data from the KOREX-17. Hyuckjong Kwon, Jee Woong Choi (Dept. of Marine Sci. and Convergent Technol., College of Sci. and Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan-si, Gyeonggi-do 15588, South Korea, hjkwon1014@gmail.com), Su-Uk Son (Agency for Defense Development, Chanwon, Gyeongnam, South Korea), Youngnam Na (Agency for Defense Development, Changwon, Gyeongnam, South Korea), Seom-Kyu Jung (Korea Inst. of Ocean Sci. & Technol., Busan, South Korea), Dajun Tang, and Taebo Shim (Univ. of Washington, Seattle, WA)

The Korea Reverberation Experiment 2017 (KOREX-17) was conducted in a shallow water located in the south of the Geoje island, Korea. During the experiment, sound propagation measurements were made using a 3.5-kHz CW signal along different 2 tracks to a distance of ~10 km from a bottom-mounted source, ARMS (Autonomous Reverberation Measurement System). The signals were received by a SRH (Self Recording Hydrophone), which was towed at a depth of ~20 m during the measurements. The sound speed profile was almost iso-velocity, and the sediment at the site was mainly composed of silt having a mean grain size of 6 phi. Since the sound propagation in shallow water is greatly influenced by sound interaction with the sediment, geoacoustic inversion was tried using a genetic algorithm based matched field processing in which the measured acoustic pressure field was compared to the simulated field predicted by a parabolic-equation based propagation model (RAM). The results are compared to the geoacoustic parameters obtained by the empirical relationship to mean grain size and a sediment layering information obtained by a chirp sonar survey. [Work supported by Agency for Defense Development, Korea (UD170014DD).]

9:30

1aUW6. Measurements of mid-frequency bottom backscattering during KOREX-17. Dong-Gyun Han, Raegeun Oh, Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., Ansan 15588, South Korea, dghandg@hanyang.ac.kr), Dajun Tang, and Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Measurements of mid-frequency bottom backscattering strength were made on May 30, 2017, as part of the Korea Reverberation Experiment (KOREX-17) in a shallow water region located at the south coast of Korea. The acoustic data were transmitted and received as functions of frequency (4, 6, and 8 kHz) and pulse length (2 and 4 ms) using semi-monostatic sonar system composed of omni-directional acoustic source and receiver, which was deployed from the *R/V* Mirae. The bottom backscattering strength were extracted from intensity-averaged reverberation level for 30 individual pings. The water depth measured by an echo sounder was about 35 m and bottom was approximately flat, having a slope less than 1° over the experimental area. The surficial sediment was estimated to be silty sediment, having a mean grain size of 6 phi from grain size analysis. In this talk, the measurement results of bottom backscattering strengths will be presented as a function of grazing angle and compared to the predictions obtained by Lambert's law and APL-UW scattering model. [Work supported by Agency for Defense Development, Korea(UD160006DD).]

9:45

1aUW7. Rocky outcrops as clutter in mid-frequency reverberation measurements. Brian T. Hefner, Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., Ansan, South Korea), and Taebo Shim (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Scattering by isolated rock outcrops stand out as prominent features in mid-frequency reverberation measured during KOREX-17, a shallow water experiment conducted off Geoje Island, Republic of Korea, in May 2017. The reverberation data were collected using the Autonomous Reverberation Measurement System (ARMS), a benthic lander with a directional source and receive array mounted on a rotation stage. This stationary system was deployed in roughly the same position on the seafloor over the course of the 9-day experiment. A side-scan sonar survey of the seafloor was conducted to identify the location and rough spatial extend of the exposed portions of the rock outcrops. This talk examines scattering by the rock outcrops as a function of time, signal waveform, and changing oceanographic conditions, with an eye toward detection in clutter environments. Since the ARMS is a fully-calibrated sonar system, with known source level and directivities, the target strengths of the outcrops are also estimated with the long-term goal of developing a scattering model of the most prominent rock outcrop which extends 7-8 m above the seafloor. [Work supported by ONR.]

10:00–10:15 Break

10:15

1aUW8. Analysis of sonar clutter using backscattered spatial correlation. Chad M. Smith, Daniel C. Brown, and John R. Preston (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu)

Sonar clutter events are one of the primary limitations of long-range active sonar systems within the littoral environment. For this reason, research towards understanding their acoustic return characteristics and ultimately their mitigation are of high importance. This paper discusses measurements and modeling of the backscattered horizontal correlation function from clutter events and comparison with returns from the littoral environment. A statistical model for returns from boundaries, as well as general expectations of returns from common clutter sources will be discussed. The horizontal correlation function is estimated via sub-aperture beamforming of the Five Octave Research Array (FORA), a towed line array system owned and operated by Penn State and funded by the Office of Naval Research. [Work supported by Office of Naval Research.]

10:30

1aUW9. Variability in surface reverberation observed with a high-frequency obstacle avoidance sonar. Mark V. Trevorrow (DRDC Atlantic, 9 Grove St., PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, mark.trevorrow@drdc-rddc.gc.ca)

Sea-surface reverberation and, in particular, near-surface bubble backscatter, can pose a strong limitation on HF sonar performance. Recent high-frequency obstacle-avoidance sonar measurements showed significant variability in low grazing angle sea-surface reverberation. The sonar used in these sea-trials operated at 90kHz covering a 90-degree angular sector with 128 beams at ranges up to 600 m. It was projected horizontally forward from a ship at a depth of 3.5 m in deep water. The statistics of the background reverberation under Sea-State 3 to 4 conditions were investigated. The time- and spatially averaged background reverberation levels agreed with well-known APL-UW models. However, the instantaneous reverberation amplitudes exhibited non-Rayleigh statistical distributions, in better agreement with log-normal distributions. This variability was attributed to scattering patchiness and surface wave effects.

10:45

1aUW10. Dynamic imaging of a gravity wave caused by laser-induced breakdown in a fluid waveguide using multi-reverberated ultrasonic waves. Tobias van Baarsel, Philippe Roux (Université Grenoble-Alpes, Université Grenoble Alpes ISTerre, Grenoble 38000, France, tobias.van-baarsel@univ-grenoble-alpes.fr), Barbara Nicolas (Créatis, Villeurbanne Cedex, France), JEROME I. MARS (GIPSA-Lab, Grenoble, France), Michel Arrigoni (ENSTA, Brest, France), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), and Steven Kerampran (ENSTA, Brest, France)

The dynamic imaging of a gravity wave propagating at the air-water interface is a complex task that requires the sampling of every point at this interface during the gravity wave propagation. Using two source-receiver vertical arrays facing each other in a shallow water environment, we manage to isolate and identify each multi-reverberated eigenbeam that interacts with the air-water interface. The travel-time and amplitude variations of each eigenbeam are then measured during the crossing of the gravity wave. In this work, we present an ultrasonic experiment in a 1 m-long, 5 cm-deep waveguide at the laboratory scale. The waveguide transfer matrix is recorded 100 times per second at a sample rate of 1.1 MHz between two source-receiver arrays while a low-amplitude gravity wave is generated by a laser-induced breakdown at the middle of the waveguide above the water surface. The controlled and therefore repeatable breakdown causes a blast wave that interacts with the air-water interface and penetrates into the water, creating ripples at the surface that propagate in both directions. The surface deformation induced by these two wave packets is also measured by two cameras which allows for independent validation of the ultrasonic inversion. The ultrasonic inversion performed from a few thousand eigenbeams lead to accurate quantitative imaging of the dynamic of the air-water interface, using either the travel-time or the amplitude variation of the ultrasonic arrivals.

11:00

1aUW11. Validity of the frozen-surface approximation for large time-bandwidth signals. Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, phines50@gmail.com), Douglas A. Abraham (CausaSci LLC, Ellicott City, MD), Stefan Murphy (DRDC, Dartmouth, NS, Canada), and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., Portland, OR)

The physical modeling of underwater acoustic propagation, scattering, and reflection, and the signal processing associated with these processes, usually rely on the frozen-surface approximation. This longstanding approximation, which dates back to Carl Eckart's seminal paper (The Scattering of Sound from the Sea Surface, 1953) assumes that the ocean surface can be modeled as being frozen in time throughout the entire pulse duration. The approximation is valid for short-duration pulses typically used by sonars in the decades following that paper; however, its applicability to present-day high duty cycle sonars is questionable. Although the assumption is based on the physics of the problem, it can have a profound effect on the signal processing, especially for large time-bandwidth signals. In this paper some experimental results will be presented to provide examples of how the approximation fails for a long duration, wideband pulse. Using this as motivation, the authors will explore some areas where incorrectly employing the approximation can introduce errors in the expected signal processing gain. While no attempt will be made to correct shortcomings in the approximation, it is hoped that the discussion may motivate renewed interest in this issue. Funding from ONR and ONR Global is gratefully acknowledged.

11:15

1aUW12. Modeling acoustic interactions with a dynamic rough sea surface boundary. Alex Higgins, Martin Siderius (Elec. & Comput. Eng., Portland State Univ., 1900 SW 14th Ave., Ste. 25-01, Portland, OR 97201, higginja@ece.pdx.edu), Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada), and Douglas Abraham (CausaSci LLC, Ellicott City, MD)

The acoustic wave scattering properties of a dynamic pressure-release surface boundary are analyzed using a numeric technique based on the finite-difference time-domain (FDTD) method. Of primary interest is to study the impact of assuming a "frozen" sea-surface on long duration sonar transmissions. Although relatively uncommon in ocean acoustics, the FDTD approach is well suited for modeling boundary roughness and motion. This technique has the additional benefit that the pressure fields are resolved over time, which allows for transient analysis of the observed wave scattering effects. The method is adapted from electromagnetic wave scattering and can properly model the physics of the observed system, which shows a frequency modulated reflection that includes a double-Doppler effect. First, a traditional analytic solution for the static smooth surface boundary ocean half-space model, the Lloyd Mirror, is compared to an equivalent simulation using the FDTD method. Then a dynamic smooth surface boundary is investigated using a modified Lloyd-mirror solution and the FDTD method. Finally, surface roughness for both static and dynamic boundary cases will be considered. Agreement is shown between FDTD simulations and the modified Lloyd Mirror model for both one-dimension and two-dimension cases. [Work supported by the Office of Naval Research.]

11:30

1aUW13. Examination of extensions to transport theory that includes rough surface scattering. Eric I. Thorsos and Frank Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

Sea surface forward scattering has important effects on shallow water propagation and reverberation at mid frequencies (i.e., 1–3 kHz) under typical sea surface roughness conditions. Coupled-mode or rough surface PE modeling of these effects require averaging results over many rough surface realizations, increasing the computational effort. An alternative method is based on transport theory, where equations are developed for propagating the moments of the field, avoiding the need for utilizing rough surface realizations. Our transport theory method is based on expanding the field in unperturbed modes, and the equations of motion are for moments of the mode amplitudes. The approach has been based on keeping terms to only first-order in the surface height h(x), making the method linear in surface height. Methods for extending the approach beyond the linear model will be described, both with using realizations with coupled modes and with attempts to extend these approaches to obtain a transport theory. A key part of the approach is the use of the Differential Algebraic Equation (DAE) method in which the range-derivative of the effective boundary condition on the mean plane of the rough surface is used instead of the effective boundary condition itself. [Work supported by ONR Ocean Acoustics.]

11:45

1aUW14. Statistical optimal filtering method for acoustical signal deblurring. Viktoria Taroudaki (Dept. of Mathematics, Eastern Washington Univ., 316 Kingston Hall, Cheney, WA 99004, vtaroudaki@ewu.edu), Michael Taroudakis, and Costas Smaragdakis (Mathematics and Appl. Mathematics and IACM, Univ. of Crete and FORTH, Heraklion, Greece)

Acoustical signals in applications of acoustical oceanography, such as ocean acoustic tomography and sea-bed classification using acoustic signals emitted from known sources, are optimally exploited if they are noise free. The effect of blur in acoustic signals has not been well studied, although the blurring mechanism might introduce severe problems in the use of the acoustic signals for specific applications, especially those using the full signal as the carrier of the relevant information. Deblurring of the signals in addition to denoising is therefore essential for the effective use of the signals. In our work, we apply a Statistical Optimal Filtering method that uses the singular value decomposition of a first estimate of the blurring matrix and statistics to deblur the signal in an efficient and effective way and to quantify uncertainty for the recovered signal. In this talk, we will present the method and discuss its effectiveness using as test case, an application of sea-bed classification based on a statistical characterization of an acoustic signal. The statistical characterization is particularly sensitive to noise and blur contamination of the exploitable signal and any attempt for getting a signal clear from noise and blur is absolutely necessary, for obtaining reliable results.

MONDAY AFTERNOON, 5 NOVEMBER 2018 THEATER (VCC), 1:00 P.M. TO 4:20 P.M.

Session 1pAA

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. Implementing acoustical designs—Non-acoustical project choices may help or hinder successful outcomes. Damian Doria (Stages Consultants LLC, 75 Feather Ln, Guilford, CT 06437-4907, damianjdoria@gmail.com) and Robert Essert (Sound Space Vision, London, United Kingdom)

The authors contributed acoustical design for many auditoria over the last thirty years. Many of those designs incorporated novel and sometimes unconventional approaches to achieving room acoustics goals. Aside from technical demands, implementing new acoustical ideas requires navigating a number of other challenges from discovering and coordinating the needs of project stakeholders through oversight of the construction and commissioning process. Consultants are called to serve as counselors in addition to our role as technical experts and designers. Much depends on the Owner's aspirations, leadership and structure, and on procurement and delivery approaches. A number of case-examples will be presented along with discussion of the sometimes unexpected challenges faced by project teams in successful realization of their designs.

1:25

1pAA2. Case study: Acoustic tuning of the new DePaul University School of Music Concert Hall. Brian Corry (Kirkegaard Assoc., 7733 Forsyth Boulevard, Ste. 1100, St. Louis, MO 63105, bcorry@kirkegaard.com) and Ian Hough (Kirkegaard Assoc., Chicago, IL)

The Holtschneider Performance Center in Chicago packs an enormous program onto a limited urban site. The largest of the four performance halls in this new home for DePaul University's School of Music is the 535-seat Concert Hall. In response to a limited footprint and a broad program of performance types, Kirkegaard devised a simple but innovative acoustical approach to the room. A single massive plaster cloud hovers in the center of the hall, supporting onstage communication and providing first reflections to the audience. The large gap between the cloud and the perimeter walls integrates the extensive volume above the ceiling with the main volume of the hall. Extensive banners above the cloud are intended to "turn off" the upper volume, effectively converting the hall from a concert hall to a

recital hall. More conventional curtains and banners below the cloud allow further fine tuning of the room's acoustics. The Concert Hall is well isolated acoustically from the rest of the 3 storey music building and exterior city noise by 18" thick concrete walls, concrete roof, and a continuous structural acoustic isolation joint.

1:45

1pAA3. Recent acoustic designs in China, challenges, and design approach. Thomas Scelo, Peter Fearnside, and Peter Exton (Performing Arts, Marshall Day Acoust., 1601, 16/F The Hollywood Ctr., 233 Hollywood Rd., Sheung Wan 0000, Hong Kong, tscelo@marshallday.com)

Marshall Day Acoustics has now completed and commissioned sixteen performing arts venues in China and has another sixteen in design or in construction. It seems an opportune time to reflect on the latest challenges, technically or otherwise and how we have successfully approached them. Projects have become bigger and more complex, yet clients' expectations and requirement for certainty in the outcome have increased. The paper will include three recent examples. The 1,500-seat concert hall and 1,000-seat drama theatre of the Jiangsu Grand Theatre both required on-site design to ensure acoustic requirements were met, in time. The 2,000-seat Qingdao Grand Theatre achieved a very low reverberation time suitable for the use of multiple complex sound systems while maintaining the hard finish look desired by the client. This required both careful predictions and laboratory testing. Finally, the 2,040-seat opera house at Shaanxi Grand Theatre that was delivered in less than three years. Atypical materials and simple design principles were implemented to meet the project timeframe. These cases are examples of successful acoustic engineering and what can be achieved when applying first principles and sound science, backed by simulations, testing and experience.

2:05

1pAA4. Recent acoustic designs in China, commissioning results. Thomas Scelo, Peter Fearnside, and Peter Exton (Marshall Day Acoust., 1601, 16/F The Hollywood Ctr., 233 Hollywood Rd., Sheung Wan 0000, Hong Kong, tscelo@marshallday.com)

Marshall Day Acoustics has recently completed three large performing arts centres in China. Some of the challenges and design approaches are presented in an accompanying paper. These include the Jiangsu Grand Theatre with a 1,500-seat concert hall, 2,300-seat opera house and a 1,000-seat drama theatre. Also included is the 2,000-seat Qingdao Grand Theatre and the 2,040-seat opera house at Shaanxi Grand Theatre. The present paper will provide details on the commissioning measurement results for room acoustics. First the principles of the measurement system and room setup will be briefly presented before the actual measurement data and derived acoustic parameter are revealed.

2:25

1pAA5. Scale model test for the acoustical design of Arts Center Incheon concert hall and acoustic measurement after construction. Kee Hyun Kwak, Hyung Suk Jang, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, bigdook@hanmail.net)

The Arts Center Incheon (ACI) concert hall was designed to have 1,760 seats, the volume of 18,000 m^3 and the reverberation time of 2.1s with seats fully occupied. To achieve the auditory and visual intimacy, the concert hall had a reverse fan shape with a combination of vineyard and balcony. Various scale models and computer simulation were used to determine the acoustical design including the volume of hall, locations of walls and shapes of finishing materials. Three types of sound diffusers were applied according to wall locations. Appropriate protrusion heights were determined based on sound diffusion targets by each part such as the stage and the sides and back of the auditorium. During the execution design stage, a 1:10 scale model of this concert hall was fabricated to measure acoustic characteristics. An auralization experiment was conducted to assess psycho-acoustic responses to the diffusion design. The acoustic measurements after construction matched the design objectives.

2:45

1pAA6. Sound diffusion design process using scale models of a concert hall and acoustic parameters. Hyun In Jo, Hyung Suk Jang, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., Hanyang University, Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com)

This study proposed a design process for diffusion walls of the Art Center Incheon concert hall by using scale models and verified the sound diffusion performance in comparison with acoustic parameters. Locations of diffusers and main diffusing surfaces were determined in a 1:50 scale model, and the heights of diffusers in the stage were set and diffusion rates were evaluated according to occupancy density in a 1:25 scale model. A 1:10 scale model enabled the real acoustic characteristics of the concert hall to be evaluated. This model was used to measure scattering/diffusion coefficients of diffusers in each part and to design the finishing shapes of walls. Horizontal diffusers with the highest protrusion were installed on the lateral walls near the stage in order to orientate scattered reflections. Impulse responses were used to investigate acoustic parameters like RT, EDT, G, and C80, and the number of peak reflection (Np) was calculated to compare diffusion performance. It turned out that, when the amount of diffusion changed, the value of Np increased but the values of RT, EDT, and G and the corresponding relative standard deviation (RSD) decreased. The ratio of 1:25 or higher scale models were efficient in designing the diffusion surfaces of the concert hall. The effectiveness of scale models was verified as the directions, locations, protrusion heights, and areas of diffusers were determined.

3:05–3:20 Break

3:20–4:20 Panel Discussion

Session 1pAB

Animal Bioacoustics: Fish and Marine Invertebrate Bioacoustics II

Bruce Martin, Cochair
JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada

Xavier Mouy, Cochair
JASCO Applied Sciences, 2305–4464 Markham Street, Victoria, BC V8Z7X8, Canada

Invited Papers

1:00

1pAB1. Underwater ecoacoustics as a monitoring tool in freshwater environments. Camille Desjonquères (Univ. of Wisconsin Milwaukee, 3209 N Maryland Ave., Milwaukee, WI 53212, desjonqu@uwm.edu), Fanny Rybak (Université Paris-Sud, Orsay, France), Toby Gifford (SensiLab, Caulfield, VIC, Australia), Simon Linke (Australian Rivers Inst., Griffith Univ., Nathan, QLD, Australia), and Jérôme Sueur (Museum national d'Histoire naturelle, Paris, France)

Biodiversity in freshwater habitats is decreasing faster than in any other environment, mostly due to human activities. Monitoring these losses can help guide mitigation efforts. In most comparative or focal studies, sampling strategies predominantly rely on collecting animal and vegetal specimens. Although these techniques have produced valuable data, they are invasive, time-consuming, and typically have limited spatial and temporal replication. There is therefore a need for the development of complementary methods. As with other ecosystems and landscapes, freshwater environments host animals producing sounds, either to communicate or as a byproduct of their life activity. Animals and processes can be recorded, remotely, by unattended equipment and provide global information on local diversity and ecosystem health. We review practical examples of progress in experimentally addressing six main challenges that freshwater ecoacoustic monitoring faces: (1) associating each sound to its emitter, (2) estimating intra-specific sound variations, (3) evaluating diurnal variation, (4) modeling sound propagation, (5) deriving links between ecological condition and sounds, and (6) developing a repository for freshwater sounds. Passive acoustics represents a potentially revolutionary development in freshwater ecology, enabling dynamic monitoring of biophysical processes to inform conservation practitioners and managers.

1:20

1pAB2. Glass sponge reef soundscapes. Stephanie K. Archer (Pacific Biological Station, Fisheries and Oceans Canada, Pacific Biological Station, 3190 Hammond Bay, Nanaimo, BC V9T 6N7, Canada, Stephanie.Archer@dfo-mpo.gc.ca), William D. Halliday (Wildlife Conservation Society, Whitehorse, YT, Canada), Amalis Riera (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada), Xavier Mouy (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Matthew Pine (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada), Anya Dunham (Pacific Biological Station, Fisheries and Oceans Canada, Nanaimo, BC, Canada), and Francis Juanes (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada)

Many structured biogenic habitats are biodiversity hotspots and thus possess unique soundscapes largely driven by the biophony. To date, the vast majority of research has focused on shallow-water biogenic habitats such as coral or oyster reefs. Glass sponge reefs are a deep-water habitat analogous in many ways to shallow-water coral reefs. These reefs are built by three species of hexactinellid sponges which form complex 3-dimensional habitats that support diverse communities of animals. Many soniferous animals, including rockfish, are strongly associated with patches of live-sponge dominated habitat within the reef foot-print. Consequently, we hypothesized that glass sponge reefs and the communities they support would generate unique soundscapes. Beginning in September of 2016, we deployed a series of underwater acoustic recorders on sponge reefs throughout Canada's Pacific continental shelf. Initial results show that recorders located on sponge reefs were significantly louder in the mid- and high-frequency bands (100–1000 Hz and 1–10 kHz, respectively). Additionally, many fish calls were detected in recordings from within sponge reefs, while few fish calls were observed at similar depths in off-reef habitats. We will discuss our understanding of the link between soundscapes and biodiversity in glass sponge reef habitats and the potential application of ecosystem-level monitoring.

1:40

1pAB3. Spatial mapping of the biophony of the fishes living in seagrass meadows. Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France), Cathy Anna Valentini Poirier, and Pierre Boissery (Agence de l'Eau RMC, Marseille, France)

Passive acoustics is well suited to assess the diversity and/or activity of marine animals, particularly if cryptic or difficult to observe as in seagrass *Posidonia oceanica* meadows. The ability to locate the emitters allows not only to detect the presence of a specific sound or specie but also to estimate source levels, the communicative space with respect to ambient noise and anthropogenic impact, the

density of vocalizing animals, their movements and potentially home ranges, etc. Here, we located fishes in seagrass meadows (Calvi, France). Two particular sounds dominated the acoustic scene of seagrass meadows: the: 300 Hz drums of *Ophidion rochei* and the ubiquitous 800Hz /*kwa*/ sound of unknown origin. The wavelengths of these fish sounds range from 5 m to 2 m. During six 48-hours period, we deployed a squared array of 4 hydrophones (20 x 50 x 50 m). Here, we describe the algorithm for automated fish sound detection and 3D localization of the sources and assess their performances on our real data. Over 1000 fish sounds were detected per night. These detections were used to create localization maps and study the spatial distribution of fish sound production, estimate source levels and the density of the vocalizing fishes within the meadow and their limits.

Contributed Papers

2:00

1pAB4. Searching for the FishOASIS: Using passive acoustics and optical imaging to identify a chorusing species of fish. Camille Pagniello, Jack Butler, Gerald L. D'Spain, Jules Jaffe, Ed Parnell, and Ana Širović (Scripps Inst. of Oceanogr., Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr. #0205, La Jolla, CA 92093-0205, cpagniello@ucsd.edu)

Marine protected areas have been established off the California coast to ensure the persistence and resiliency of the marine ecosystems found here. Kelp forest habitats, in particular, support a diverse assemblage of fishes, many of which produce sound. From May to September 2017, a low-frequency (325–545 Hz) chorus was recorded near the kelp forests off La Jolla, California. The chorus begins each day approximately a half-hour before sunset and lasts for about 3–4 hours. During these times, spectral levels around 400 Hz increased by approximately 30 dB over, although there is significant day-to-day variability in received level. To identify the chorusing fish species, a Fish Optical and passive Acoustic Sensor Identification System (FishOASIS) was developed, consisting of a four-channel SoundTrap ST4300 acoustic recorder and four Sony α7s II cameras. Frequency-domain beamforming was used on signals recorded by the four-element, 20-m aperture, tetrahedral-shaped array to estimate the location of the fish chorus, which appears to be fairly fixed over time. The chorus also was used as a source of opportunity to measure transmission loss in order to determine whether kelp forests can act as acoustic refuges by sufficiently attenuating chorusing sounds. [Research supported by California Sea Grant and NSERC Postgraduate Scholarship-Doctoral.]

2:15

1pAB5. Benthic biophonic assemblages, their environmental divers, eco-acoustic scores at the level of the Western Mediterranean basin, and their implications for large-scale ecosystem monitoring. Lucia D. Iorio, Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Cathy Anna Valentini Poirier, and Pierre Boissery (Agence de l'Eau RMC, Marseille, France)

Benthic invertebrate assemblages are known to produce an important biophony composed of short transient sounds emitted while hunting, feeding, moving, for territorial defense, etc. Although they exhibit dial variations, these sounds are present year-round, 24 hours a day and have the potential to provide information on the habitat and organism-environment relationships. Here, we describe benthic invertebrate sounds (BIS) of two key Mediterranean habitats, coralligenous reefs and seagrass meadows. Then, we assess the environmental drivers of BIS variability for each habitat. 129 400 000 BIS sampled in the summers of 2015–2017 from 135 recording sites over more than 1000 km coastline were used for this study. Each sound was characterized by 28 acoustic features. Principal Component Analyses were performed to identify the most contributing features in terms of diversity and intensity. To quantify and characterize invertebrate assemblages at each site, we defined an eco-acoustic score based on BIS abundance, diversity, and intensity. These scored acoustic assemblages were then tested for environmental drivers such as site, biocenosis, depth, ecosystem health, anthropogenic pressures, etc. We hereby provide an exhaustive ocean basin-wide picture of the benthic biophony and discuss its implications as environmental proxies.

2:30–2:45 Break

2:45

1pAB6. Using passive acoustics to localize vocalizing oyster toadfish (*Opsanus tau*). Rosalyn Putland, Alayna Mackiewicz, and Allen F. Mensinger (Dept. of Biology, Univ. of Minnesota Duluth, 1035 Kirby Dr., Duluth, MN 55812, rputland@d.umn.edu)

Identifying where fish inhabit is a fundamentally important topic in ecology and acoustic tools can help management to prioritize acoustically sensitive times and areas. In this study, passive acoustic monitoring is presented as a viable tool for monitoring the positions of vocalizing fish species, like the oyster toadfish. Time of arrival differences (TOADs) of sound recordings on a four-hydrophone array were used to pinpoint the location of male oyster toadfish, *Opsanus tau,* a sedentary fish that produces boatwhistle vocalizations to attract females. Coupling the TOAD method with cross correlation of the different boatwhistles, individual toadfish were mapped during three-hour periods at dawn, midday, dusk, and midnight to examine the relationship between temporal and spatial trends. Seven individual males were identified within 24.2 m of the hydrophone array and up to 18.2 m of the other individuals. The advantages and disadvantages of using the TOAD method to localize individual fish will be discussed. Additionally, preliminary data on how individual toadfish respond to the anthropogenic sound of passing motorized vessels as well as conspecific boatwhistles will be introduced.

3:00

1pAB7. Automatic detection and localization of croaker's fish calls using beamforming. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@mail.tohoku-gakuin.ac.jp), Kazuki Yamato (Gunma Univ., Sendai, Japan), Ryuzo Takahashi, Tomohito Imaizumi (NRIFE, Fisheries Res. and Education Agency, Kamisu-shi, Ibaraki-ken, Japan), and Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan)

Many kinds of fish, including croaker, produce species-specific low-frequency sounds associated with courtship and spawning. A recording system was used to monitor underwater fish call sounds. We have proposed the method to detect croaker calls from data measured by a single hydrophone. However, it was difficult to detect the desired calls at a high detection rate because of a low signal-to-noise ratio. We proposed the method using beamforming to improve the detection rate. At first step, fish calls are detected from data measured on one hydrophone using the previous method, which detects calls automatically using the acoustic features, that is, duration and periodicity. At second step, additional calls are detected by beamforming the four-channel data. At third step, detected calls are localized by using the time differences of arrivals. It was clarified that the detection rate using the proposed beamforming method was higher than that under the previous method with a single data channel. In addition, it was shown that fish calls could be localized from the acoustic data measured during several weeks. Therefore, this method could monitor sounds from croaker in the ocean.

3:15

1pAB8. Complexity-entropy based approach for detection of fish choruses. Shashidhar Siddagangaiah (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taipei 106, Taiwan, shashi.18j@gmail.com)

Increasing anthropogenic noise around the world ocean are affecting the marine ecology. Recently, acoustic indices (AI) were utilized to quantify

the biophony in the marine soundscape. However, these AI's employed in complex marine environment, dominated by several anthropogenic and geophonic sources are yet to be understood. In this study, we have introduced a method based on complexity-entropy (*C-H*) for detection of biophonic sounds originating from fish chorus. The fish chorus detection performance of *C-H* was compared with AI's such as acoustic complexity index (ACI), acoustic diversity index (ADI), and bioacoustics index (BI). We have utilized the data collected at Changhua (A1) and Miaoli (N1). During the Spring of 2016 and 2017, the region N1 was exposed to continual shipping activities, due to which there was ~10 dB increase in the low frequency (5–500 Hz) noise levels. This enabled us to evaluate the fish chorus detection performance of various AI's and *C-H* method, and the robustness in the presence and absence of shipping activities. The results presented in this study shows that, during the fish chorusing hours, the introduced entropy is positively correlated with Pearson's correlation coefficient (P_{cc}) > 0.95 and complexity is anticorrelated with P_{cc} < -0.95. Therefore, the introduced *C-H* method has potential implication in efficient detection of fish chorus and overcome the limitations confronted by AI's such as ACI, ADI, and BI.

3:30

1pAB9. Sounds from the Amazon: Piranha and prey. Rodney A. Rountree (The Fish Listener, 23 Joshua Ln., Waquoit, MA 02536, rrountree@fishecology.org) and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

The underwater soundscape of an upper tributary of the Amazon River was studied in a four-week survey from 3 to 26 July 2012 within the Pacaya-Samiria National Reserve, Peru. Over 550 individuals representing over 70 species of fishes were auditioned for sound production. In addition, over 641 minutes of natural sounds from the river were recorded from 22 sites. We demonstrate that closely related species of piranha can be distinguished by their hand-held disturbance sounds. Similar piranha sounds were recorded in the wild at locations where piranha were known to be actively feeding. Sounds of catfishes and other fishes were significantly more frequent at piranha feeding sites. Thus, piranha sounds appear to be excellent indicators of local piranha feeding activity and suggest that passive acoustic monitoring (PAM) can be an effective tool for studies on piranha behavior, feeding activity, and impact on prey fishes.

3:45

1pAB10. Buzzing sounds used as a mean of intra-specific interaction during agonistic encounters in male European lobsters (*Homarus gammarus*)? Youenn Jézéquel (Laboratoire des Sci. de l'Environnement Marin, UBO, CNRS, IRD, Ifremer, LIA BeBEST, UMR 6539, IUEM, 12 Rte. de penhuel, Plouzané 29280, France, youenn.jezequel@univ-brest.fr), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Jennifer Coston-Guarini (Laboratoire des Sci. de l'Environnement Marin, UBO, CNRS, IRD, Ifremer, LIA BeBEST, UMR 6539, IUEM, Plouzané, France), Jean-Marc Guarini (UPMC (Paris-6), UMR 8222 LECOB, Observatoire Océanologique de Banyuls sur Mer, UPMC, Paris, France), and Laurent Chauvaud (Laboratoire des Sci. de l'Environnement Marin, UBO, CNRS, IRD, Ifremer, LIA BeBEST, UMR 6539, IUEM, Plouzané, France)

Passive acoustics is a useful non-invasive tool to collect behavioral information in marine species. This is the case for temperate crustaceans which are known to emit a large variety of sounds through diverse mechanisms. But despite numerous studies in tanks, little is known about their ecological meaning, particularly for decapods of high commercial interest. When stressed by handling, the European lobster (*Homarus gammarus*) vibrates its carapace and produces low frequency "buzzing sounds" that can be characterized in tanks. In this presentation, we discuss a straightforward experimental approach to investigating the role of these buzzing sounds in male European lobsters by combining passive acoustics and behavioral analysis (video and accelerometry). We recorded sound and video simultaneously during agonistic encounters. Based on the video, an ethogram was created with a total of 30 behaviors regrouped by agonistic levels. During agonistic encounters, European lobsters emitted buzzing sounds in association with stressful events such as claw grasping or tail flipping. Our results suggest that these sounds may be used by *H. gammarus* to maintain dominance around its shelter.

4:00

1pAB11. Describing new fish sounds from the northeast Pacific: A quantitative approach. Amalis Riera (Biology, Univ. of Victoria, School of EOS, Bob Wright Ctr. A405, UVic, Victoria, BC V8P5C2, Canada, ariera@uvic.ca), Rodney A. Rountree (none, Waquoit, MA), and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

In order to identify fish sounds in British Columbia soundscapes, we need catalogues of validated fish sounds from Pacific species. These catalogues will help in comparing validated examples to unknown sounds found in long-term autonomous recordings. In addition, it is important that the sounds be described quantitatively for accurate identification and comparison. Our goal is to contribute to building such catalogues and to fill knowledge gaps in fish acoustic behaviour to support studies on the impact of anthropogenic noise on Pacific fishes. Since distinguishing sound sources in the ocean is extremely difficult, we are collaborating with aquaria, marine science centers and ocean-based aquaculture facilities to monitor, audition and record the acoustic behaviour of captive and semi-captive fish species. To date, we have acquired over 3,000 hours of recordings in these conditions. Sounds that we have already validated include Arctic cod (*Boreogadus saida*), walleye pollock (*Gadus chalcogrammus*), and grunt sculpin (*Rhamphocottus richardsonii*) vocalizations. Here, we will present a description of these newly validated sounds.

4:15

1pAB12. Acoustic recordings of Pacific salmon (*Oncorhynchus* spp.) from a hatchery on Vancouver Island. Kelsie Murchy (Biology, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3N5, Canada, kmurchy@uvic.ca), Xavier Mouy (School of Earth and Oceans Sci., Univ. of Victoria, Victoria, BC, Canada), and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

Sound production in fish has been documented; however, the diversity of species that create sound is not fully understood. Pacific salmon (*Oncorhynchus spp.*) are ecologically, economically, and culturally important in the northeast Pacific. Recent declines in specific species and stocks have increased their public interest. Other species from the family Salmonidae produce sounds, but there is little evidence of any species of Pacific salmon producing sounds. Recording salmon in the wild would be difficult but local hatcheries allow for a unique opportunity to listen for salmon sounds in a semi-natural environment. Chinook salmon (*O. tshawytscha*), pink salmon (*O. gorbuscha*), and coho salmon (*O. kisutch*) were recorded using two stationary acoustic recorders that were deployed at a salmon hatchery in Qualicum beach Vancouver Island, British Columbia, for three consecutive weeks in September and October 2017. Audio files were collected in 5 minute subsections and examined for salmon sounds. Here, we present spectrograms and time-frequency composition of potential sounds found for Chinook, pink, and coho salmon. All sounds were then compared to sounds produced by other soniferous fish and to the hearing range of Pacific salmon.

Session 1pAO

Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Arctic Acoustical Oceanography II

Peter F. Worcester, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225

Hanne Sagen, Cochair
Polar Acoustic and Oceanography Group, Nansen Environmental and Remote Sensing Center, Thormøhlensgt 47, Bergen 5006, Norway

Mohsen Badiey, Cochair
University of Delaware, Newark, DE 19716

Contributed Papers

1:00

1pAO1. Arctic ambient noise statistics at the edge of the Canada Basin. Christopher Whitt, Bruce Martin (JASCO Appl. Sci., 202-32 Troop Ave., Dartmouth, NS B3B1Z1, Canada, christopher.whitt@jasco.com), Sean Pecknold (DRDC Atlantic, Dartmouth, NS, Canada), Mohsen Badiey (Univ. of Delaware, Newark, DE), and Aidan Cole (JASCO Appl. Sci., Halifax, NS, Canada)

Knowledge of ambient noise is important for the design and operation of acoustic observation and communication systems. Several concurrent year-long acoustic datasets from the Canada Basin Acoustic Propagation Experiment (CANAPE) were collected in 2016–2017. Selected data were analyzed to investigate soundscape temporal and spatial characteristics in an area on the western slope of the Chukchi shelf and Canada Basin, north of Barrow Alaska. From October 2016 to October 2017, four 8-element vertical arrays were deployed in water depths between 100 and 300 m. Each array recorded at approximately 15% duty cycle at several sample rates between 4000 Hz and 64000 Hz and at 24-bit resolution. One-minute, 10-second, and 1-second root-mean-square sound pressure levels were computed to generate ambient noise statistics and summarized in empirical probability density functions (PDF) for various bands. Fit functions for ambient noise were determined for each empirical PDF. Impulse detection was used to investigate potential correlation of ice cracking with ambient noise. A multivariate correlation of ambient noise levels was performed with several environmental parameter covariates, including ice cover, wind speed, and air temperature. These correlations may form the basis for predictive models for ambient noise modeling in the arctic.

1:15

1pAO2. Results from the "Arctic ocean under melting ice" acoustic thermometry experiment. Espen Storheim, Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Thormøhlensgate 47, Bergen 5006, Norway, Espen.storheim@nersc.no), Matthew Dzieciuch, Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA), Eva Falck (Geophysical Inst., Univ. of Bergen, Longyearbyen, Norway), and Florian Geyer (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway)

The Fram Strait is the main gateway for heat exchange between the Arctic Ocean and the Atlantic Ocean, hence estimating the heat transport is important to understand the on-going climate change. An acoustic system for acoustic thermometry was developed under the ACOBAR project. Results from this experiment showed that it was important to monitor the heat content in the north-going West-Spitzbergen current and the south-going East-Greenland current separately. Additionally, the complex oceanographic conditions in this region make it difficult to separate the different arrivals of the acoustic signals in the time domain. The UNDER-ICE experiment, funded by the Research Council of Norway, is the third acoustic thermometry experiment carried out by NERSC in the Fram Strait. Five moorings were deployed from 2014 to 2016, monitoring the north-going and the south-going currents separately. Transmissions were made every third hour for two years along 8 transects, and two moorings were augmented with additional oceanographic sensors. Results from the processing and analysis of the acoustic data are presented, including time series of the depth-range average temperature along the different transects. Oceanographic measurements and a comparison between acoustic observations and modeling results are also presented.

1:30

1pAO3. Using a regional ocean model to understand the structure and sampling variability of acoustic tomography arrivals in Fram Strait. Florian Geyer, Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Nansen Environ. and Remote Sensing Ctr., Thormølens Gate 47, Bergen 5006, Norway, florian.geyer@nersc.no), and Bruce Cornuelle (Scripps Institution of Oceanogr. UCSD, La Jolla, CA)

A regional ocean model for Fram Strait allows to understand the variability and structure of acoustic tomography arrivals. The eddy-permitting model (52 vertical layers and 4.5 km horizontal resolution) was evaluated using long-term moored hydrography data and time series of depth-range averaged temperature obtained from the inversion of acoustic tomography measurements. Geometric ray modelling on the ocean model fields reproduces the measured arrival structure of the acoustic tomography experiment. The combination of ocean and acoustic model gives insights into acoustic propagation during winter and spring. Overlapping arrivals coming from different vertical angles can be resolved and explained. The overlapping arrival of sound channel rays and bottom-reflected rays has implications for the inversion of tomography data in Fram Strait. The increased knowledge about the ray-length variations of bottom-reflected rays is valuable information for choosing appropriate observation kernels for the data assimilation of acoustic tomography data in Fram Strait.

1:45

1pAO4. Effect of sea-ice parameters on acoustic propagation in different Arctic scenarios. Gaute Hope, Hanne Sagen, and Halvor Hobæk (Polar Acoust. and Oceanogr., Nansen Environ. and Remote Sensing Ctr., Thormøhlensgate 47, Bergen 5006, Norway, gaute.hope@gmail.com)

Depending on the setup and purpose of an acoustic propagation system (communication, navigation, tomography, seismic exploration) in ice covered areas, different parameters of the sea-ice are more significant in the interaction of acoustic waves with the ice. In homogeneous ice, with a source close to the surface, significant energy can be transferred to flexural waves and other ice-coupled modes. The modes in the ice are strongly dependent on the elastic parameters of the ice, as well as the thickness of the ice-plate. Additionally, discontinuities in the ice are very important to their propagation. In the inhomogeneous and fractured pack ice in the Arctic Ocean, where the cold water layer results in an acoustic surface channel, the majority of the acoustic waves will interact with the sea-ice at high incidence angles. Here, the roughness of the underside is more significant. Measurements from seismic experiments in an ice-covered fjord on Svalbard as well as long-range propagation in the ice-covered Fram Strait, together with simulations using OASES, are shown and used to gauge the effect of the different ice parameters. Although there is insufficient foundation for interpolation between the two extremes, the characteristics of the scenarios which determine the dominant ice-parameter are described.

2:00

1pAO5. A short history of ONR ice camps: 1978–1994. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

The Office of Naval Research sponsored Arctic field programs almost every year from 1978 to 1994 during the height of the Cold War. Almost all of them had an acoustics component coupled with observations for physical oceanography, geoacoustics, plate tectonics, ice mechanics, and used both active and passive methods. In 1978, these started with emphasis on basin reverberation and ended in 1994 with trans Arctic Ocean tomography. They had acronyms from CANBARX (Canadian Basin), FRAM I—IV, MIZEZ (Marginal Ice Zone), PRUDEX (Prudoe Bay) CEAREX and SIMI/TAP (Sea Ice Mechanics/Trans Arctic Propagation). The author was the chief scientist for most of these programs and will provide an overview of ONR's efforts in the Arctic during the Cold War. [Work supported by ONR.]

2:15

1pAO6. Characteristics of the Arctic environment in the southern Beaufort Sea from Ice Exercise data. John E. Joseph, D. Benjamin Reeder, and Derek R. Olson (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jejoseph@nps.edu)

In early March of 2016 and 2018, the Naval Postgraduate School participated in the biennial naval Ice Exercise (ICEX) conducted in the southern Beaufort Sea. Oceanographic and acoustic data sets collected near the ice camps during both events are compared. While the drift track of the ice camp during ICEX-18 was approximately one-degree south of the track in ICEX-16, there are important similarities in the oceanographic structure shown in both datasets. These characteristics have significant impacts on sound propagation in the region and may affect the performance of acoustic systems such as naval sonar and UUV navigation. Of particular interest are properties at the interface between the cold, fresh surface layer and the contrasting warm, saline Pacific Summer Water (PSW) that lays immediately below it. Sensors indicate turbulent mixing of high-spice and low-spice water occurs at this interface. PSW also sets up a stable subsurface sound channel with Pacific Winter Water and Atlantic Water layers below it. Strength of the sound channel varies from year to year; however, historical data from this region indicates an increasing trend. Other oceanographic features found in the upper 200 m of the under-ice water column in our 120-kHz echosounder dataset are discussed.

2:30

1pAO7. Spectral, spatial, and temporal characteristics of underwater ambient noise in the Beaufort Sea in 1994 and 2016. Rui Chen (Mech. Eng., MIT, 77 Massachusetts Ave., Rm. 5-223, Cambridge, MA 02139, ruic@mit.edu), Andrew J. Poulsen (Appl. Physical Sci., Lexington, MA), and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

Climate induced changes in the Arctic Ocean have severely impacted underwater acoustic communication and navigation; understanding underwater noise characteristics is critical to improving the performance of these operations. Ambient noise from the Beaufort Sea recorded in experiments more than 20 years apart (SIMI94 and ICEX16) are compared to determine differences that may be attributed to the region's rapidly changing environment. Both datasets are collected at spring time under packed ice conditions with 32 element vertical line arrays. Spectral comparison shows noise within 20–300 Hz band is 30–40 dB louder in 1994 than 2016, suggesting the ice cover during SIMI94 was more acoustically active. Beamforming results show ambient noise vertical directionality is focused near the horizontal during SIMI94 but more spread in elevation during ICEX16, with a robust noise notch at the horizontal. Numerical modeling demonstrates that this difference may be attributed to ambient noise during ICEX16 being dominated by surface noise sources at discrete ranges rather than the historical assumption of a continuous and uniform distribution of sources. Temporal statistics of transient ice events show more transient activity during SIMI94 and appear to support the new proposed surface source distribution for ICEX16. [Work supported by ONR and DARPA STO.]

2:45

1pAO8. Multipath arrival tracking for marine vehicles utilizing pattern recognition. Jordan Fouquette and Henrik Schmidt (Mech. and Ocean Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm 5-204, Cambridge, MA 02139, henrik@mit.edu)

In recent years, interest in the Arctic Region has been steadily growing as it has become more accessible due to continued ice recession. Extreme temperatures and significant ice cover have created a unique and challenging acoustic environment. At increased distances, individual ray path data become inconsistent due to improper ray path identification and fading. Through the use of pattern recognition, a multipath arrival tracking algorithm was developed to utilize the unique characteristics associated with each individual ray path for long range tracking purposes. This tracking algorithm analyzes the amplitude and arrival time patterns amongst all individual ray paths in order to accurately identify each ray path as scattering and fading occurs, thereby increasing range-tracking capabilities. This becomes especially useful in the Arctic Region as contacts of interest can be tracked regardless of their position above, below, or within the Beaufort Duct—a sound duct from 100 to 200 m depth. Simulations covering the numerous depth combinations of sources and receivers with respect to the Beaufort Duct highlight the effectiveness that is presented by utilizing multipath arrival data. The developed algorithm takes advantage of these unique patterns in order to provide a tracking capability for marine vehicles to employ. [Work supported by ONR.]

3:00

1pAO9. Double-difference tracking of migrating Bowhead Whales using autonomous vector sensors in the Beaufort Sea. Ludovic Tenorio-Hallé (Scripps Inst. of Oceanogr., 1044 Loring St., San Diego, CA 92109, ludovictenorio@gmail.com), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA), Alexander Conrad (Greeneridge Sci., Inc., Santa Barbara, CA), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), and Katherine H. Kim (Greeneridge Sci., Inc., Santa Barbara, CA)

Passive acoustic monitoring has become a standard method for detecting bowhead whale (*Balaena mysticetus*) activity in Arctic waters. Between 2007 and 2014, over 40 autonomous vector sensors, known as DASARs, were deployed in the Beaufort Sea during the bowhead whale migration

season. Individual DASARs can estimate azimuth, allowing calls to be localized by triangulation using multiple DASARs. However, these bearings are subject to calibration biases, and individual sensors were not precisely time-synchronized, making relative time-of-arrival information unreliable for standard localization purposes. Double-difference methods have previously been applied in seismology to obtain high-precision relative positions of earthquakes by measuring changes in relative travel-times between multiple events over widely distributed seismic sensors. This same concept has also been used to track fin whales on a seafloor seismic network. Here, the double-difference method is applied to previously localized bowhead whale calls in order to improve their relative positions. The approach uses changes in both relative call travel-times and bearings, detected at multiple DASARs, to determine high-precision relative locations of these calls despite the presence of systemic timing and bearing errors in the measurements. The resulting positions may allow tracking of individual whales, which would provide insight into the function of these calls.

3:15–3:30 Break

3:30

1pAO10. Estimating North Pacific Right Whale calling depths in the Bering Sea via modal dispersion. Dana Wright (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, 3737 Brooklyn Ave., Seattle, WA 98105, dana.wright@noaa.gov), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Margaux Thieury (The École Nationale Supérieure de Techniques Avancées de Bretagne, Brest, France), Aileen Fagan (U.S. Coast Guard Acad., New London, CT), Chris Verlinden (U.S. Coast Guard Acad., San Diego, CA), Catherine L. Berchok, and Jessica Crance (Marine Mammal Lab., NOAA Alaska Fisheries Sci. Ctr., Seattle, WA)

Calling depth distributions and ranges are estimated for two types of calls produced by critically endangered eastern North Pacific right whales (NPRW) in the Bering Sea, using passive acoustic data collected with bottom-mounted single-hydrophone recorders at the 50 m isobath. Nonlinear time resampling of 12 NPRW "upcalls" and 20 broadband "gunshots" typically isolated 3 to 4 individual mode arrivals below 200 Hz. Matched-mode processing (MMP) methods were then used to estimate range, depth, and propagation environment. When plotted as a function of range and frequency, MMP ambiguity surfaces reveal the existence of large sound speed gradients in the sediment, along with strongly downward-refracting sound speed profiles during certain summer/fall seasons. Refractive propagation effects from these profiles can induce dramatic changes in signal structure that required new developments in "warping" techniques. Gunshot sounds were generally produced at a few meters depth, while upcall depths clustered between 10 and 25 m, consistent with previously published bioacoustic tagging results from North Atlantic right whales. Current work is examining whether consistent differences exist in the calling depths of bowhead, humpback, and right whales, which could potentially provide a feature for species classification of ambiguous calls. [Work sponsored by NPRB.]

3:45

1pAO11. Acoustic response of a migrating bowhead whale population to open-ocean ambient noise levels. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), Katherine H. Kim, and Alexander Conrad (Greeneridge Sci., Inc., Santa Barbara, CA)

Automated and manual acoustic localizations of migrating bowhead whale calls in the Beaufort Sea were used to examine their acoustic response to changes in wind-driven, continuous, ambient noise levels. At low noise levels, the population's source levels and calling rates increased with increasing noise level, with the source level distribution adjusting to maintain a consistent functional detection range, estimated to be between 20 and 60 km. However, once noise levels exceeded the 75thpercentile of their long-term distribution, source level increases failed to keep pace with further increases in noise level, thereby reducing the population's detection range and associated communication space. Call production rates, on the other hand, continued to increase even up to the highest noise levels. Migrating bowhead whales thus attempted to maintain long-range call detectability by adjusting their source level and calling rate. Beyond a certain noise level, whales cannot increase their source levels, but do continue to increase their calling rate. The results provide context for interpreting the effects of industrial noise on bowhead whale acoustic behavior; for example, distant airgun signals stimulate an increase in mean call production rate equivalent to a 26 dB increase in natural ambient noise levels. [Work sponsored by NPRB.]

4:00

1pAO12. Assessment of safety zones for marine mammals at noise exposure from industrial activity in the Russian Arctic. Alexander Vedenev (P.P. Shirshov Inst. of Oceanology of the Russian Acad. of Sci., 36 Nakhimovsky Prospekt, Moscow 117997, Russian Federation, vedenev@ocean.ru), Konstantin Avilov (A.A. Blagonravov Inst. for Machine Sci. of the Russian Acad. of Sci., Moscow, Russian Federation), and Andrey Lunkov (Prokhorov General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation)

E&P of hydrocarbon along with shipping are the main sources of man-made noise in the ocean. But very few information on the noise from industry activities in Arctic is currently available. Over the next decade we expect the increasing level of anthropogenic sound in the Russian Arctic due to planned large scale construction of Oil&Gas projects including the Shtokman project in the Barents Sea, one of the world's largest natural gas deposits. Noise from seismic surveying, pile driving and construction can damage hearing or disquiet marine mammals. To protect marine mammals from pulsed noise of seismic survey, pile driving and chronic continuous noise at different stages of the Shtokman project construction we estimated the sizes of safety zones with SPL thresholds of 180 dB (zone of injury) and 120 dB for continuous noise (behavior disturbance). Modeling of the transmission loss was done by Normal Mode and PDPE models based on the environmental characteristics in the Barents Sea and real spectra of noise sources—construction vessels and seismic airgun array. The results demonstrate changes in the Safety Zones footprints depending on the stage of construction, season, specific environmental conditions like ice cover interface or presence of gas saturated sedimentary layer.

4:15

1pAO13. Monitoring the Arctic acoustic environments with the International Quiet Ocean Experiment. Philippe Blondel (Phys., Univ. of Bath, Claverton Down, Bath, Avon and NE Somerset BA2 7AY, United Kingdom, p.blondel@bath.ac.uk) and Hanne Sagen (NERSC, Bergen, Norway)

The northern high-latitude regions, including the Arctic Ocean, are becoming increasingly important as a result of global warming and their growing economic and political interests. Sea ice reduction is facilitating resource exploration, marine transport, and other economic activities in the regions. Warming waters lead to shifts in marine ecosystems and in soundscapes. Exploitation of resources in the Arctic is expected to grow in the coming decades, offering new opportunities for marine and maritime industries. To measure the environmental impact of ocean noise at a variety of spatial and temporal scales, the International Quiet Ocean Experiment (http://iqoe.org/) established in late 2017 a working group on Arctic Acoustic Environments. The first activities of the Working Group are focusing on identifying locations and times of existing and past acoustic studies in the Arctic Ocean, and synthesise the state-of-the-art on sounds, past, present, and future in the Arctic Ocean. WG activities at the Arctic Observing Summit 2018 (Davos, Switzerland) are linking with indigenous communities and other local stakeholders, to address emerging trends in marine transport and Arctic resource exploitation, and to plan for where/when the optimal acoustic surveys could be, and what metrics they should prioritise.

4:30

1pAO14. Multi-year measurements of the underwater noise field directionality in the shallow Beaufort Sea during open-water and drifting ice flow conditions. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), Katherine H. Kim, and Alexander Conrad (Greeneridge Sci., Inc., Santa Barbara, CA)

Between 2007 and 2014, over 35 Directional Autonomous Seafloor Acoustic Recorders (DASARs) were deployed over a 280 km swath of the

Beaufort Sea continental shelf (20–55 m depth) during the open-water season, in order to monitor the fall (westward) bowhead whale migration. DASARs have one omnidirectional pressure sensor and two orthogonal particle motion sensors, which permit instantaneous measurements of the azimuths of both transient signals and continuous noise between 20 and 500 Hz. The lack of significant shipping or industrial noise in this region provided a rare opportunity to directly measure the properties of wind-driven noise in expanding ice-free regions. Here, we map the azimuthal directionality of the diffuse Beaufort ambient noise field as a function of frequency and space across seven seasons. The dominant directionality of the diffuse ambient noise field varied strongly with frequency and was highly correlated with the received power spectral density. Certain directional features of the ambient noise field remained stable over seven deployment seasons, suggesting that judicious processing of the ambient noise soundscape could provide underwater navigational information in arctic waters. The influence of local drifting floes and sheets of ice on this directionality is also examined. [Work sponsored by ONR.]

MONDAY AFTERNOON, 5 NOVEMBER 2018 SIDNEY (VCC), 1:00 P.M. TO 5:15 P.M.

Session 1pBA

Biomedical Acoustics and Physical Acoustics: Therapeutic Ultrasound Transducers

Tatiana D. Khokhlova, Cochair
Harborview Medical Center, University of Washington, 325 9th Ave., Box 359634, Seattle, WA 98104

Adam D. Maxwell, Cochair
University of Washington, 1013 NE 40th St., Seattle, WA 98105

Invited Papers

1:00

1pBA1. Therapeutic ultrasound transducers conformal to medical needs. Cyril Lafon, Apoutou N'Djin, David Melodelima, Françoise Chavrier, Raphaël Loyet, and Jean-Yves Chapelon (LabTAU, INSERM, 151, cours Albert Thomas, Lyon 69424, France, cyril.lafon@inserm.fr)

The use of ultrasound for therapy has seen a growing interest since the nineties and the development of endorectal transducers for ablating thermally prostatic tissues. Different devices are currently under clinical evaluation for the treatment of many disorders. Numerical modelling plays a significant role in these developments. For each clinical need, it allows defining the geometry of the transducer and the exposure conditions. The technology for manufacturing the transducer is chosen according to the required acoustic output, density of elements, frequency bandwidth, and medical use (endocavitary, disposable, implantable…). The imaging modality for treatment monitoring is also critical and should be considered in the design process. The goal of this presentation will be to list the various challenges faced during the design of a therapeutic transducer, discuss some solutions that were proposed, and illustrate the process with different examples.

1:20

1pBA2. Rapid prototyping for therapeutic ultrasound transducers. Timothy L. Hall (Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109, hallt@umich.edu) and Adam D. Maxwell (Univ. of Washington, Seattle, WA)

Introduction: Piezoelectric materials are readily available from many suppliers at low cost. Finished ultrasound transducers, however, are significantly more expensive particularly for custom designs. Acoustics research often requires many iterations before a successful experiment design is achieved. This talk will present examples and what we've learned from ten years experience at the University of Michigan constructing transducers with rapid prototyping. Methods: Selective Laser Sintering, Stereolithography, and Fused Deposition Modeling techniques were all tested for transducer construction. Acoustic properties (sound speed, absorption, and impedance) were measured for various materials in each category. Ageing and degradation due to water ingress were assessed as well. Results: Housings for electrical/water isolation were successfully constructed using each technology. Professional quality machines yielded sufficient precision to create focused array structures up to 5 MHz. A majority of stereolithography materials showed very poor water compatibility degrading with exposure over time. Most materials have too low acoustic impedance for good matching layers. Particle filled stereolithography resins have increased density and sound speed that raise the impedance to more suitable values. Acoustic lenses were successfully constructed from stereolithography materials allowing focused transducers to be constructed from low cost flat piezoelectric materials.

1:40

1pBA3. Dual-mode capacitive micromachined ultrasonic transducer arrays for high intensity focused ultrasound and imaging. Ji Hoon Jang, Chienliu Chang, Morten F. Rasumssen, Kevin Brenner, Quintin Stedman (Stanford Univ., Edward L. Ginzton Lab Box 131, Stanford, CA 94305, qstedman@stanford.edu), Arif S. Ergun (TOBB Univ. of Economics and Technol., Stanford, CA), and Butrus T. Khuri-Yakub (Stanford Univ., Stanford, CA)

High Intensity Focused Ultrasound (HIFU) therapy requires imaging guidance to ensure that the correct area is treated. Guidance may be provided by magnetic resonance imaging or ultrasound imaging. With piezoelectric transducers, ultrasound imaging guidance is typically done with a separate transducer array since it is difficult to obtain sufficient imaging and HIFU performance with the same transducers. However, Capacitive Micromachined Ultrasonic Transducers (CMUTs) can have sufficient bandwidth and efficiency to allow both HIFU therapy and imaging to be done with the same transducer array. Additionally, CMUTs have much lower self-heating than piezoelectric transducers and require less cooling when used for HIFU. This means that CMUT-based HIFU systems can be made smaller than other HIFU systems to date, allowing access to treatment areas that were not previously possible. In addition, using the same array for imaging and treatment eliminates challenges with co-registration. We have constructed a dual-mode HIFU and imaging probe using a 32x32 CMUT array integrated with an application-specific integrated circuit. Ultrasound imaging guided ablation of ex-vivo tissue has been demonstrated with the probe, with rapid switching between HIFU and imaging mode.

Contributed Papers

2:00

1pBA4. Various approaches for designing phased arrays for high-intensity focused ultrasound therapies: From sparse to fully-populated configurations. Oleg Sapozhnikov, Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105; Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Pavel Rosnitskiy (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Leonid Gavrilov (N.N. Andreyev Acoust. Inst., Moscow, Russian Federation)

High-intensity focused ultrasound (HIFU) therapies are often performed using multi-element phased arrays. Independent variation of the amplitudes and phases at the array elements allows electronic steering of the focus and compensation for aberrations. To suppress the formation of grating lobes, the arrangement of the array elements should be non-periodic. Three approaches that have been recently employed in our studies to solve this problem are presented and discussed: a random element arrangement [L.R. Gavrilov and J.W. Hand, IEEE Trans. UFFC 2000], a multi-armed spiral arrangement [V.A. Khokhlova *et al*., Physics Procedia 2016], and recently proposed array design with tight packing of the elements based on the capacity-constrained tessellation [P.B. Rosnitskiy *et al*., IEEE UFFC 2018]. The efficiency of two arrays with the same geometric and physical parameters is compared: a 256-element array with a compact 16-spirals layout of circular elements and a fully populated array comprising polygonal elements of equal area. It is shown that for the same intensity at the elements, the fully-populated array provides twofold higher total power while maintaining the same electronic focusing capabilities as compared to the spiral one which can be beneficial for high-power applications such as histotripsy. [The work was supported by NIH R01EB7643 and R01EB025187, RFBR 17-02-00261, and Ph.D. student stipends from the "Basis" Foundation and of the President of Russia SP-2644.2018.4.]

2:15

1pBA5. High throughput system for preparing samples for genomic and epigenetic assays. Thomas Matula, Karol Bomsztyk, Brian MacConaghy, and Adam D. Maxwell (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu)

Genomic and epigenomic therapies hold great promise for "fixing" faulty genes. In genomics, DNA is sequenced to determine the order of the nucleotides, while in epigenetics, immunoprecipitation of chromatin is used to locate specific proteins of interest. The first step for these therapies is to identify the specific sequence or gene responsible for the disease. To prepare samples for these assays, DNA or chromatin is sheared into fragments, most often using cavitation. However, standard tools such as ultrasonic horns can be highly inconsistent, leading to poor assays. In order to overcome the problems of inconsistent fragmentation, we designed and built a transducer array (free field pressures $P^+ = 30$ MPa, $P^- = 12$ MPa) capable of processing samples directly in 96-well microplates. An array of transducers is mounted below a microplate, coupled to a lens array that focuses acoustic energy into each well of a microplate. Intense cavitation is generated in each well. After treatment, gel electrophoresis shows that DNA and chromatin are fragmented to a range of sizes extending as low as 100 base pairs (34 nm). Importantly, this process results in consistent fragmentation that leads to consistent repeatable assays. [Funded by NIH R33CA191135, R21GM111439, R01DK103849 and Life Sciences Discovery Fund #12330479.]

Invited Papers

2:30

1pBA6. Low frequency (20 kHz), patch-like ultrasound applicator for chronic wound treatment. Peter A. Lewin, Olivia Ngo, Evan Niemann, Vivinya Gunasekaran, Prabagar Sankar, Alec Lafontant, Sumati Nadkarni (School of Biomedical Eng., Sci., and Health Systems, Drexel Univ., 3141 Chestnut St., Bossone 701, Philadelphia, PA 19104, lewinpa@drexel.edu), Rose Ann DiMaria-Ghalili (College of Nursing and Health Professions, Drexel Univ., Philadelphia, PA), Michael Neidrauer, Leonid Zubkov (School of Biomedical Eng., Sci., and Health Systems, Drexel Univ., Philadelphia, PA), Michael Weingarten (Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), and David Margolis (Dermatology, Perelman School of Medicine Univ. of Pennsylvania, Philadelphia, PA)

Chronic wounds, such as venous and diabetic ulcers, cost the U.S healthcare system alone, close to $25 billion annually. Hence, a reduction of healing time directly translates into savings of treatment related expenses. This work describes the implementation of patch-like, un-tethered and clinically viable therapeutic ultrasound applicator. The device uses well-defined non-cavitational and non-thermal levels of ultrasound energy; its peak acoustic output pressure amplitude was intentionally limited to 55 kPa, corresponding to a spatial peak temporal peak intensity of 100 mW/cm^2. A small (n = 8) pilot study targeting diabetic ulcers treatment was performed and

indicated that with its light weight (<20g), and circular (40 mm dia) disk shape architecture this applicator is suitable to be embedded in wound dressing. The average time to wound closure was 4.7 weeks for subjects treated with the active device, compared to 12 weeks for subjects treated with a sham applicator, suggesting that patients with diabetic ulcers may benefit from the proposed treatment. [Work supported by the NINR grant5R01NR015995. The contents of this presentation are solely the responsibility of the authors and do not necessarily represent the official views of the NIH.]

2:50

1pBA7. Treatment monitoring for sonothrombolysis in deep vein thrombosis: Receiver array design. Christopher Acconcia, Ryan M. Jones, and Kullervo Hynynen (Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N3M5, Canada, chrisacc@sri.utoronto.ca)

The current treatment standard for deep vein thrombosis (DVT) is anticoagulation therapy, which does little to address long term morbidity compared to clot removal approaches. High intensity ultrasound can resolve clots by generating bubble clouds that erode thrombi. However, lack of appropriate treatment monitoring is a limiting factor in its widespread adoption. Passive cavitation imaging is capable of high frame rate, volumetric imaging, the combination of which has been shown to be important for monitoring the onset and development of bubble clouds inducing clot erosion (Acconcia *et al.* 2017). The results from our previous work motivated development of a device tailored to DVT with a relevant geometry (e.g., semi-cylindrical). Transmit simulations for such a device were conducted in a human thigh model (Smirnov and Hynynen 2017) with a design based on the modular transducer technology developed in our lab (Ellens *et al.* 2015). Here, we examine the integration of a sparse, randomly distributed receiver array within a semi-cylindrical arrangement of transmit modules for acoustic-based monitoring. Using a multi-layered propagation model, cavitation sources were localized to the femoral vessel, the accuracy of which depended on the inclusion of phase corrections. The receiver size was shown to be an important consideration with implicit trade-offs between directivity and channel SNR. Volumetric rates of ~1 MHz should be achievable with a modest number of receivers (128) in the presence of experimentally derived noise conditions.

3:10–3:25 Break

Contributed Papers

3:25

1pBA8. Creating uniform ultrasound fields in the brain without an array. Luke A. Richards, Robin Cleveland, and Eleanor P. Stride (Dept. of Eng. Sci., Oxford Univ., Inst. of Biomedical Eng., Old Rd. Campus, Oxford, Oxfordshire OX3 7DQ, United Kingdom, luke.richards@eng.ox.ac.uk)

There has been a recent interest in using patient specific ultrasound lenses to correct for the phase aberration caused by the skull in focused transcranial ultrasound. Here, we apply acoustic lenses to the problem of creating a highly uniform field inside the skull, motivated by the requirements of targeted and ultrasound responsive nanodroplets being developed for the treatment of brain metastases. The phase and amplitude required for uniform sonication was determined from simulations of ultrasound propagation utilising CT data as a basis for the geometry of the patient's skull. From these simulations, silicone lenses were produced, capable of introducing phase inhomogeneity in the incident ultrasound beam to compensate for the effects of transmission through the skull. Experimental measurements with an *ex vivo* skull showed that phase uniformity could be at least partially restored within the skull cavity. Methods for modulating the amplitude of incident ultrasound have also been investigated. Stereolithography was used to directly print in highly attenuating (15 dB/cm at 1 MHz) materials, allowing rapid production of customised and selective acoustic absorbers. A 3D printed transducer backing was also studied, able to suppress transducer output in certain regions, and provide up to 15 dB dynamic range across the surface of a planar ultrasound transducer.

3:40

1pBA9. Enhanced shock scattering histotripsy with pseudo-monopolar ultrasound pulses. Yige Li, Timothy L. Hall, Zhen Xu, and Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, 2135 Carl A. Gerstacker Bldg., 2200 Bonisteel Boulevard, Ann Arbor, MI 48109, yigeli@umich.edu)

Shock scattering histotripsy (SSH) involves a complex interaction between positive and negative phases of an acoustic burst to initiate a robust cavitation bubble cloud. To more precisely study these effects and optimize SSH therapy, we constructed a frequency compounding transducer to generate pseudo-monopolar ultrasound pulses. The transducer consisted of 113 individual piezoelectric elements with various resonant frequencies (250 kHz, 500 kHz, 750 kHz, 1 MHz, 1.5 MHz, 2 MHz, and 3 MHz). For each resonant frequency, a nearly 1.5-cycle pulse could be generated. Pseudo-monopolar peak positive pulses were generated by aligning the principal peak positive pressures of individual frequency components temporally so that they added constructively, and destructive interference occurred outside the peak-positive-overlapped temporal window. After inverting the polarity of the excitation signals, pseudo-monopolar peak negative pulses were generated similarly by aligning principal peak negative pressures. Decoupling the positive and negative acoustic phases could have significant advantages for therapeutic applications enhancing precision, avoiding cavitation at tissue interfaces, and reducing the acoustic aperture required for effective therapy. For example, we show that 16 SSH bubble clouds can be generated using only peak positive pulses following a single peak negative pulse. This results in a precise elongated lesion within red-blood-cell phantoms.

3:55

1pBA10. Development of a freely available simulator with graphical interface for modeling nonlinear focused ultrasound fields with shocks. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle; Phys. Faculty, Moscow State Univ., CIMU, APL, University of Washington, Seattle, Washington 98105, verak2@uw.edu), Petr V. Yuldashev, Ilya Mezdrokhin, Pavel Rosnitskiy, Maria M. Karzova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA; Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Measurement-based modeling is gaining acceptance as a standard tool for characterizing the nonlinear fields of existing therapeutic ultrasound devices and designing new ones. Here, a freely available simulation tool is presented for modeling axially symmetric, strongly nonlinear HIFU beams with shocks in a layered propagation medium such as water and different types of tissue. Two nonlinear wave equations are included in the simulator: the KZK equation generalized to include an equivalent source boundary condition for strongly focused beams and the Westervelt equation in a

nonlinear wide-angle parabolic representation. Both equations are solved in the frequency domain and permit definition of the HIFU transducer as an annular array. The geometrical parameters and power output of the array, electronic focus steering along the beam axis, and acoustic properties of the layered propagation medium can be configured via graphical interface. Visualization and output of various acoustic field parameters such as peak positive and negative pressures, shock amplitude, intensity, heat deposition rate, and total power of a beam are also provided. The simulator can be used for transducers without ideal symmetry through definition of an equivalent radially symmetric source. Widespread availability of such simulation tools will help advance standardized utilization of measurement-based modeling and facilitate the adoption of such approaches for HIFU treatment planning. [Work supported by NIH R01EB7643, R01EB025187, and RSF №14-12-00974.]

4:10

1pBA11. Nonlinear ultrasound fields generated by a 256-element spiral array for boiling histotripsy. Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98104, tdk7@uw.edu), Wayne Kreider, Christopher Hunter, Mohamed A. Ghanem, Bryan Cunitz (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ.,Moscow, Russian Federation), and Vera A. Khokhlova (Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Seattle, WA)

Boiling histotripsy (BH) is a HIFU approach that uses millisecond-long ultrasound bursts with high-amplitude shocks to mechanically fractionate tissue. In this work, the performance of a BH pre-clinical system operating at 1.5 MHz was characterized by hydrophone measurements in water and lesion formation capabilities were tested in *ex vivo* tissue. The system comprises a 256 element array driven by Verasonics Ultrasound Engine with a 1.2 kW external power source enhanced by an additional capacitor bank. The maximum pulse average acoustic power of the system was 2.2 kW (as measured by radiation force balance), sustained for 10 ms with <15% drop in amplitude. At the focus, a fully developed shock of 100 MPa amplitude formed at 275 W acoustic power. The steering range defined at -3 dB in both linear and nonlinear beam was 19 mm laterally and 38 mm axially forming the region where 100 MPa shocks can be achieved with less than twofold increase in acoustic power. Within the axial steering range, shock formation characteristics varied substantially due to changes in the effective transducer F-number. The system was successfully used to produce volumetric BH lesions in *ex vivo* tissue using electronic steering only. [Work supported by NIH R01EB7643, R01GM122859, and R01EB025187.]

Invited Paper

4:25

1pBA12. New international standards for high pressure therapeutic ultrasound fields. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

High pressure levels of strongly focused therapeutic transducers create new challenges for formulating international standards for measurement and simulation. Earlier efforts of Working Group 6, High Intensity Therapeutic Ultrasound and Focusing Transducers, Technical Committee 87 of the International Electrotechnical Commission, resulted in a standard for measuring high power (up to 500 W) and a specification for measuring ultrasound fields at lower amplitudes using existing linear methods and scaling up to higher therapeutic levels. Presently, work is underway on a new document to describe direct approaches for characterizing ultrasound pressure fields in water up to 100 MPa through the use of state of the art optical and robust hydrophones. Because the real goal of our efforts is to determine therapeutic pressure levels in tissue at a target site, we have initiated work on a second hybrid approach which combines characterization of the radiated field close to the transducer in water with nonlinear field simulation programs. In this way the characterization data which preserve the uniqueness of a particular transducer are used as input to a simulator program for realistic field prediction. Our group is addressing how to simulate nonlinear propagation in heterogeneous absorbing tissues accurately and to benchmark simulation software.

Contributed Papers

4:45

1pBA13. Needle hydrophone transfer function model for characterizing therapeutic transducers. Keith A. Wear and Yunbo Liu (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

A rigid piston model has been developed to predict needle hydrophone transfer functions. The model predicts that "effective" sensitive element diameter depends on the frequency and can exceed geometrical sensitive element diameter by a factor of up to 2.25. The model predicts harmonic distortion for hydrophone measurements of focused pressure waves, which were modeled using nonlinear plane wave theory for spectral characteristics and Gaussian radial pressure distributions. The pressure wave model was validated by comparing to hydrophone (sensitive element diameter: 85 microns) measurements performed in focal planes of source transducers (3.5 MHz, 64 mm diameter, 89 mm focus and 5 MHz, 19 mm diameter, 38 mm focus). Root mean squared difference (RMSD) between the model fit and measured relative strengths of the first four harmonics was 9% (3.5 MHz transducer) and 11% (5 MHz transducer) where Blackstock's sigma parameter (*JASA*, 39, 1019–1026, 1966), which measures extent of nonlinearity, was optimized to fit the data. RMSD between model and measured half width half maxima was 10 $\pm$ 5%. Tone bursts from each transducer were measured using needle hydrophones (sensitive element diameters: 200, 400, 600, and 1000 microns). RMSD between model and measured harmonic distortion was 12 $\pm$ 4% for the first four harmonics.

5:00

1pBA14. Holographic characterization of therapeutic array transducers with assessment of measurement limitations and robustness. Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Seattle, WA), Mohamed A. Ghanem (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Seattle, WA)

Acoustic holography measurements and corresponding numerical projection methods have recently gained acceptance for characterizing the fields generated by high intensity therapeutic ultrasound (HITU) applications. To facilitate the standardization of such measurement-based simulation methods, it is important to understand how practical measurement challenges and limitations impact the results. Toward this end, holography measurements were acquired and analyzed for characterization of two therapeutic array transducers, each with 256 elements. For different focusing configurations, measurements comprised 2D holography scans as well as 1D scans used to independently quantify the focal lobe. The fields projected from holograms were compared to independent 1D scans to evaluate the impacts of hydrophone directivity, non-orthogonality of the axes of the positioner used to move the hydrophone, and overall mechanical stability of measurements. Results demonstrate that directivity effects exceeded those expected based on hydrophone size and altered focal pressures on the order of 10%. Non-orthogonality of the positioner shifted apparent focal locations by measurable distances, which could be confused with small shifts in device fixturing. However, holographic reconstruction of the ultrasound field structure near the focal lobe was robust with respect to directivity and non-orthogonality, which enables compensation by relative calibration of source output levels. [Funding support by NIH R01-EB025187, R01-EB007643, and P01-DK043881.]

MONDAY AFTERNOON, 5 NOVEMBER 2018 RATTENBURY A/B (FE), 1:00 P.M. TO 2:15 P.M.

Session 1pEAa

Engineering Acoustics: Transducer Characterization

Caleb F. Sieck, Chair

Code 7160, NRC Postdoctoral Research Associateship Program, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Contributed Papers

1:00

1pEAa1. Using and selecting small condenser format microphones for impulsive noise measurement. James Weir (Bruel & Kjaer North America, 3079 Premire Pkwy, Ste. 120, Duluth, U.S. 30097, GA, jim.weir@bksv.com)

The small format condenser microphone is the most often used transducer for impulsive measurements of firearms and suppressors. In this paper, the properties and characteristics of these microphones will be detailed and how the selection and use of these condenser microphones will affect the consistency and accuracy of measurements. The goal is to raise awareness and measurement consistency for developers, regulators, and user/hobbyists. The short wavelengths of sound at high frequencies are physically manipulated by the physical design of the transducers. A thorough testing of these products reveal these characteristics and their effect on the accuracy of the resulting measurement. The directivity, frequency response, impulse response, and damping characteristics will be compared. Results will be represented, along with anecdotal stories of real life implementation.

1:15

1pEAa2. Acoustic beam pattern for a push-pull laser interferometric sound sensor. Michael S. McBeth (SSC Atlantic, SSC Atlantic/NASA Langley Res. Ctr., 11 West Taylor St., M.S. 207, Hampton, VA 23681, m.s.mcbeth@ieee.org)

A push-pull laser interferometric sound sensor uses two parallel laser beams where both beams are exposed to the sound filled fluid. In arrays of conventional piezoelectric hydrophones, each array element produces a voltage and the beam pattern is obtained by deriving the response as a function of acoustic wave angle in terms of electrical power divided by the maximum response power along the acoustic axis. In a push-pull laser interferometric sound sensor the response function is an optical path difference or phase difference between the two laser beams as a function of acoustic wave angle. In an analogous way, we derive a response function in terms of the optical path difference between two optical line sensors as a function of acoustic wave angle and divide by the response along the acoustic axis. This normalized response function is squared to obtain the beam pattern for a push-pull laser interferometric sensor. We compare this beam pattern to that of a conventional line array.

1:30

1pEAa3. MEMS microphone based on the membrane with bias voltage and FET (field effect transistor) mechano-electrical transduction. Junsoo Kim, Hoontaek Lee, Chayeong Kim, Donghwan Seo (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), San 31 Hyoja-dong, Nam-gu, Pohang, Gyeongsangbuk-do 37673, South Korea, chak7258@postech.ac.kr), Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyungbuk, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

Most commercially available microphones use a capacitive method, and their structure and performance are saturated to some extent. However, due to the limitation of the capacitive transduction method, the roll-off at the low frequency is inevitable, and there is a limit in reducing the mechanical thermal noise caused by the squeeze film damping that occurs between the membrane and backplate structure. Proposed FET (field effect transistor) based MEMS microphone detects a change in the source-drain current according to the gate voltage change of the FET induced by vibrating the membrane with the electric field. In the case of a MEMS microphone using

the proposed FET, low frequency roll-off according to the energy conversion does not occur, and the size of the backplate can be drastically reduced as compared with the conventional MEMS microphone, thereby further reducing the mechanical thermal noise, leading to the possibility of achieving the higher SNR (signal to noise ratio). In this study, design and fabrication, performance test of the proposed FET based MEMS microphone are conducted. [Work supported by CMTC, UM15304RD3.]

1:45

1pEAa4. Comparison of acoustic quality according to the types of flat panel display: Direct drive with exciter speaker. Hyung Woo Park (IT, Soongsil Univ., 1212 Hyungham Eng. Building 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 06978, South Korea, pphw@ssu.ac.kr), SungTae Lee, Kwanho Park (LGDisplay, Paju, South Korea), and Myungjin bae (IT, Soongsil Univ., Seoul, South Korea)

With the technological improvements in the display industry, display panel manufacturers have changed the picture quality from simple high definition to augmented reality or virtual reality. In the previous research, the sound field implementation and sound quality improvement using organic light-emitting diode (OLED) flat panel was introduced. Recently, we have developed a technology for implementing sound in a drive even on a high-quality liquid crystal display. The LED panel has a multi-layer structure, and the backlight unit has a disadvantage of the wavelength transmission structure. In this study, we proposed a method to analyze LCD more effectively through previous studies. Also, we can play left channel and right channel like analysis 2 channel flat panel speakers with one panel. This flat panel speaker consists of direct drive actuator and diaphragm and is used as the outer glass layer of OLED and multilayer LCD. In the LCD panel, we configured the enclosure and adjusted the exciter for the output powers of the exciter were controlled to ensure. Furthermore, the vibration was transmitted to the last flat sheet. We also implemented a clear-sounding direct-drive exciter speaker flat panel display.

2:00

1pEAa5. A study on the optimal speaker position for improving sound quality of flat panel display. SungTae Lee (Res., LGDisplay, 37-8, LCD-ro 8beon-gil, Wollong-myeon, Paju 10844, South Korea, owenlee@lgdisplay.com), Hyung Woo Park (IT, Soongsil Univ., Seoul, Seoul, South Korea), Kwanho Park (Res., LGDisplay, Paju, South Korea), and myungjin bae (IT, Soongsil Univ., Seoul, South Korea)

As a technological improvement in display industry, flat panel manufacturers have changed the screen quality from simple high definition to 5 sensation satisfaction. The sound quality is improved, which is the easiest way to enhance the senses satisfaction. The technology, which in a previous meeting in Minneapolis, is using an exciter speaker. That is, the vibrate panel itself is used to make and play the sound for multimedia information. In this study, the optimum position of the exciter was studied to improve the sound quality of the proposed multi–channel flat–panel speaker. We studied the pattern of the acoustic energy emitted by the exciter and the position and the radiation pattern of the speaker to prevent distortion of the sound when watching the display from a distance. The panel encloses the exciter and the back plane. We also studied the method of construction.

MONDAY AFTERNOON, 5 NOVEMBER 2018 RATTENBURY A/B (FE), 3:00 P.M. TO 4:30 P.M.

Session 1pEAb

Engineering Acoustics: General Topics in Engineering Acoustics

Caleb F. Sieck, Chair

Code 7160, NRC Postdoctoral Research Associateship Program, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Contributed Papers

3:00

1pEAb1. Production method for simulcast in 22.2 multichannel sound broadcasting. Takehiro Sugimoto and Tomoyasu Komori (Sci. & Technol. Res. Labs., NHK, 1-10-11 Kinuta, Setagaya-ku, Tokyo 1578510, Japan, sugimoto.t-fg@nhk.or.jp)

The 8K with 22.2 multichannel (22.2 ch) sound broadcasting is planned to be launched since December 2018 in Japan. NHK is going to simulcast 2-channel stereo and 5.1 surround together with the 22.2 ch sound for backward compatibility with existing home reproduction equipment, e.g., AV receiver and soundbar. However, individual production for each audio format is impractical because it needs many engineers and a lot of production equipment compared with a production in single format. Hence, an efficient production method for simulcast which does not demand additional production resources is required. The first proposal is a tone compensation for downmixing from the 22.2 ch sound to the other audio formats. The energy spectrum of the downmixed signal is compensated to harmonize with that of the 22.2 ch sound. The second proposal is a loudness chase to have the loudness level of the downmixed signal follow that of the 22.2 ch sound. The loudness level of the downmixed signal usually differs from that of the 22.2 ch sound, so that the loudness chase successively adjusts the loudness level without unnatural level change and matches them by the end of the program.

3:15

1pEAb2. Distributed acoustic sensing for near-surface seismic applications. Richard D. Costley, Gustavo Galan-Comas (U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil), Kent K. Hathaway (U.S. Army Engineer Res. & Development Ctr., Kitty Hawk, NC), Stephen A. Ketcham (U.S. Army Engineer Res. & Development Ctr., Alexandra, VA), and Clay K. Kirkendall (Naval Res. Lab., Washington, DC)

Distributed acoustic sensing (DAS) has been used in the oil and gas industries for vertical seismic profiling (VSP) for approximately the past ten years. Its use for near-surface seismic surveys has also been investigated. This study relates to these types of applications. DAS consists of a buried optical fiber cable with one of its fibers connected to electro-optical instrumentation, i.e., an optical interrogator. The interrogator injects pulses of coherent light into the fiber and receives backscattered light. Seismic disturbances deform the cable, strain the interrogated fiber, and change the fiber's optical path length. The resulting phase changes of the backscattered light are recorded by the interrogator. These phase change measurements, proportional to strain, are probed through time at spatially sequential (i.e., "distributed") segments along the length of the fiber. The length of each segment is referred to as the gauge length. A single fiber is thus a nominal array of sensors with the gauge length defining the spatial separation. Here, we present DAS signals from ground vibrations caused by impact-hammer and electro-magnetic-shaker sources. The signals were recorded simultaneously by geophones for comparison. Interrogator modifications allowed selectable gauge lengths, 10, 4 and 2 meters. The experimental results demonstrate the benefits of higher resolution.

3:30

1pEAb3. Imaging aeroacoustic sources in a wind-tunnel with a massive array of MEMS microphones. Yinshi Zhou, Vincent Valeau (Institut PPRIME UPR 3346, CNRS-Université de Poitiers-ENSMA, 6 rue Marcel Doré TSA 41105, Poitiers Cedex 9 86073, France, vincent.valeau@univ-poitiers.fr), Jacques Marchal (Institut Jean Le Rond d'Alembert, Sorbonne Université, CNRS, SAINT-CYR-L'ECOLE, France), Régis Marchiano, and françois ollivier (Institut Jean Le Rond d'Alembert, Sorbonne Université, CNRS, Paris, France)

This presentation deals with the identification of aeroacoustic sources in the open section of an anechoic wind-tunnel by using a three-dimensional (3D) array of 256 microphones. The antenna is made of three perpendicular planar arrays enclosing the test-section, and the microphones are digital MEMS microphones. The data processing is based on the beamforming technique associated with a deconvolution method (CLEAN) developed in 3D to improve the spatial resolution. Some preliminary tests with a well-controlled artificial source validate the set-up and the calibration procedure. Some measurements are carried out with aeroacoustic sources generated by the interaction of obstacles with the wind-tunnel flow. The academic case of a cylinder emitting aeolian tones is first considered. A more complex case is then considered with a symmetric wall-mounted NACA airfoil. The different sources (trailing edge noise, tip noise, and junction noise) are successfully identified in a 3D volume in terms of position. The study demonstrates that aeroacoustic source identification is achievable in three dimensions by using a massive array of very cheap MEMS microphones when associated to appropriate data processing techniques.

3:45

1pEAb4. Active control of a highly directional primary source by compact novel secondary sources. Qi Hu and Shiu-Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, KLN, Hung Hom Na, Hong Kong, qi.bs.hu@connect.polyu.hk)

The active noise control in free space for a global attenuation is always very challenging, especially for extended and complex radiators. This work focuses on a finite simple line source possessing strong directivity of multi-lobe radiation pattern as the primary source, which is frequently encountered in traffic noise control problems modelling a busy road or the train noise. Conventional secondary sources of compact size radiate the sound wave omnidirectionally, resulting in tangible pressure level increase in off-axis areas. This paper introduces a novel constructed control source, which has reasonably directional sound radiation capability even with a very compact size in low-frequency. Comparing to common directive sources, such as axially oscillating baffled pistons, the novel sources can significantly improve the active control result in a global way.

4:00

1pEAb5. Prediction of noise from an automotive turbocharger centrifugal compressor using three-dimensional computational fluid dynamics. Rick Dehner, Ahmet Selamet, and Emel Selamet (Mech. and Aerosp. Eng., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, dehner.10@osu.edu)

The noise generated by centrifugal compressors for automotive turbochargers consists of narrow and broadband components. Narrowband noise occurs near integer multiples of the shaft rotational frequency, and the prevailing order is typically equal to the number of main impeller blades (impeller blade-pass frequency). However, for a centrifugal compressor without a ported shroud recirculating casing treatment, such as the current contemporary configuration, broadband noise is dominant, typically in the 4–12 kHz range. This study applies a three-dimensional computational fluid dynamics model to this compressor installed on a steady-flow turbocharger stand. As the compressor flow rate is reduced at constant rotational speed, broadband noise is elevated as a result of deterioration of the flow-field near the inducer (inlet) of the impeller, due primarily to flow separation from the suction surface of impeller blades. Detached eddy simulations provide reasonable agreement with the corresponding experimental measurements in terms of capturing performance, as well as acoustics, thereby shedding insight into this noise source mechanism associated with such flow separation.

4:15

1pEAb6. A study on sudden unintended acceleration estimation engine sounds by vehicle type. Uk-Jin Song (Dept. Information and TeleCommun. Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul 156743, South Korea, imduj@ssu.ac.kr), Seonggeon Bae (Kangnam Univ., Youngin, South Korea), and Myungjin Bae (Soongsil Univ., Seoul, South Korea)

A car is one of the road transport that carries the passenger or freight by transmitting power generated by the engine to the wheel. The automobile has been continuously developed up to modern, so that the power can be transmitted to the wheel with higher efficiency. Recently, however, an accident involving the suspicion of mechanical defect has occurred through the electronicization of internal parts of automobiles. One of them is estimated to be a sudden unintended acceleration. A sudden unintended acceleration is a phenomenon in which the rpm of the engine rises rapidly regardless of the intention of the driver, resulting in a rapid acceleration of the vehicle speed. Accidents involving automobiles are decreasing due to various efforts to reduce the incidence rates, but accidents with the sudden unintended acceleration are on the rise. Therefore, in this paper, we want to analyze the engine sounds of accidents claiming sudden unintended acceleration obtained by a black box by the vehicle type.

Session 1pNS

Noise, Physical Acoustics, Signal Processing in Acoustics, and ASA Committee on Standards: Supersonic Jet Aeroacoustics II

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Seiji Tsutsumi, Cochair
JEDI center, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara 252-5210, Japan

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

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. An investigation of the interactions between impulsively started and steady supersonic jets. Karthikeyan Natarajan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., EAD, PB 1779, Old Airport Rd., Bangalore 560017, India, nkarthikeyan@nal.res.in), Suriyanarayanan P, Lakshmi Venkatakrishnan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., Bangalore, Karnataka, India), and Sankaran Sathiyavageeswaran (Satish Dhawan Space Ctr., Indian Space Res. Organization, India, Sriharikota, Andhra Pradesh, India)

The ignition of the solid rocket booster strap-on and the following IOP wave have a severe influence on the dynamic as well as the acoustic loads experienced by a launch vehicle during lift-off. The jet from the strap-on interacts with the core jet and its established flow over the jet blast deflector, influencing the acoustic field experienced by the launch vehicle. The existing literature provides very little understanding of such interactions. This work investigates the interaction between the jet from SRB and the core, by simulating them with an impulsively started supersonic jet in close proximity to another established jet. While the flow for the core nozzle was allowed directly from the settling chamber, a novel approach using a quick opening valve was used for initiating the flow through the strap-on nozzle. The evolution of a strong shock wave emanating from the strap-on nozzle and the ensuing vortex ring, their interaction with the core jet flow, etc., was captured using schlieren. The results document the fluid dynamic interactions such as the evolution of a strong shock wave emanating from the strap-on nozzle and the ensuing vortex ring, their interaction with the core jet flow, etc., using flow visualizations and acoustic measurements.

1:25

1pNS2. Flow and acoustics of an isothermal supersonic 2:1 rectangular jet with an adjacent surface. Ephraim Gutmark, Florian Baier, and Aatresh Karnam (Aerosp. Eng., Univ. of Cincinnati, Rhodes Hall 799, Cincinnati, OH 45221-0070, ephraim.gutmark@uc.edu)

In integrated airframe/propulsion system configurations, additional noise sources can be created from the interactions between the jet flow and the adjacent airframe surfaces. Another situation of surfaces of close proximity occur during takeoff and landing when the ground is close enough to cause the jet-surface interference. The impact of a plate oriented parallel to the axis of a 2:1 rectangular supersonic jet ($D_e = 20.65$mm) at the minor axis side is studied. Plate offset (h) distances of h = 0, 1, 2, and 3 equivalent diameters from the nozzle lip to the surface are included. The impact of the plate is studied at nozzle pressure ratios (NPRs) of 2.5–4.5 when the design Mach number is 3.67, at a temperature ratio of TR = 1.1. Mean and turbulent flow field data are acquired using streamwise particle image velocimetry (PIV). Trends of shock cell spacing, potential core length, and shear layer development are analyzed. Near and far field data are taken and correlated with the flow field details. The offset from the nozzle is shown to vary flow and acoustic properties and impact screech tones.

1:45

1pNS3. Laser optical measurement of acoustic phenomena in the near field of a supersonic jet using 2-D position sensitive detector. Koji Okamoto, Kazuya Fukatsu, Masahito Akamine (Dept. of Adv. Energy, Graduate School of Frontier Sci., The Univ. of Tokyo, 5-1-5, Kashiwanoha, Kashiwa, Chiba 277-8561, Japan, k-okamoto@edu.k.u-tokyo.ac.jp), and Susumu Teramoto (Dept. of Aeronautics and Astronautics, Graduate School of Eng., The Univ. of Tokyo, Bunkyo, Tokyo, Japan)

Acoustic measurement close to noise sources is significant to understand the generation mechanisms of jet noise. Microphones are generally used for the acoustic measurement, but they may disturb the flow and acoustic fields when they are used in the near field of a jet. In this study, an optical measurement method using a laser and 2-D position sensitive detector (PSD) is proposed for the acoustic measurement in the near field. In this method, 2-D PSD detects the angle and direction of the refraction of the laser path by acoustic wave passing, so that the propagating direction, as well as the acoustic intensity, is expected to be measured by this method. To discuss its validity, this method was applied to the acoustic measurement of a correctly expanded supersonic jet. In this experiment, microphone measurement was also carried out simultaneously, and cross correlation between the signals of these two measurements is discussed. Also, the measured spectra and propagating directions for different frequencies are compared with those of the acoustic intensity vector measurement.

2:05

1pNS4. Experimental investigation of the impacts of total temperature non-uniformities on the flow and acoustic fields of a heated supersonic jet. Kyle A. Daniel (Aerosp. and Ocean Eng., Virginia Tech, 460 Old Turner St., Blacksburg, VA 24060, kyled1@vt.edu), David Mayo (Mech. Eng., Virginia Tech, Blacksburg, VA), Kevin T. Lowe (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), and Wing Ng (Mech. Eng., Virginia Tech, Blacksburg, VA)

In recent years, the noise produced by tactical aircraft has become a growing concern due to stricter community noise standards and the negative health effects it has on flight support personnel. This study proposes the examination of a new novel noise reduction method involving thermal non-uniformities in heated supersonic jets. Thermal non-uniformities have the advantage of increasing turbulent mixing in jets without the use of additional hardware and can most likely be implemented in afterburning engines with minimal modification. In the course of this study a thermally non-uniform Mach 1.5 heated jet with a centered and offset thermal non-uniformities will be examined. It will be shown that thermal non-uniformities reduce the far-field radiated noise up to ~2dB at peak frequencies and have a measurable impact on the directivity of radiated noise, including a narrowing of the Mach wave emission angle and a non-uniform azimuthal directivity. These changes in the acoustic field are directly related to global changes in the turbulence development observed in the jet. These effects include a shortened potential core, increased shear layer thickness, and a decreased mean flow in regions of peak Reynolds shear stress. These effects are captured by stereoscopic PIV and near and far-field microphone measurements.

2:25

1pNS5. Visualization of acoustic phenomena of imperfectly expanded supersonic jets using acoustic-triggered conditional sampling analysis. Yuya Sekiguchi, Koji Okamoto (Graduate School of Frontier Sci., Univ. of Tokyo, Kashiwanoha 5-1-5, Kashiwa, Chiba 277-8561, Japan, 9601548985@edu.k.u-tokyo.ac.jp), and Susumu Teramoto (Graduate School of Eng., Univ. of Tokyo, Bunkyo, Tokyo, Japan)

For rockets and supersonic aircrafts, acoustic waves from an exhaust jet cause problems of vibration and noise. To reduce the influence of acoustic waves effectively, it is important to reveal characteristics of acoustic phenomena, such as source locations and the relation between the flow and acoustic phenomena. For this purpose, the authors carried out visualization of acoustic field of a correctly expanded supersonic jet in the previous studies, and proposed the acoustic-triggered conditional sampling analysis to obtain acoustic information from high speed schlieren movies. This method is also expected to be applied to other jet acoustic phenomena. In this study, this method is applied to imperfectly expanded supersonic jets and the relation between the flow and acoustic phenomena will be discussed.

2:45–3:00 Break

3:00

1pNS6. Design and evaluation of a high-speed aeroacoustic wind tunnel. Joana Rocha (Dept. of Mech. and Aerosp. Eng., Carleton Univ., 1125 Colonel By Dr., Mackenzie Building, ME 3135, Ottawa, ON K1S 5B6, Canada, Joana.Rocha@carleton.ca)

The high-speed aeroacoustic wind tunnel (HSAWT) at Carleton University is a new facility commissioned with the purpose of facilitating experimental studies of turbulent boundary layer (TBL) induced surface pressure fluctuations. This research is primarily intended for applications related to aircraft cabin noise generation from structures exposed to high-speed flow. This open-jet, blowdown wind tunnel is a unique facility in Canada and one of a few aeroacoustic wind tunnels in the world capable of achieving speeds up to Mach 0.8. Flow is delivered from a nozzle with dimensions of 6.1 cm $\times$ 15 cm to a test section enclosed within an anechoic chamber with internal dimensions of 1.9 m $\times$ 0.88 m $\times$ 1.95 m. This study details the complete design methodology for all major wind tunnel components, including the numerical simulations performed in the validation of the designed components. Results of preliminary test section flow characterization and chamber background noise measurements are discussed. Finally, experimental results of the TBL surface pressure fluctuation spectral behavior developed over a flat test section plate are compared with established data and empirical models available in literature.

Contributed Papers

3:20

1pNS7. Resolvent analysis for jet noise source identification. Ethan M. Pickering, Georgios Rigas, Oliver T. Schmidt, and Tim Colonius (Mech. Eng., California Inst. of Technol., 1200 East California Boulevard, Mail Code MC 104-44, Pasadena, CA 91125, pickering@caltech.edu)

We use resolvent analysis and spectral proper orthogonal decomposition (SPOD) to deduce the acoustic sources for an isothermal Mach 1.5 round jet. Both physics-based resolvent analysis and data-driven SPOD (using a high-fidelity, experimentally-verified, large-eddy simulation (LES) database) provide a basis for predicting the perturbation field. Singular value decomposition of the resolvent operator based upon the LES baseflow provides optimal volumetric forcing modes, or sources, and their associated linear responses. To identify physically relevant resolvent modes, comparisons are made between the highest gain responses and the highest energy SPOD modes computed directly from LES realizations. The prevalence of the associated resolvent forcing modes in the data are then assessed by projecting them onto the full LES nonlinear terms. The resulting distributions are presented and a jet noise model leveraging these forcing statistics is discussed. [This research was supported by a grant from the Office of Naval Research (Grant No. N00014- 16-1-2445). Ethan Pickering was supported by the Department of Defense (DoD) through the National Defense Science & Engineering Graduate Fellowship (NDSEG) Program.]

3:35

1pNS8. Fundamental study on the acoustic fields caused by the interference of supersonic jet with a perforated plate. Yo Murata, Kazutoki Iwasa (The Univ. of Tokyo, 7-3-1, Hongo Bunkyo-ku, Tokyo 113-8656, Japan, ymurata@fiv.t.u-tokyo.ac.jp), Tatsuya Ishii (Japan Aerosp. Exploration Agency, Tokyo, Japan), Koji Okamoto (The Univ. of Tokyo, Kashiwa, Chiba, Japan), and SHIGEHIKO KANEKO (The Univ. of Tokyo, Tokyo, Japan)

The supersonic jet contains intensive sound sources. It is known that when this jet collides with the perforated plate, a complicated flow field is formed on the plate and a high-amplitude sound is radiated as compared with the case without the perforated plate. Attention is paid to understanding the sound generating mechanism and proposing the measures to alleviate the sound pressure levels. In this study, two types of fundamental model tests have been carried out with a simple combination of a nozzle and a perforated plate. One is acoustic measurement in an anechoic facility. Test results indicated the amplified sound pressure levels in the forward arc of the nozzle. The aperture geometry suggested certain suppression in sound pressure levels in the arc. The other test is an optical visualization with Schlieren photographs. The photographs tried to account for reflection of high-amplitude sound waves on the plate downstream the nozzle.

3:50

1pNS9. Coherence analysis of the simulated sound field of a highly-heated laboratory-scale jet. Kevin M. Leete, Kent L. Gee (Brigham Young Univ., Brigham Young University, Provo, UT 84604, kevinmatthewleete@gmail.com), Junhui Liu (Naval Res. Lab., Washington, DC), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and S. Hales Swift (Brigham Young Univ., Provo, UT)

Large eddy simulations (LES) have successfully reproduced the flow and acoustic fields generated by laboratory-scale jets. However, laboratory-scale test conditions ordinarily do not match those of full-scale military aircraft. Recently, Liu *et al.* (AIAA 2016–2125) showed that accounting for a variable ratio of specific heats leads to increased accuracy in LES simulations of heated jets. As a step towards modeling operating parameters similar to military aircraft, they produced a simulation of a highly-heated jet with a temperature ratio of seven. This paper shows the levels in the field as well as the axial and azimuthal coherence trends of this simulated jet. Large axial coherence lengths are found in the direction of maximum radiation with decreased coherence towards the sideline. Azimuthally, coherence length scales are greatest in the region of maximum radiation and generally decrease with increasing frequency. Additionally, evidence is seen for multiple self-coherent and mutually-incoherent radiation lobes within the region of maximum radiation. [Work supported by USAFRL through ORISE.]

4:05

1pNS10. Acoustic/drag interaction and its impact on future aircraft design. Christopher Jasinski (Mech. Eng., Univ. of Hartford, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com)

For several decades, acoustic liner technology has substantially reduced the noise created by commercial jet engines. Conventional placement of the perforate-over-honeycomb liner is in the fore and aft bypass duct of an engine nacelle; however, future aircraft design may incorporate liners on additional surfaces, such as the underside of wings and the fuselage. In order to use liner technology in novel locations, the impact of the porous facesheet and core of the liner on the aerodynamic drag of the craft must be fully understood. Work conducted at the University of Notre Dame has experimentally evaluated this drag, as well as the impact of acoustic fields on the aerodynamic drag produced by acoustic liners. Measurements of several quantities across the full boundary layer profile show this impact is substantial and quantifiable. Such results for several liner facesheets will be presented. Through further testing of the acoustic impedance of various facesheets at the University of Hartford, the goal of striking design balance between acoustic performance and aerodynamic reliability can become more clear. Fluid mechanic, acoustic, and direct aerodynamic measurements for several liner samples will be presented and discussed.

4:20

1pNS11. Comparison between predicted and measured Stevens perceived level of sonic booms for community response testing. William Doebler (NASA Langley Res. Ctr., M.S. 462, NASA Langley Res. Ctr., Hampton, VA 23681, william.j.doebler@nasa.gov)

In 2011, NASA conducted a flight test to study community response and perception of low amplitude sonic booms in the Waveforms and Sonic Boom Perception and Response (WSPR) experiment. Several noise monitors were placed in a 1.6 sq. km area to record the booms. In future sonic boom community response flight tests, it will not be practicable to place noise monitors at the locations of every survey respondent. Therefore, noise predictions are necessary. Using NASA's PCBoom code, the Stevens Mark VII Perceived Level (PL) was predicted at each noise monitor locations using the real aircraft trajectories and weather balloon data taken during the WSPR test. These predictions are then compared to the measured PL values. This comparison will inform the methodology for estimating noise exposure in future flight tests.

4:35

1pNS12. Updated evaluation of sonic boom noise metrics. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov), Sara R. Wilson (Systems Eng. and Eng. Methods Branch, NASA Langley Res. Ctr., Hampton, VA), and Jonathan Rathsam (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

There is no internationally agreed-upon standard noise metric that can be used to quantify sonic boom levels from overflight of supersonic aircraft. Several laboratory studies have investigated perception of sonic booms in outdoor environments, as well as in indoor environments, where transmission loss alters the spectral content and level of the sonic boom and additional factors such as secondary rattle noise and vibration affect perception. Each study has previously been evaluated separately for performance of noise metrics, and the results do not clearly indicate a preferred metric. Meta-analyses have also been performed to evaluate the performance of metrics across five of these studies. An additional sixth study of human response that incorporates rattle noise resulting from sonic booms at representative levels was recently conducted and has been added to the meta-analysis. The analysis considered thirteen metrics with three meta-analysis methodologies: individual subject ratings for indoor studies, mean ratings for indoor studies, and mean ratings for both indoor and outdoor studies. Considering both indoor and outdoor studies, this latest analysis confirmed previous results on the five metrics with highest correlation (r^2) across studies: PL, BSEL, DSEL, ESEL, and ISBAP (Indoor Sonic Boom Annoyance Predictor).

Session 1pPA

Physical Acoustics, Noise, and ASA Committee on Standards: Outdoor Sound Propagation II

Vladimir E. Ostashev, Cochair
U.S. Army Engineer Research and Developmnet Center, 72 Lyme Rd., Hanover, NH 03755

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

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

Contributed Papers

3:30

1pPA1. Frequency dependent atmospheric scattering of infrasound: Observations and modeling of controlled explosions. Catherine de Groot-Hedlin (Scripps Inst. of Oceanogr., Univ. of California at San Diego, 9500 Gilman Dr., La Jolla, CA 92037-0225, chedlin@ucsd.edu)

In deriving equations relating to the amplitude of infrasound signals from large explosions to the energy of the source, the assumption is generally made that the amplitude scales linearly with the source yield. However, results from experiments with large-yield detonations carried out at the Utah Test and Training Range (UTTR) west of Salt Lake City have cast doubt on that assumption, at least within the near field. In 2016, acoustic and infrasound sensors were placed at ranges up to 90 km east of the UTTR test site, and sources with yield ranging from 1700–17700 kg were detonated during the summertime. In several cases, detonations occurred only several hours apart. Predictable changes in amplitude might be expected, given that only small-scale spatial variations in the atmosphere occur over that time. The fact that this is not observed may be attributable to frequency dependent effects in atmospheric scattering. I report on the observations and show the degree to which amplitude variations can be predicted with numerical modeling. Numerical computations of the Navier-Stokes equations governing acoustic propagation are performed to investigate infrasound propagation for these events. The modeling allows for nonlinear propagation within an azimuthally symmetric atmospheric model, and incorporates accurate weather information.

3:45

1pPA2. Estimation of atmospheric sound transmission loss due to sea roughness by mean of an effective impedance: A numerical validation. Diego Turo (Mech. Eng., Catholic Univ. of America, The Catholic University of America, 620 Michigan Ave. NE, Washington, DC 20064, turo@cua.edu), Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), and Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, Washington, DC)

Preliminary numerical studies have been demonstrated that sea roughness introduces an excess of transmission loss of atmospheric sound propagation. However, numerical solution of the parabolic equation that describes this phenomenon is computationally intensive, particularly when the spatial domain is large (>1 km). This study evaluates the validity of implementing an effective impedance (a flat absorbing surface) in the place of a rough surface. The effective impedance is a parametric complex quantity that is different for each sea state. The parameters of the impedance are estimated by best fitting the numerical solution of a two-dimensional finite-difference time-domain solver. The effectiveness and validity of using this impedance in place of a rough surface is evaluated by direct comparison of the solutions of the Crank-Nicolson parabolic equation obtained with and without the impedance.

4:00

1pPA3. Sea surface contribution to sound transmission loss: Numerical comparison between three-dimensional and cylindrical symmetry approximation solutions. Andrea Vecchiotti, Luca Gasparini (Mech. Eng., Università Politecnica delle Marche, Senigallia (AN), Italy), Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), Joseph F. Vignola, and Diego Turo (Mech. Engeneering, The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, turo@cua.edu)

The atmospheric sound propagation over the sea depends on several factors including but not limited to temperature, wind speed, relative humidity, and roughness of the surface. The objective of this study is to determine the contribution to sound transmission loss of the sea surface roughness. The governing equations are implemented for a three-dimensional domain using a finite element method solver. The sea surface is assumed to be perfectly reflective and its shape is pseudo-randomly generated by using the Pierson and Moskowitz model. In previous studies, the cylindrical symmetry approximation has been widely used to allow a fast computation of the equations. However, the implementation of a three-dimensional domain allows a more realistic scenario and is here used as a tool to quantify the deviation from the approximated solution. An appropriate correction factor to be used when applying the cylindrical symmetry approximation is determined.

4:15

1pPA4. Experimental characterization of atmospheric boundary layer humidity profile. Teresa J. Ryan (Dept. of Eng., East Carolina Univ.,248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Joseph F. Vignola, John Judge, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents a measurement system designed to capture the humidity profile in the near-surface (up to 100 m) atmospheric boundary layer. The sensor array combines a portable static array of humidity sensors as well as GPS-synchronized pressure, temperature, and humidity sensors mounted to an unmanned aerial vehicle. These humidity measurements support an ongoing effort that is investigating the influence of humidity profile on acoustic ducting phenomena. These measurements will inform sensitivity

studies using a numerical model being developed in a parallel effort. Preliminary measurements are presented for a range of experimental conditions such as foreshore or littoral, freshwater wetland, and dry coastal plain. The aim of this overall effort is an improved numerical model of acoustic refraction and attenuation over moderate to long distances above a sea surface that accounts for sea state as well as boundary layer wind, temperature, and humidity profiles.

4:30–4:45 Break

4:45

1pPA5. On atmospheric humidity and acoustic ducts. Joseph F. Vignola, Diego Turo, John Judge (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), and Teresa J. Ryan (Engineering, East Carolina Univ., Greenville, NC)

The development of various propagation ducting phenomena (elevated or surface) has been studied extensively for electromagnetic and electro-optical wave propagation. The influence of changes in temperature, pressure, and humidity gradients on the strength and geometry of the duct in the electromagnetic propagation problem is well-documented, and there is substantial literature regarding the influence of temperature gradient on atmospheric acoustic propagation. Analytical results indicate that moisture gradients in warm air above a sea surface contribute meaningfully to the refraction of sound. This work uses numerical simulation to explore the atmospheric conditions that would support development of acoustic ducts or wave guide conditions during moderate range (<1 km) acoustic propagation. The aim of this overall effort is an improved numerical model of acoustic refraction and attenuation over moderate to long distances above a sea surface that accounts for sea state as well as boundary layer wind, temperature, and humidity profiles.

5:00

1pPA6. Measurements of the farfield characteristics of an automotive ultrasonic sensor with a vertical temperature gradient. Sang Hyun Kim, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, ares3833@naver.com), and Youngsoo Choi (Hyundai MOBIS, Yongin-si, South Korea)

With the coming of autonomous vehicles on the horizon, a variety of sensor platforms such as ultrasonic, radar, lidar, and image sensors are currently under development. Among these, ultrasonic sensors play a key role in the autonomous parking system. An ultrasonic sensor emits and receives a sound pulse (usually centered around a few dozen kHz), and computes the distance to a nearby object assuming a constant sound speed. However, the assumption of constant sound speed is often invalid because of the meteorological conditions, and could cause errors in distance estimation. In this talk, we discuss the farfield characteristics of a typical automotive ultrasonic sensor in the presence of a vertical stratification of temperature hence sound speed. The sound speed profile on a hot summer day is derived from temperature measurements with a vertical thermocouple array, and the farfield acoustic pressure is measured using a 2-D array of microphones. The measurements indicate that refraction due to the stratification of sound speed can be significant enough to alter the farfield performance of an ultrasonic sensor, and thus should be taken into account in the design of the autonomous parking system.

5:15

1pPA7. Experimental study on aerodynamic noise characteristics of high-lift configuration with variable gap leading-edge slat. Weishuang Lu, Peiqing Liu, and Hao Guo (Key Lab. of Aero-Acoust. (Beihang University), Ministry of Industry and Information Technol., Xueyuan Rd. No.37, Haidian District, Beijing 100191, China, lujiaww@163.com)

The aircraft high-lift devices are important components to ensure the safety of take-off and landing, and it is also the main source of airframe noise in the process of take-off and landing, especially the leading-edge slat. The slat that reduces gap by rotating with the fixed axis is proposed in this paper, and experimental study of far-field aerodynamic noise characteristics of high-lift configuration with the slat is also carried out. The experimental results show that discrete tones in the low-middle frequency range disappear and the broadband frequency amplitude also slightly decline, resulting in a significant decrease in the overall sound pressure level of the high-lift configuration with slat gap of 0, compared with the configuration with conventional leading-edge slat. However, in the process of rotation from conventional slat to seamless slat, more prominent discrete tone in low-middle frequency range or humps in mid-high frequency range appear at angles of attack below 6°, which give rise to the overall increase in the sound pressure level. At angles of attack above 6°, these aero-acoustic phenomena are not obvious. Hence, the overall sound pressure levels of high-lift configuration gradually decrease with the rotation process in larger angles of attack.

Session 1pPP

Psychological and Physiological Acoustics: Understanding Limitations on Auditory Spatial Acuity

Andrew D. Brown, Chair
Physiology & Biophysics, University of Colorado School of Medicine, 12800 East 19th Avenue, RC1-N, Rm 7401G, Mail Stop 8307, Aurora, CO

Chair's Introduction—2:00

Invited Papers

2:05

1pPP1. Sense, the single neuron, and multi-dimensional stimuli. Victor Benichoux (Institut Pasteur, 25 rue du Dr Roux, Paris 75015, France, victor.benichoux@pasteur.fr)

Sensory processing disorders are characterized by poor behavioral performance in psychophysical tasks relative to the general population. To assess the etiology of those deficits, a tempting possibility is to look for abnormalities in the information available in responses of specific groups of neurons, which could then become the target of rehabilitation. I discuss here basic limitations in linking behavioral performance to neuronal responses. Our understanding of sensory neurons comes from measuring *tuning curves,* which describe how the response varies when a dimension of the stimulus is varied. In reality, however, neural responses are modulated by many stimulus features, e.g., in the auditory system: amplitude and spectrum of the stimulus, spatial position, temporal properties, etc. I illustrate the challenges that this tuning heterogeneity raises when trying to predict results of behavioral experiments from neuronal responses. I first exhibit effects in theory using Fisher Information, then analyze explicit optimal decoders trained to estimate spectral and spatial properties of an auditory stimulus from neural responses. I show how heterogeneity in tuning is a major contributor to the robustness of the encoding of multiple stimulus dimensions. Finally, I described how this changes the interpretation of neural constraints on behavioral acuity.

2:25

1pPP2. Changes in the minimum audible angle may be related to systematic distortions in sound localization and motion perception. W. Owen Brimijoin (Facebook Reality Labs, MRC/CSO Inst. of Hearing Res., Fl. 3, New Lister Bldg., 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, owen.brimijoin@nottingham.ac.uk)

Systematic distortions in our perception of acoustic space have been repeatedly documented in the literature. For example, listeners exhibit a characteristic overestimation of azimuth when they are asked to point a laser at a sound source while keeping their head and eyes fixated ahead of them. Motion perception also appears to be non-uniform as a function of azimuth, as demonstrated by the fact that listeners need roughly twice as much motion at the side than at the front to judge the two motions as equivalent. This talk will make the case for a common source of these two perceptual inconsistencies, namely, the change in minimum audible angle (MAA) as a function of azimuth. Classically speaking, the MAA is thought of as a threshold measure that increases at more eccentric angles because of a change in the availability of binaural cues. Regardless of the source, we argue here à la Fechner that azimuth overestimation and motion perception non-uniformity are suprathreshold consequences of the change in acuity. By treating, e.g., 20° of acoustic space at the side as equivalent to 10° at the front, a model of the perceptual topology of acoustic space can capture several previously reported findings in spatial hearing.

Contributed Paper

2:45

1pPP3. Time-series analysis of binaural perception and uncertainty using eye movements. Matthew Winn (Speech Lang. & Hearing Sci., Univ. of Minnesota, 164 Pillsbury Ct, Minneapolis, MN 55455, mwinn83@gmail.com) and Gabrielle O'Brien (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Uncertainty affects sensory perceptions, even for stimuli well above detection thresholds. Using an anticipatory eye-movement paradigm popularized in studies of linguistic perceptual certainty and competition, we measured perception of binaural cues in dichotic narrowband noises that varied by center frequency, bandwidth, envelope fluctuations and interaural correlation. Participant's eye gaze served to mark detection of change in the ILD or ITD. We tracked the timing and direction of saccades, as well as the number of saccades ("guesses") as a metric of perceptual uncertainty. Results showed gradually increasing performance accuracy for greater cue magnitude, coupled with fewer saccades overall. There was also a change in the distribution of saccade timings and number of saccades depending on whether cues were low-frequency ITD, high-frequency envelope ITD, correlated-noise ILD, or uncorrelated-noise ILD. All ILD results were consistent with a model of detection based on interaural envelope decorrelation; performance was particularly poorer, slower, and more variable for stimuli whose envelopes had greater fluctuation, but only when channels were uncorrelated. Comparison with behavioral accuracy and reaction time will be discussed. This study highlights the potential benefits of time-series analysis in psychoacoustic experiments in order to reveal some subtle effects of stimulus properties on perceptual judgments and certainty.

3:00

1pPP4. Effects of interaural delay, center frequency, and no more than ”slight” hearing loss on binaural processing: Behavioral data and quantitative analyses. Leslie R. Bernstein and Constantine Trahiotis (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, University of Connecticut Health Ctr., Farmington, CT 06032, lbernstein@uchc.edu)

Recent neurophysiological studies concerning “hidden hearing loss” have motivated investigators to attempt to uncover behaviorally-measured auditory deficits that might exist in human listeners who would be categorized via standard audiometry as having normal hearing or, at most, “slight” hearing loss. We hypothesized that behavioral measures of binaural auditory performance could reveal the presence of deficits that may not be discoverable via standard audiometric testing or via measures of monaural processing. That hypothesis was based on the notion that monaural (perhaps, peripheral) deficits would be effectively multiplied via binaural interaction. Groups of listeners, all having no greater than “slight” hearing loss were tested. Binaural detection thresholds were elevated across a wide range of center frequencies for listeners whose absolute thresholds at 4 kHz exceed 7.5 dB HL. Quantitative predictions made via an interaural correlation-based model of binaural processing suggest that the elevated binaural detection thresholds observed for listeners having slightly elevated absolute thresholds stem solely from their having elevated levels of internal noise. They appear to stem neither from reduced sensitivity to signal-dependent changes in information derived from the stimuli as processed internally nor from increased noise along the listener’s putative internal delay line.

3:20–3:35 Break

3:35

1pPP5. Behavioral and neural spatial acuity can be persistently reduced by early temporary hearing loss. Daniel J. Tollin and Kelsey Anbuhl (Physiol. & Biophys., Univ. of Colorado School of Medicine, University of Colorado Anschutz Medical Campus, 12631 E 17th Ave., Aurora, CO 80045, Daniel.Tollin@ucdenver.edu)

Children experiencing asymmetrical hearing early in life, typically due to conductive hearing loss (CHL) associated with ear infection, often display reduced behavioral spatial acuity that persists long after hearing is restored, suggesting abnormal central auditory development. We hypothesized that persistent CHL disrupts the experience-dependent fine-tuning of binaural hearing necessary in the developing auditory system to support normal behavioral spatial acuity. Using an animal model (the guinea pig), we found that chronic unilateral CHL during development (induced by an earplug) caused the brain to generate a less precise representation of auditory space. When the hearing loss was reversed by simply removing the earplug, both the spatial acuity of single neurons in the inferior colliculus and behavioral performance in a simple sound location acuity task were ~threefold worse than animals that had not worn earplugs, as if the sense of auditory space had been blurred. Overall, the results suggest that experiencing even temporary hearing loss during early development can persistently alter the normal maturation of the auditory brain circuits that are necessary to support good spatial acuity. Thus, the maladaptive plasticity in these circuits due to temporary hearing loss during development can place limitations on spatial acuity in adulthood. [Support: R01-DC011555, T32-DC012280, F31-DC014219.]

3:55

1pPP6. Does electrode placement affect the interaural-time-difference acuity in bilateral cochlear-implant listeners? Olga A. Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland, College Park, 4954 North Palmer Rd., Bldg. 19, R. 5607, Bethesda, MD 20889, olga.stakhovskaya.ctr@mail.mil), Joshua G. Bernstein (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Jack H. Noble (Dept. of EECS, Vanderbilt Univ., Nashville, TN), Kenneth K. Jensen (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Eden Prairie, MN), Michael Hoa, Hung J. Kim (Dept. of Otolaryngology-Head and Neck Surgery, Georgetown Univ. Medical Ctr., Washington, DC), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD)

Postlingually deafened bilateral cochlear-implant (CI) listeners can show limited interaural-time-difference (ITD) sensitivity, even when tested using highly controlled time-synchronized research processors. It is assumed that ITD discrimination requires interaural frequency-matched inputs. However, current bilateral CI programming procedures do not account for potential interaural place-of-stimulation mismatch. This study investigated the magnitude of interaural place-of-stimulation mismatch and its effects on ITD sensitivity in bilateral CI listeners. Ten bilateral CI listeners were tested on a two-interval left-right ITD discrimination task. Loudness balanced, 300-ms, 100 or 200 pulse-per-second, constant-amplitude, monopolar pulse trains were delivered to single-electrode pairs using time-synchronized research processors. ITD just noticeable differences (JNDs) were measured for five reference electrodes evenly distributed along the array in one ear, and for at least five comparison electrodes, generating ITD tuning curves. The interaural mismatch was estimated using both the ITD tuning curves and differences in the angular insertion depth from computed-tomography (CT) scans. Data showed that ITD tuning curves were relatively broad when compared to the amount of physical interaural place-of-stimulation mismatch from the CT scans. This suggests that most bilateral CI listeners do not experience appreciably reduced binaural sensitivity due to differences in electrode placement.

Contributed Paper

4:15

1pPP7. Lateralization of interaural time and level differences measured with cochlear implant sound processors. Alan Kan and Ruth Litovsky (Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu)

Recently, our lab developed a novel technique that enables binaural sensitivity to be measured via unsynchronized clinical cochlear implant (CI) sound processors. By using a transposed-tone complex with carrier frequencies at the center frequency of the patient MAP, bilateral CI users have measurable sensitivity to interaural time and level differences (ITD and ILD, respectively) that is on par with results from bilaterally synchronized research platforms. In this work, we measure the extent to which a transposed-tone complex presented via clinical processors can be mapped to lateral locations in the head as a function of changes in ITD and ILD. In addition, lateralization was measured when ITDs and ILDs co-varied. Results show that listeners are able to lateralize the tone complex throughout the range of the head. However, the slope of the ITD lateralization function appears to be shallower than that of ILDs. When ITDs and ILDs co-vary, the slope of the lateralization function appears to follow that of ILDs, suggesting that bilateral CI users may not be combining the two cues for sound source lateralization.

Invited Paper

4:30

1pPP8. Exploiting spatial audio cues for pilot navigation in degraded visual environments. Heath Jones, Lana Milam, Stephanie Karch (APPD, USAARL, Bldg. 6901, Fort Rucker, AL 36362, heath.g.jones2.ctr@mail.mil), Henry Williams (Naval Medical Res. Unit Dayton, Dayton, OH), and Brian Simpson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Aviation mishaps resulting from degraded visual environments (DVEs) represent a significant loss in military personnel and aircraft every year. DVEs are considered any type of environmental condition (e.g., sand, snow, or fog) that visually obstructs the pilot and can cause spatial disorientation. The current study represents a tri-service effort exploring the implementation of spatial audio cueing techniques to aid pilot navigation in DVEs. Directional cueing (i.e., indicating the location of target waypoints) was achieved by spatializing an auditory stimulus using the SoundLab audio rendering package and convolving audio signals with (non-individualized) head-related transfer functions. Two spatial cue conditions were tested, either rendered dynamically in reference to the pilot's head via head tracking or with respect to aircraft heading. Data were collected from pilots operating a full-motion UH-60 Black Hawk flight simulator at the U.S. Army Aeromedical Research Laboratory. Pilots performed multiple flight maneuvers, such as a "turn to target" localization task, a side-step maneuver positioning the aircraft over a stationary target, and an "approach to moving target" task. Performance was assessed by measures of localization error, completion time, and failure rate. Findings from this study provide information on sensory cueing display countermeasures for helicopter flight in DVEs.

Session 1pSA

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Advances in Thermoacoustics

Matthew Kamrath, Cochair
Signature Physics Branch, US Army ERDC-CRREL, 72 Lyme Road, Hanover, NH 03755

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Invited Papers

1:00

1pSA1. Thermoacoustic waste heat recovery engine. Comparison of simulation and experiment. Thomas W. Steiner and Maarten Elferink (Etalim, 62 West 8th Ave., Ste. 400, Vancouver, BC V5Y-1M7, Canada, tsteiner@etalim.com)

A thermoacoustic engine heated with a 10–20 kW 600C waste heat exhaust stream was designed and built. The heat is extracted from the exhaust with a microchannel counter flow heat exchanger and delivered to the hot side of the regenerator with a self-ciculating loop. This loop is driven using Venturi driven streaming of the helium working gas so that no high pressure and high temperature pump is required. The prototype system performed close to simulated expectations. A peak of 570 W electrical power was measured as was a peak conversion efficiency of waste heat enthalpy to electrical energy of 5%.

1:20

1pSA2. Monocoque shell and tube heat exchanger for thermoacoustics. Robert M. Keolian, Kevin J. Bastyr, Ray S. Wakeland, and John F. Brady (Graduate Program in Acoust., The Penn State Univ., 732 Holmes St., State College, PA 16803, keolian@psu.edu)

Heat exchangers can be a limiting factor to thermoacoustic power density. They can also be difficult to construct because of a tendency toward many small parts due to the small size of the thermal penetration depth. We present a heat exchanger that consists of around five pieces and yet has very small feature size and is easy to construct in large numbers. A sacrificial mandrel, consisting of a sheet with a lattice of specially shaped holes, is formed by casting in a mold with a lattice of specially shaped pins. The mandrel is plated with nickel, electroless nickel, or other metal. The mandrel is then removed by melting or dissolving, leaving a monocoque tube bundle. A screen mesh may be joined to one or both faces of the monocoque to act as fins to improve thermal contact to the working fluid that flows through the tubes. Coolant flows around the tubes within the monocoque shell. For a typical device, we find that without a mesh on the monocoque, lost work is minimized with tubes on the order of 8 mm long and spaced by 0.4 mm, but with a 100-mesh copper screen on both faces, the tubes can be a more manageable 3 mm long spaced by 4 mm.

1:40

1pSA3. Standing-wave and traveling-wave thermoacoustics in solid media. Haitian Hao, Carlo Scalo (Mech. Eng., Purdue Univ., Herrick Labs, 177 S. Russell St., Rm. 1007, West Lafayette, IN 47906, haoh@purdue.edu), Mihir Sen (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN), and Fabio Semperlotti (Mech. Eng., Purdue Univ., West Lafayette, IN)

Thermoacoustics is a multi-physics process in which heat and mechanical power (associated with waves) can be converted into one other. This process has been exploited to design different types of modern-day energy conversion devices, such as thermoacoustic engines (TAE) and refrigerators (Swift, 1988), since the first experimental assessment of the phenomenon by Soundhauss (1850) in the glass blowing process. To date, all the thermoacoustic devices are fluid-based, using mostly air or Helium as the working medium. Our study explores for the first time the possibility of achieving thermoacoustic energy conversion in solid media, by laying out the fundamental theory and showing numerical evidence of the existence of both standing and traveling thermoacoustic waves in solids. Consistent with established results for fluid-based TAEs, the growth-rate-to-frequency ratio of traveling waves in solids is found to be significantly higher than that of standing waves. While sold-state thermoacoustics (SSTA) share some commonalities with their fluid counterpart, some important and noteworthy distinctions are present. For example, solids have the potential to be highly engineered (e.g., metamaterials), with their properties tuned to enhance thermoacoustic energy conversions. The theoretical investigation of this mechanism may motivate novel ideas designing a new generations of ultra-compact, highly efficient thermoacoustic devices.

2:00

1pSA4. Study on a coaxial thermoacoustic system: Effect of the edge shape on the acoustic intensity. Yukihiro Takeyama (Elec. and Electron. Engineering, Doshisha Univ., 1-3 Tatara-miyakodani, Kyotanabe-city, Kyoto, Kyotanabe 610-0321, Japan, ctwb0351@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Electron. System Eng., Univ. of Shiga Prefecture, Hikone, Japan), Masaya Saeki (Elec. and Electron. Engineering, Doshisha Univ., Kyoto, Japan), Daichi Kuroki, and Yoshiaki Watanabe (Biomedical Information, Doshisha Univ., Kyotanabe, Japan)

For the downsizing, the coaxial thermoacoustic system (coaxial-type) has been proposed. The coaxial-type is expected to have high efficiency because the energy conversion with traveling wave phase like looped-tube is realized. Additionally, this system is more compact than the looped-tube. The circular flow path of the coaxial-type is made of an annular flow path between the inner and the outer tube and a cylindric flow path in the inner tube. By propagating in the cylindric flow path after passing through the annular flow path, a traveling-wave sound field is generated. The edge space between the annular flow path and the inner tube was studied due to the reduction of the propagation loss. The influence of the inner tube length and the edge shapes of outer tube on the energy conversion was examined. The two edge shapes were used: the flat one and the spherical one. It was confirmed the sound energy generated in the system with the spherical shape is higher than the one with the flat shape. It is suggested a traveling-wave sound field is easily generated by the spherical shape because of the focus effect. Thus, the edge shape is one of important points for designing the coaxial-type.

2:20

1pSA5. Effect of the resonance frequency on the work flow with adjusting the internal temperature of the stack in a standing-wave thermoacoustic-system. Mana Sugimoto (Doshisha Univ., Tatara Miyakodani 1-3, Kyotanabe, Kyoto 610-0321, Japan, ctwb0349@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Pref., Hikone, Shiga, Japan), Yuya Kurata, Yuto Kawashima, and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe, Kyoto, Japan)

We discuss the stack which is the key device of the thermoacoustic system. We elucidate the stack mechanism to realize further energy conversion efficiency improvement. We propose installing a heater inside the stack. This heater adjusts the internal temperature of the stack and controls the heat flow in the stack. Controlling the heat flow in the stack can also affect the mutual conversion of the heat flow and the work flow. In this report, the effect of the resonance frequency with the standing-wave thermoacoustic-system and the internal temperature of the stack on the heat flow in the stack was experimentally investigated. The resonance frequency was changed with the total length of the system. As a result, the lower the frequency, the larger the amount of the work flow generated in the stack. This can be influenced by the increase and decrease of the dissipation with changing the resonance frequency. Also, at any resonance frequency, the amount of the work flow generation increased with controlling the heat flow in the stack. The heat flow can be controlled by adjusting the internal temperature of the stack at any resonance frequency.

2:40

1pSA6. High power-density thermoacoustic sound projectors. Ali E. Aliev (Alan G. MacDiarmid Nanotech Inst., Univ. of Texas at Dallas, P.O.Box 830688, BE 26, Richardson, TX 75083, Ali.Aliev@utdallas.edu)

Design, fabrication, and performance of large size and high power-density encapsulated thermoacoustic sound projectors driven by short pulses will be presented. The analysis of enhanced energy conversion efficiency using proper signal processing and low-dimensional thermodynamics will be given. [Work supported by the Office of Naval Research and Army Research Office.]

3:00–3:15 Break

3:15

1pSA7. Investigations in performance of a gas-combustion infrasound source. Chad M. Smith and Thomas B. Gabrielson (Graduate Program in Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, chad.smith@psu.edu)

Designing a coherent (non-explosive) sound source within the infrasound band (0.1–4 Hz in our case) is a difficult engineering challenge. This is due to the large air mass that must be moved to created useful signal levels. The fundamental simple source equation, which will govern almost any human-made infrasound source due to the long wavelengths, shows this fundamental difficulty. As frequency decreases, the volume velocity must increase by the inverse factor of the frequency decrease in order to maintain an equal pressure amplitude at equal range. In other words, 1000 times the cubic volume of air must be moved at 0.1 Hz as compared to 100 Hz, in order to maintain an equal output pressure amplitude! For this reason, a novel method has been proposed to skirt this engineering challenge; using the large energy density available in gas combustion for periodic thermal expansion of an air mass. This talk builds on previous work by the authors (Smith and Gabrielson, J. Acoust. Soc. Am. **137**, 2407, 2015 and Smith and Gabrielson, J. Acoust. Soc. Am. **143**, 1808, 2018) and presents comparisons of a 1^{st}-order thermodynamic model and empirical measurements using a large liquid-propane burner system from a hot air balloon.

3:35

1pSA8. Modeling thermal effects in the time domain. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu)

Modeling thermal transfer from fluid to a solid surface is well understood. However, the mathematical description of this behavior includes transcendental functions in the frequency domain. This precludes some kinds of modeling in the time domain. In some simple geometries, however, it has been shown that a continued fraction expansion of the transcendental functions is possible that allows for a finite bandwidth solution in the time domain. This paper will explore the use of these methods for time domain modeling of thermal-acoustic systems.

3:55

1pSA9. Phase diagram of coupled thermoacoustic chaotic oscillators. Rémi A. Delage, Yusuke Takayama, Sato Moeko, Hiroaki Hyodo (Mech. Eng., Tohoku Univ., Adv. Mech. Systems and Design, Graduate School of Eng., Tohoku University 6-6-01, Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, delageremi@hotmail.fr), and Tetsushi Biwa (Mech. Eng., Tohoku Univ., Sendai, Miyagi, Japan)

Linear analysis of thermoacoustic oscillations have been extensively made, but there is still a lack of knowledge on nonlinear phenomena such as synchronization and chaotic oscillations. Thermoacoustic chaotic oscillations have been reported in Taconis and combustion oscillators but these systems do not allow a well-controlled experiment because of the extreme temperature conditions. By making the system acoustically dissonant, we recently succeeded in generating thermoacoustic chaotic oscillations with modest temperatures. In this study, we present spatiotemporal evolution of thermoacoustic chaos from pressure measurements, and report an experimental bifurcation diagram when two thermoacoustic chaotic oscillators are dissipatively coupled to each other. The two-parameter bifurcation diagram is constructed by varying the frequency mismatch and the coupling strength. Complete chaos synchronization is observed in the region with a frequency mismatch of less than 1% of the uncoupled oscillator, with on-off intermittency phenomenon for low coupling strength. In other regions, synchronization between quasiperiodic oscillations and that between limit-cycle oscillations and oscillation death are observed as well as asynchronous states.

Contributed Papers

4:15

1pSA10. Passive structural monitoring based on matched processing. Emma Lubeigt (Scripps Inst. of Oceanogr., Univ. of California San Diego, 8820 Shellback Way, La Jolla, CA 92037, elubeigt@ucsd.edu), Sandrine Rakotonarivo, Serge Mensah, Jean-François Chaix (Aix-Marseille Université, CNRS, Centrale Marseille, Lab. of Mech. and Acoust., Marseille, France), William A. Kuperman, Jit Sarkar (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), François Baqué, and Gilles Gobillot (CEA Cadarache, Saint-Paul-Lez-Durance, France)

The fourth generation of nuclear reactor designs use liquid sodium as the core coolant. Due to both significant primary loads and thermal variations, important supporting structures are susceptible to thermo-mechanical fatigue. Periodic inspections are planned in order to check the health of the immersed structures, particularly to detect or to characterize potentially hazardous damage. The presented approach relies on both the *a priori* knowledge of the inspected structures (geometry and properties) and the low-frequency ambient noise in the reactor vessel. Indeed, the latter is used as a source of opportunity to inspect the immersed elastic structures. Two kinds of potential defects could occur: localized defects (e.g., fatigue cracks) and global defects (e.g., deformation of the structure). The feasibility of using data derived from ambient noise for detecting and localizing these defects is carried out on an academic example: a cylindrical shell immersed in water. A cross-correlation based procedure is used to construct the replica and datasets. Then, techniques, such as Matched Field Processing and Matched Mode Processing, are investigated to detect and locate the damage. Both conventional and adaptive processors are implemented and their performance compared. Presented results are obtained from numerical data.

4:30

1pSA11. Suspension optimization for a hermetic compressor assembled on a refrigerator. Alexandre A. Pescador Sarda (DEMEC, UFPR, Av. Cel. Francisco H. dos Santos, 100, Curitiba, PR 81530000, Brazil, pescador@ufpr.br)

Noise annoyance generated by hermetic compressors is evaluated based on the sound power level (SPL) parameter measured in a reverberation room or using a semi-anechoic chamber. However, when the machine is assembled on a refrigerator, the SPL can be altered depending on the new system configuration and the way the machine is assembled in the final product. The vibration generated in the electrical machine is transmitted to the refrigerator resulting in noise at the surface. The aim of this study was to model the suspension of a compressor assembled in a flexible base and minimize the vibratory power flow transmitted to the base through an optimization process, taking into account parameters such as the spring position. Decreasing the power flow to the refrigerator base results in a reduction in the global levels of noise and vibration at the base plate.

4:45

1pSA12. Artificial Neural Network-DeltaEC Hybrid Model for Thermoacoustic Systems: New Synergistic Approach. Anas M. Abdelrahman and Xiaoqing Zhang (Dept. of Refrigeration and Cryogenics, School of Energy and Power Eng., Huazhong Univ. of Sci. and Technology, 1037#, Luoyu Rd., Hong shan District, Wuhan, China, Wuhan 430074, China, arahman@hust.edu.cn)

Despite the wide-spread use of DeltaEC model in thermoacoustics community, intrinsic nonlinear phenomena are still hindering its applicability to real practical situations due to assumptions and limitations. In the present study, artificial neural network (ANN) as an intelligent technique is hybridized with conventional DeltaEC model to provide a new synergistic approach called (ANN-DeltaEC) hybrid model for thermoacoustic research field. The aim of this paper is to improve the prediction accuracy of single DeltaEC model by integrating it with distributed and synergistic neural networks. One application for this new approach has been conducted on one standing wave thermoacoustic heat engine based on published literature work to predict the acoustic wave parameters, namely, oscillating frequency and acoustic pressure amplitude under given design considerations of stack geometry and resonator length. The results from hybrid synergistic model had been proven to be desirable in its accuracy compared to experimental work and better than the results of DeltaEC model itself. The present work had shown the capability of the new synergistic approach in accurately predicting the outputs for any new given inputs within given range. Further applied research will be devoted in order to identify complex mappings between thermoacoustic system parameters and their corresponding responses.

Session 1pSCa

Speech Communication, Psychological and Physiological Acoustics: Coupling Phonetics and Psycholinguistics I

Ann Bradlow, Cochair
Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL

Yue Wang, Cochair
Linguistics, Simon Fraser University, 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada

Ratree Wayland, Cochair
Linguistics, University of Florida, 2801 SW 81st Street, Gainesville, FL 32608

Chair's Introduction—1:00

Invited Papers

1:05

1pSCa1. Phonetics, psycholinguistics, and language-specificity. Ann Bradlow (Dept. of Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

How does linguistic knowledge influence speech production, perception, and learning? In this presentation, I will review three sources of empirical evidence for language specificity in some of the most basic processes of speech communication. Each line of evidence will highlight the contributions of the remarkable Jongman-Sereno collaboration—the honorees of this special session—and their continuing influence on generations of students. First, we will consider language specific phonetics as it relates to the acoustic details of speech categories across languages with different sound inventories, including Modern Greek, German, English, and Spanish. Second, we will examine the influence of language background on speech perception through comparisons of pitch perception by speakers of languages with and without lexical tone. Finally, we will focus on speech training in adults when presented with novel speech patterns, asking in particular how general learning principles interact with language-specificity for perceptual adaptation to second-language or foreign-accented speech. In each of these domains—production, perception, and learning—we will see the expanding circles of interaction amongst Jongman, Sereno, and their many students and colleagues. Critically, these Jongman-Sereno contributions underscore the importance of cross-language comparison coupled with a highly collaborative research approach for understanding speech processing and learning.

1:25

1pSCa2. Experience, attention, and context in the processing of systematic variation in spoken language. Lynne C. Nygaard (Dept. of Psych., Emory Univ., Atlanta, GA 30322, lnygaar@emory.edu)

The acoustic speech signal is characterized by enormous variability. Characteristics of individual speakers and groups of speakers profoundly change the way in which linguistic structure is realized. A substantial body of research suggests that language users track, retain, and use systematic variation to restructure linguistic representation and processing in order to maximize intelligibility of spoken language. Less clear is whether sources of variation differ in relevance during speech processing and how relevance changes as a function of experience and context. Research will be presented examining talker-, task-, and listener-related factors that mediate memory and learning of systematic variation in spoken language. The findings suggest that although listeners dynamically adapt to systematic changes in linguistic structure as a function of experience, this adaptation depends on the characteristics and frequency of particular sources of variations, the modulation of attention driven by the structure of the learning environment, and expectations and subsequent sensitivity to socially relevant variation. The considerable behavioral and representational plasticity that is characteristic of speech perception and spoken language processing may depend in part on the social, linguistic, and contextual relevance of the variation associated with both individual talkers and classes of talkers.

1:45

1pSCa3. Learning foreign-language sounds in adulthood: Listening, speaking, and individual differences. James M. McQueen, Jana Krutwig (Donders Inst. for Brain, Cognition and Behaviour, Radboud Univ., Montessorilaan 3, 6525 HR, Nijmegen 6525 HR, Netherlands, j.mcqueen@donders.ru.nl), Lisette Jager (Ctr. for Linguist, Leiden Univ., Leiden, Netherlands), Peter Desain (Donders Inst. for Brain, Cognition and Behaviour, Radboud Univ., Nijmegen, Netherlands), Jurriaan Witteman, and Niels 0. Schiller (Ctr. for Linguist, Leiden Univ., Leiden, Netherlands)

Adult native Dutch speakers tend to have difficulty learning the English /æ/-/ɛ/ contrast because both English vowels can be assimilated to Dutch /ɛ/. Two experiments examined how this contrast is acquired by Dutch adults and the relationship between perception and production in this process. In Experiment 1, a four-day perceptual training protocol on the /æ/-/ɛ/ contrast was combined with related or unrelated production practice (participants said the target word or a phonologically unrelated word after each perceptual decision). There was improvement over the four days in perception and production, but no effect of type of production practice. Perceptual training can thus boost production learning even when participants have to produce the new vowels. Some individuals, however, are more successful in acquiring foreign sounds. In Experiment 2, Dutch students were followed longitudinally over their first year studying English at university. Preliminary results indicate that, in a passive oddball paradigm testing for a MisMatch Negativity (MMN) effect, students could discriminate English /æ/ from English /ɛ/ already at initial test, but Dutch /ɛ/ from English /ɛ/ only at final test. We will ask whether students who show a larger increase in MMN over time improve more in pronunciation of /æ/ and /ɛ/.

2:05

1pSCa4. Evidence for a 3-component model of speech production with phonology-extrinsic timing. Alice Turk (The Univ. of Edinburgh, PPLS, 3 Charles St., Edinburgh EH10 4ED, United Kingdom, a.turk@ed.ac.uk) and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

This talk will present evidence supporting a model of speech production which includes phonology-extrinsic timing, and consists of 1) a Phonological Planning Component to plan the goals for an utterance, including the segmental and prosodic structure for an utterance and non-grammatical goals such as speaking quickly or in a particular style, 2) a Phonetic Planning Component to plan the quantitative details of the acoustic goals and how they will be achieved articulatorily, and 3) a Motor-Sensory Implementation Component to ensure that the goals are achieved on time. Supporting evidence from the literature includes findings of greater timing precision at movement endpoints compared to other parts of movements, suggesting the separate control of the timing of movement endpoints compared to other parts of movement. This evidence presents a challenge to models in which all parts of a movement trajectory are controlled by the same equation of motion, but is consistent with models in which 1) abstract, symbolic phonological representations map onto spatial and temporal characteristics of movement endpoints, 2) movements are planned to reach the endpoints on time, and 3) speakers give priority to the accurate implementation of the part(s) of movement most closely related to the phonological goals.

2:25

1pSCa5. What the /f/? We're not done with fricatives yet. Integrating across time and frequency bands. Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, E11 SSH, Iowa City, IA 52242, bob-mcmurray@uiowa.edu), Marcus Galle (Psychol. Sci., Univ. of Texas Rio Grande Valley, Brownsville, TX), Ashley Farris-Trimble (Linguist, Simon Fraser Univ., Iowa City, IA, Iowa), and Michael Seedorff (Biostatistics, Univ. of Iowa, Iowa City, IA)

Fricatives represent an extreme version of lack of invariance. This illustrated by Jongman, Wayland and Wong's (2000) singularly comprehensive look at the acoustics of a set of speech sounds. They show that fricatives require dozens of cues; and each cue is affected by multiple context factors. Later work with this corpus solved these problems with simple models. But we're not done yet. I present new studies that raise two additional problems highlighted by fricatives. First, listeners must integrate cues over long timescales: frication cues may arrive several hundred milliseconds before the vocoid. I present several experiments that used eye-tracking to determine when cues are used. Unlike, prior results with stops and vowels, they show evidence for an encapsulated memory buffer preceding lexical access—lexical decisions are delayed until both cues arrive. Second, for fricatives, listeners must integrate information across distinct frequency bands. This was highlighted in a study of hybrid cochlear implant users who integrate low frequency residual acoustic hearing with high frequency electric hearing. Unexpectedly, hybrid listeners performed worse on fricatives than those with only electric hearing. Thus, the problem of combining information across frequency bands may be suboptimally solved in acoustic + electric CI users, suggesting new principles of cue-integration.

2:45

1pSCa6. Coupling tonetics and perceptual attunement: The psychophysics of lexical tone contrast salience. Denis K. Burnham (MARCS Inst. for Brain, Behaviour and Development, Western Sydney Univ., Locked Bag 1797, Penrith South, Sydney, NSW 1797, Australia, denis.burnham@westernsydney.edu.au) and Leher Singh (Psych., National Univ. of Singapore, Singapore, Singapore, Singapore)

As with the coupling phonetics and psycholinguistics by such luminaries as Sereno and Jongman, we couple (a) discrepant cross-lab results in infant lexical tone perceptual attunement studies and (b) an adult cross-language lexical tone perception study. Three trajectories of lexical tone perception over infants' first year have been found over labs, in which infants from differing language backgrounds have been tested with differing stimulus materials, viz., (i) Incremental (Singh's lab—Singaporean Mandarin vs Singaporean English language background infants, Singaporean Mandarin tone contrasts; Tsao's lab-Taiwanese and English language infants, Taiwanese Mandarin contrasts); (ii) Decremental (Burnham's lab—Australian English but not for Cantonese or Mandarin language infants, Thai contrasts; Yeung/Werker's lab—American English but not for Mandarin and Cantonese language infants, Cantonese contrasts); and (iii) U-shaped (Kager's lab—Dutch infants, Mainland Mandarin contrasts). To investigate these discrepancies, we tested discrimination of Singaporean Mandarin, Beijing Mandarin, Hong Kong Cantonese, and Bangkok Thai tone contrasts by Singaporean Mandarin, Beijing Mandarin, Hong Kong Cantonese, Bangkok Thai, and Sydney Australian English listeners. Despite some discrepancies, this coupling revealed some congruences between psychophysical level of difficulty of tone contrasts within and between languages in the adult study, and the different developmental trajectories in the infant studies.

Session 1pSCb

Speech Communication, Psychological and Physiological Acoustics: Coupling Phonetics and Psycholinguistics II (Poster Session)

Ann Bradlow, Cochair
Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL

Ratree Wayland, Cochair
Linguistics, University of Florida, 2801 SW 81st Street, Gainesville, FL 32608

Yue Wang, Cochair
Linguistics, Simon Fraser University, 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada

All posters will be on display and all authors will be at their posters from 3:30 p.m. to 5:30 p.m.

Contributed Papers

3:30

1pSCb1. Variability in speaking rate of native and non-native speakers. Melissa M. Baese-Berk, Kayla Walker (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu), and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

The bulk of the work on non-native speech has focused on average differences between L1 and L2 speakers. However there is growing evidence that variability also plays an important role in distinguishing L1 from L2 speech. While some studies have demonstrated greater variability for non-native than native speech (e.g., Baese-Berk & Morrill, 2015; Wade *et al*., 2007), others have demonstrated that under some circumstances non-native speech maybe *less*variable and that variability in non-native speech may shift as a function of many factors, including task (Baese-Berk & Morrill, to appear; Baese-Berk, Morrill, & Bradlow, 2016) and L1-L2 pairing (Vaughn, Baese-Berk, & Idemaru, to appear). In this study, we ask how variability manifests in L1 and L2 speech by speakers from a variety of language backgrounds. Specifically, we ask whether a speaker whose L1 speaking rate is highly variable is also highly variable in their L2. We also ask whether variability in speaking rate in L1 or L2 differs as a function of task (e.g., read vs. spontaneous speech) and complexity of the task (e.g., more or less complicated reading passages). The results of this study will inform our understanding of the myriad complex factors that influence non-native speech.

1pSCb2. Does native language temporal experience transfer to audio-visual synchrony perception? Dawn Behne (NTNU, Psych., Trondheim NO-7491, Norway, dawn.behne@svt.ntnu.no) and Yue Wang (SFU, Burnaby, BC, Canada)

The temporal alignment of what we hear and see is fundamental for the cognitive organization of information from our environment. Research indicates that a perceiver's experience influences sensitivity to audio-visual (AV) synchrony. We theorize that experience that enhances sensitivity to speech sound distinctions in the temporal domain would enhance sensitivity in AV synchrony perception. With this basis, a perceiver whose native language (L1) involves duration-based phonemic distinctions would be expected to be more sensitive to AV synchrony in speech than for an L1 which has less use of temporal cues. In the current study, simultaneity judgment data for the syllable /ba/ were collected with 23 steps of AV alignments: from audio preceding the video (audio-lead) to the audio and video being physically aligned (synchronous) to video preceding the audio (video-lead). Two groups of participants differing in L1 experience with phonemic duration were included: native speakers of Norwegian (binary phonemic quantity distinction) and English (no phonemic quantity distinction). Preliminary results based on measures the audio-lead threshold (ALT) support the hypothesis that native language experience may influence broad mechanisms of timing, such as those moderating AV synchrony perception. Findings contribute to understanding the underpinnings of experience and AV synchrony perception.

Invited Paper

1pSCb3. Building a multilingual ultrasound corpus. Kelly Berkson, Kenneth de Jong, Steven M. Lulich, and Malgorzata E. Cavar (Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

This paper presents a corpus of three dimensional ultrasound data focused on tongue shape during speech sound articulation. Our ultimate goal is to collect data pertinent to phonetic structures from as many languages as possible (at present upwards of 20 languages are represented) and to curate these data in an open access corpus that is freely available for use by other researchers. In this presentation, we review the structure of the corpus and present a series of case studies illustrating the ways in which three-dimensional data are being used to address questions of phonetic interest. Examples of such areas include: examining the articulation of laterals; determining the point of articulation for dorsal consonants; analyzing consonant and vowel coarticulation patterns; and elucidating how coupling of the tongue body with the tongue blade and root affect place and manner of articulation. Like the work of Allard Jongman and Joan Sereno, this project is inherently collaborative and includes ample opportunity for using guided investigation of targeted research questions to ease novice scholars into the research process.

Contributed Papers

1pSCb4. The role of social expectation in the perception of gay speech. Dominique A. Bouavichith (Linguist, Univ. of Michigan, Lorch Hall #455C, 611 Tappan St., Ann Arbor, MI 48109, dbouavichith@gmail.com)

Previous sociophonetic studies have characterized lengthened /s/ as one of several acoustic correlates of gay(-sounding) speech (Linville 1998; Rogers *et al.* 2000). It has also been shown that listeners adjust their perceptual expectations when given social information about a speaker (Strand & Johnson 1996; Nidzielski 1999; McGowan 2015). There is evidence that /s/ duration serves as a cue in the perception of a speaker's sexual orientation, but only when paired with another acoustic cue (Levon 2007). Can a social cue serve this role instead? Given these findings, how might the timecourse of lexical activation be affected as listeners are given social information about a speaker? The present investigation addresses these questions using eye tracking in a visual world paradigm to examine the effect of social expectation on the timecourse of lexical activation. Participants (N = 22) heard /CVs/ and /CVsC/ words with digitally lengthened /s/, in two test conditions. First, participants had no social information about the speaker; second, the speaker's sexuality was given implicitly. Gaze patterns differed based on listeners' experience levels with gay speech, with a higher latency to response for high-experience listeners. This suggests that social expectation varies as a function of listener experience with a sociolinguistic variety.

1pSCb5. Retention of speech sound production and perception in young international adoptees. Mirjam Broersma (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, P.O. Box 9103, Nijmegen 6500 HD, Netherlands, m.broersma@let.ru.nl), Wencui Zhou (Max Planck Inst. for PsychoLinguist, Nijmegen, Netherlands), and Anne Cutler (Western Sydney Univ., Sydney, NSW, Australia)

While international adoptees commonly report not remembering their birth language, studies have shown that with re-exposure they learn to perceive birth-language sounds faster (Choi *et al*., PNAS, 2017; Pierce *et al*., PNAS, 2014, Singh *et al*., DevSci, 2011) and to pronounce them more accurately (Choi *et al*., RSOS, 2017) than non-adopted controls. We assessed birth-language memories in much younger adoptees than tested before, investigating whether memories were episodic or abstract in nature and which process—imitation or perception—survived longest. Participants were (Experiment 1) 21 Cantonese and (Experiment 2) 25 Mandarin adoptees in the Netherlands and 47 Dutch control children, aged 4-11. They were trained on perception of Cantonese/Mandarin affricate and tone contrasts, and tested on perception and production (imitation; recordings assessed by native-speaker identification and rating). For perception, adoptees initially performed similar to controls but outperformed them after training, similar to findings for adoptees tested as adults. For production, however, adoptees already outperformed controls from the pre-test, in contrast to previous results for adult adoptees. Thus, while international adoptees retain abstract memories of their birth language phonology which helps them in relearning the sounds later in life, imitation ability stands the test of time better than perception ability.

1pSCb6. English listeners categorize murmured stops based on aspiration, not prevoicing. Luca Cavasso and Henny Yeung (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, lcavasso@sfu.ca)

Previous perceptual studies of English stop voicing focus on Voice-Onset Time (VOT). Aspiration is generally subsumed into VOT, yet [1] complicates this, evincing a trading relation between intensity of aspiration noise and VOT. Our study is the first to examine the role of VOT and aspiration in English listeners' perception of non-native plain and murmured stops. We recorded Marathi talkers producing /CVsV/ nonce words beginning with /t/, /t^h/, /d/, /d^h/, e.g. /d^{h}aːsaː/, or their velar counterparts. The following acoustic measures were taken for each token: — Duration of prevoicing — After Closure Time (ACT) [2], i.e., the interval between release and periodicity — Pre-Vocalic Interval (PVI) [3], which includes ACT and the murmured portion of the vowel — Intensity of aspiration noise Canadian English listeners rated the stops on a 6-point scale from voiced to voiceless. Only results for murmured stops are discussed here. Despite variability of prevoicing duration in these tokens (range = 200ms, sd = 43), the factor did not correlate significantly with a token's average rating ($p>0.60$). However, ACT (r=.53, $p<0.001$), PVI (r=0.39, $p<0.001$), and the mean intensity of aspiration noise ($r = 0.58$, $p<0.02$) did. Thus, aspiration, not prevoicing, best accounts for perceptual differences between murmured stops. [1] Repp, "Relative amplitude of aspiration noise…," *Lang. Speech* 22, 1979; [2] Mikuteit & Reetz, "Caught in the ACT…," *Lang. Speech* 50, 2007; [3] Berkson, "Capturing breathy voice…," *Kans. Work. Pap. Ling.* 33, 2012.

Invited Paper

1pSCb7. Linguistic experience and musical training in shaping Mandarin tone perception by trilingual non-native Cantonese listeners. Si Chen (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong 00000, Hong Kong, qinxi3@gmail.com), Yike Yang (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), Ratree Wayland, and Yiqing Zhu (Univ. of Florida, Gainesville, FL)

Mandarin tones are perceived categorically by native listeners, but not by non-native listeners (e.g., Francis *et al*., 2003; Hallé *et al*., 2004; Xu *et al*., 2006). Vowel quality, stimulus duration, and language background also significantly contributed to categorical perception of tones among native and non-native listeners (Chen *et al*., 2017). In comparison to pitch production, it was found that a relative shorter duration is required to perceive than to produce pitch contours, with non-tonal listeners needing longer duration to detect a change in the pitch direction. Duration asserts a stronger effect on between- and within-category discrimination patterns among tonal listeners. Fewer studies investigated the effects of stimulus duration and vowel quality in trilingual non-native speakers with and without musical training. Our study examines categorical perception of resynthesized pitch stimuli by 13 trilingual Cantonese musicians and 13 Cantonese non-musicians. We manipulated tones on both low and high vowels ([a] and [i]) to create 7-step, level-to-falling and level-to-rising pitch continua on both [a] and [i] vowels with 9 different duration values. Cantonese speakers participated in identification and same-different tasks.

Contributed Papers

1pSCb8. Investigating the representation of tonal alternations in context. Yu-Fu Chien (Chinese Lang. and Lit., Fudan Univ., Rm.701, West Wing Guanghua Bldg., Nm.220, Handa Rd. Yangpu District, Shanghai, Shanghai 200433, China, chien_yc@fudan.edu.cn), Hanbo Yan (Chinese Studies and Exchange, Shanghai Int.. Studies Univ., Shanghai, Shanghai, China), and Joan A. Sereno (Linguist, The Univ. of Kansas, Lawrence, KS)

Phonological alternations pose challenges to models of spoken word recognition in how surface information is mapped onto stored representations in the lexicon. In Mandarin, a full Tone3 (213) is reduced to an abridged tone (21) (half-third sandhi) when followed by Tone1, Tone2, or Tone4. In addition, Mandarin Tone3 is replaced by Tone2 when followed by another Tone3 (third-tone sandhi). Two experiments used auditory-auditory priming lexical decision to investigate the alternating representations of Tone3. In Experiment 1, targets consisted of disyllabic words with a half-third or full-third FIRST syllable. These targets were preceded by either a half-third prime, a full-third prime, or a control Tone1 prime. RT data showed facilitation effects for both half-third and full-third prime conditions, with no first syllable by prime type interaction. In Experiment 2, third-tone sandhi targets consisted of disyllabic words with a half-third or full-third SECOND syllable. These targets were preceded by either a half-third prime, a full-third prime, or a control Tone1 prime. The data also showed both half-third and full-third priming effects, without any second syllable by prime type interaction. The results suggest that Mandarin Tone3 is stored as half-third (21) and full-third (213) forms, with both of these tone3 phonological alternations activated.

1pSCb9. Do low intelligibility voices induce false memories? Stephanie Chung, Sophie Bishop, and Molly E. Babel (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, swychung@alumni.ubc.ca)

Spoken language is highly variable with phonetic variation that is attributable to multiple dimensions, including, but not limited to, differences in dialect, mood, native language, sociolinguistic dimensions, and talker-specific idiosyncrasies. Phonetic variation has documented effects on other aspects of linguistic processing. For example, Sumner and Kataoka found that phonetic variation affected semantic processing in semantic priming and false memory tasks, to which they concluding that socially dispreferred (i.e., voices or accents that are socially stigmatized or less desirable) phonetic variation received less attention in encoding, leading to a lack of priming and higher false memory rates [Sumner & Kataoka, 2013, JASA-EL, 134(6)]. In this study we compare the intelligibility and propensity to induce false memories in a between-subjects design. Intelligibility is assessed as response latency in a speeded shadowing task, and false memories are measured using standardized false memory lists where reporting of critical lure items not in the auditorily presented list are counted as false memories. The speaker set consists of five speakers of English, representing two local accents, two nonnative accents, and one native nonlocal accent. The results of this study contribute to our understanding of how phonetic variation affects word recognition and semantic processing.

1pSCb10. Acoustic-phonetic consequences of clear speech in Cantonese vowels and lexical tones. Angela Cooper (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5G4K2, Canada, angela.cooper@utoronto.ca), Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL), and Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

In adverse communicative contexts, speakers will modify their speech style to enhance intelligibility, which can involve a range of acoustic and articulatory adjustments. Prior work has focused on the influence of speaking style on spectral (e.g., vowel space expansion) and temporal distinctions (e.g., voice-onset time). Thus, this study investigated the extent to which such adjustments extend to the pitch dimension, namely lexical tone. Ten native Cantonese speakers were recorded producing five monosyllables with six Cantonese tones embedded in sentence contexts in both clear and conversational speaking styles. Acoustic analyses revealed that speakers produced two distinct styles, with longer vowel durations and greater vowel dispersion in clear relative to conversational speech. However, with regards to lexical tone production, the results indicated only a marginal re-adjustment of tones within the tone space as a result of clear speech. Specific contour tones, such as the low- and high-rising tones, saw increases in their F0 offset. However, mean F0, F0 range and tonal dispersion did not differ substantially between styles. These findings indicate that speakers maintain tonal stability across speaking styles, suggesting that, relative to segments, lexical tones may not be prioritized as phonological contrasts vital to enhance the overall intelligibility of the utterance.

1pSCb11. Speech acoustics, phonetics, and phonology unified by an optimal dynamic model of sound communication. Pierre Divenyi (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, pdivenyi@ccrma.stanford.edu), Mohamad Mrayati (Ctr. for Strategic Development, Riyadh, Saudi Arabia), and René Carré (none, Grenoble, France)

A book by us (Speech: A dynamic Process, de Gruyter, 2017) presents an optimal communication model for the production of sounds through dynamic deformation of a simple tube using the criterion that a maximum contrast between consecutive sounds must be achieved by expending a minimum effort. Its output corresponds well to fundamental articulatory speech data (places of articulation, F1-F2-F3 trajectories). When driven by a periodic source, the model generates natural-sounding vowels and, through constricting the tube at distinctive regions, also stop consonants. From this model's viewpoint, phonetic systems of vowels and consonants arise as the direct consequence of production dynamics capable of producing essentially any vowel or stop consonant. The good correspondence between characteristics of the model and those of speech leads us to conclude that (1) the model is explanatory, (2) because it is acoustically optimal so must be speech production, and (3) an acoustically anchored phonology will derive from it. Based on acoustics, the model thus identifies a natural link between phonetics, and phonology. Propositions of our approach have also been tested in perceptual studies.

Invited Paper

1pSCb12. The effect of minimal neighbors on acoustic detail in the production or real and nonce words in Italian. Olga Dmitrieva (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu) and Chiara Celata (Scuola Normale Superiore di Pisa, Pisa, Italy)

Previous research on English demonstrated that the existence of minimal neighbors in the lexicon can lead to the enhancement of the differentiating acoustic property, such as Voice Onset Time (VOT) in voiceless stops (Baese-Berk & Goldrick, 2009). In this study, the effect of minimal lexical neighbors (e.g., *cara* "dear" is a minimal neighbor for *gara* "match") on the acoustic realization of voicing and gemination was examined in real and nonce words in Italian. Thirty native speakers of Tuscan Italian read one of the two stimuli lists,

one containing only geminated and voiced stimuli and the other containing only singleton and voiceless stimuli, such that a given participant was never exposed to both members of a minimal pair. The results showed that real words with minimal neighbors had longer negative VOT if voiced or positive VOT if voiceless than real words without minimal neighbors. However, the existence of a real minimal neighbor did not affect VOT duration in nonce words. Analysis of nasal and voiceless stop geminates is ongoing.

Contributed Papers

1pSCb13. Computer-vision analysis shows different facial movements for the production of different Mandarin tones. Saurabh Garg (Pacific Parkinson's Res. Ctr., Univ. of Br. Columbia, 442 East, 55th Ave., Vancouver, BC V5X1N4, Canada, srbh.garg@gmail.com), Lisa Tang, Ghassan Hamarneh (School of Comput. Sci., Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman (Dept. of Linguist, Univ. of Kansas, Lawrence, KS), Joan A. Sereno (Dept. of Linguist, Univ. of Kansas, Kansas City, KS), and Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

We aim to identify visual cues resulting from facial movements made during Mandarin tone production and examine how they are associated with each of the four tones. We use signal processing and computer vision techniques to analyze audio-video recordings of 21 native Mandarin speakers uttering the vowel /ɜ/ with each tone. Four facial interest points were automatically detected and tracked in the video frames: medial point of left-eyebrow, nose tip (proxy for head movement), and midpoints of the upper and lower lips. Spatiotemporal features were extracted from the positional profiles of each tracked point. These features included distance, velocity, and acceleration of local facial movements with respect to the resting face of each speaker. Analysis of variance and feature importance analysis based on random decision forest were performed to examine the significance of each feature for representing each tone and how well these features can individually and collectively characterize each tone. Preliminary results suggest alignments between articulatory movements and pitch trajectories, with downward or upward head and eyebrow movements following the dipping and rising tone trajectories, faster lip-closing toward the end of falling tone production, and minimal movements for the level tone.

1pSCb14. Perception of voiceless nasals in Mizo and Angami. Pamir Gogoi and Ratree Wayland (Dept. of Linguist, Univ. of Florida, Gainesville, FL 32611, pgogoi@ufl.edu)

This study is a perceptual analysis of place of articulation of voiceless nasals and their voiced counterparts in Mizo and Angami, two Tibeto-Burman languages spoken in North-East India. The voiceless nasals in Mizo and Angami differ in terms of their characteristics of voicing. In Mizo, there is a portion of voiced nasal murmur before the vowel starts, which has been compared to the voiceless nasals in Burmese (Bhaskararao & Ladefoged,1991). However, in Angami, the nasal murmur remains voiceless throughout (Bhaskararao & Ladefoged,1991; Blankenship *et al*.,1993). A previously performed acoustic analysis of cues for place of articulation in both the languages shows that the cues were more robust in the transition portion of the vowel following the voiceless nasal (Gogoi, 2018). The perception of voiceless nasals by native speakers have not yet been studied. However, existing literature on Burmese voiceless nasals have shown that there are sufficient acoustic cues in the voiced murmur portion of the voiceless nasals for discrimination of the place of articulation (Dantsuji,1986). This study investigates whether the perception results correspond to the acoustic analysis previously observed.

1pSCb15. The role of gender, ethnicity, and rurality in Mississippi back-vowel fronting. Wendy Herd, Joy Cariño, Meredith Hilliard, Emily Coggins, and Jessica Sherman (MS State Univ., 2004 Lee Hall, Drawer E, MS State, MS 39762, wherd@english.msstate.edu)

While tense back-vowel fronting is primarily associated with young women throughout most of the United States, it has been documented in the speech of both men and women from a wide range of ages in the South (Fridland, 2001; Clopper, Pisoni, and de Jong, 2005). However, most of the work documenting back-vowel fronting has been limited to White speakers and has not explored social variation related to ethnicity with the exception of Fridland and Bartlett (2006) who found that both White and Black speakers in Memphis exhibited back-vowel fronting. The present study uses isolated productions of *beat, boot, book, boat,* and *bought* by 73 participants from Mississippi to explore the role of gender, ethnicity, and rurality in back-vowel fronting. Gender and ethnicity—but not rurality—proved significant, and a significant gender by ethnicity interaction was found. These results indicate that White speakers from Mississippi front tense back vowels more than Black speakers and that White women are currently leading back-vowel fronting in this region. These results are consistent with previous findings that men participate in back-vowel fronting in the South, but they contradict Fridland and colleagues' findings that men lead /u/-fronting and that Black speakers are participating back-vowel fronting.

1pSCb16. Bilingual word familiarity in Cantonese and English. Lauretta Cheng (Linguist, Univ. of Michigan, Ann Arbor, MI), Khia A. Johnson, and Molly E. Babel (Linguist, Univ. of Br. Columbia, 2613 West Mall, Totem Field Studios, Vancouver, BC V6T 1Z4, Canada, khia.johnson@alumni.ubc.ca)

Bilingual speakers are typically unbalanced in their vocabularies in each language, with each language's lexicon being representative of the experiences and domains in which the bilingual uses that language. This can create a challenge in creating word lists for speech experiments, as the frequency counts from publicly available corpora do not represent the appropriate vocabulary domains. In this paper, we report on a word familiarity rating task in Cantonese and English that was designed to pretest stimuli in both languages to confirm the items were matched in familiarity for use in subsequent speech tasks. Participants fell into four different groups: Cantonese-English bilinguals who grew up in a Cantonese-dominant location, Cantonese-English bilinguals who grew up in an English-dominant location, North American English speakers with no Chinese language experience, and international English speakers with no Chinese language experience. All listeners were presented with blocks of Cantonese words/nonwords and English words/nonwords and were asked to rate the familiarity of each item. We compare these ratings across participant groups and in relation to corpora for each language to assess how the listener groups perform. We discuss these results in the context of designing stimuli lists that are appropriate for diverse, multilingual populations.

1pSCb17. Characterizing the coordination of speech production and breathing. Jeffrey Kallay (Univ. of Oregon, Eugene, OH), Melissa Redford, and Ulrich Mayr (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu)

A robust, positive correlation between inhalation depth and subsequent utterance length in read speech has been interpreted as an anticipatory effect. Since anticipatory posturing of the respiratory system during phrase preparation has important theoretical implications, the present study was designed to rigorously test the effect. Healthy college-aged participants learned by rote a passage with equal numbers of short and long sentences, which were randomized to create a zero (lag-1) autocorrelation sequence that controlled for priming effects. Rote learning was used to control for visual cues to utterance length. Multiple repetitions of the passage produced by six speakers were acoustically segmented into pause-speech intervals for preliminary analyses, with pause intervals coded for breath intake or no intake. Analyses on data that retained the experimental structure indicated a strong effect of preceding utterance length on the presence/absence of breath intake, but no effect of subsequent utterance length. Analyses of

breath pause durations indicated only significant negative correlations with utterance length, and no relation between intake duration and utterance length. Together, the results suggest only an effect of physiological recovery on speech breathing under conditions that better approximate spontaneous production than in previous studies. [Work supported in part by NIH award number R01HD087452.]

1pSCb18. The effects of second language proficiency on speech production in noise. Saya Kawase (Faculty of Sci. and Eng., Waseda Univ., 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, skawase@aoni.waseda.jp)

Speech production is a dynamic process and talkers produce variance of speech in order to maximize intelligibility and minimize articulatory efforts (cf. hyper- and hypo-articulation model). Prior research has shown that native (L1) talkers modify their speech produced in noise, such as increasing intensity, and changing formant frequency (especially F1 for vowel production). However, it is still unknown to what extent L2 talkers accommodate the environmental need. For example, speaking in noise is less familiar for L2 talkers, especially English as foreign language (EFL) learners. In addition, EFL learners lack interaction in their L2 usage, i.e., their speech may not be successfully modified considering listeners' facilitation. The current study thus tested this with Japanese EFL learners (high and low L2 proficiency). The target stimuli were English tense/lax vowel production (/i-ɪ/ and /u-ʊ/), which is known that Japanese learners of English tend to assimilate the spectral differences and instead use the durational contrast. Our preliminary results showed that both groups of the learners produced spectral enhancement of F1. However, only the high proficiency group used the temporal modification on /i/, potentially due to their faster speaking rate. These results suggest that different modification strategies were observed based on their L2 proficiency.

1pSCb19. Implicit biases in monolingual and bilingual speech perception. Ethan Kutlu and Ratree Wayland (Dept. of Linguist, Univ. of Florida, P.O. Box 115454, Gainesville, FL 32611-5454, denkutlu@ufl.edu)

Accented speech shows a great deal of variation from the "native" norms that can lead to processing difficulty. Previous research on listeners' performance on accented speech shows that the more varieties of accented speech the listeners are exposed to, the better their processing is (Baese-Berk, Bradlow, & Wright, 2013). Since many bilingualism studies suggest that bilinguals have advantages in allocating different resources such as attention during language processing (Costa, Hernández, & Sebastián-Gallés, 2008), there is a necessity to investigate bilingual listeners' processing of accented speech. In this pilot experiment, we will measure intelligibility and perceived degrees of accentedness of Tamil and British English presented in noise among American English monolinguals, and American English and Spanish Bilinguals. We will also perform different cognitive tests as well as an Implicit Association Test to see if prior implicit biases towards South Asian individuals affect the listeners' intelligibility and accentedness rating scores. We hypothesize that bilinguals will be more accurate in the intelligibility task compared to monolinguals due to their exposure to more than one language. However, it remains to be seen whether this advantage will be equally realized in both varieties of English accented speech.

1pSCb20. Adaptation of native clusters with non-native phonetic patterns. Harim Kwon (George Mason Univ., Fairfax, VA) and Ioana Chitoran (Université Paris Diderot, 175 rue du Chevaleret, Paris 75013, France, ioana.chitoran@dartmouth.edu)

Non-native consonant clusters are modified to conform with native phonotactics. This study investigates how onset clusters that are phonotactically licit, but have non-native phonetic patterns, are adapted to match the native patterns. We tested Georgian speakers in a sentence completion task using CCV/CVCV sequences produced by a French talker. Georgian differs from French in having (1) longer inter-consonant timing lag for CCVs, and (2) default initial prominence for CVCVs. The long inter-consonantal lag often results in a transitional vowel in Georgian. Georgian participants (n = 11) first saw the target CCV/CVCV sequences embedded in a Georgian carrier phrase and read the phrase aloud (baseline). Then they heard the target sequences produced by a French talker while seeing the Georgian carrier phrase with an empty slot, and produced the phrase completed with the heard targets (test). Participants' test productions reflected modifications of French targets towards their native phonetic patterns, not only by producing occasional transitional vowels that are absent from French CCV targets, but by deleting the unstressed first vowel in French CVCVs (/pøta/ produced as /pta/). We claim that the effects of native language on adaptation of non-native sequences are not limited to their segmental composition, but also involve their phonetic implementation.

1pSCb21. The relation between production and perception of Mandarin tone. Keith K. W. Leung and Yue Wang (Linguist, Simon Fraser Univ., Robert C. Brown Hall Bldg., Rm. 9201, 8888 University Dr., Burnaby, BC V5A 1S6, Canada, kwl23@sfu.ca)

This pilot study explores the potential production-perception relation of Mandarin tones by examining the association between the acoustic features of native Mandarin tone production and perception of such features, using Tone 2 (rising tone) as a testing case. The perception stimuli were resynthesized by varying F0 onset (spectral feature) and F0 turning-point location (temporal feature) orthogonally forming a grid of tone contours. The endpoints of these dimensions were determined by natural Tone 1 and 3 productions, presumably corresponding to perceptual boundaries for Tone 2. A parabola was fitted to the resynthesized tones to determine the overall shape of the tone contour. Native Mandarin participants were asked to identify the best exemplar of Tone 2. The acoustic analysis of Tone 2 productions by the same participants included measurements of the same features. Preliminary correlation analysis showed a positive production-perception correlation for the temporal feature of F0 turning-point location, but no significant correlation for the spectral features of F0 onset and F0 contour shape, indicating F0 turning-point location as a critical perceptual cue for Tone 2. Further research will include additional tones and additional acoustic features found to characterize these tones to identify the nature of production-perception relation of Mandarin tones.

1pSCb22. Cue enhancement and effects in [l] and [n] identification in L2 English. Bin Li (Linguist and Translation, The City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon Tong 000, Hong Kong, binli2@cityu.edu.hk)

[l] and [n] in English sharing certain articulatory characteristics are acoustically similar (Ladefoged 2003, Lass 1996), which has been reported to cause confusion in consonant recognition (Mermelstein 1977, Espy-Wilson 1992, Pruthi 2004), and adds significantly to the perceptual challenge that Chinese L2 learners of English are faced with (Li 2006). Young Cantonese speakers, for example, often get confused with the English contrast because /l/ and /n/ are considered in free variation and in the process of merging into one in contemporary Cantonese. Despite the articulatory and acoustic similarities, [l] and [n] differ in many ways such as consonantal duration and changes at the constriction release (Li, Zhang, Wayland 2012). This study modified temporal and spectral correlates of the two consonants at various positions in a word, so as to test if cue enhancement could affect L2 consonant perception. University students in Hong Kong were recruited to identify sounds of the contrasting pair in English. Their results suggest that lengthening the consonant and consonant-vowel transition may help the learners better identify the sounds.

1pSCb23. Categorical perception of Mandarin tones by Cantonese trilingual children. Shiyue Li, Si Chen, and Angel Chan (CBS, The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 999077, Hong Kong, shiyue.li@connect.polyu.hk)

It is generally argued that children's tone perception gradually becomes adult-like after age 6. In addition, previous studies suggest that stimulus duration, vowel quality, and musical training significantly contribute to categorical perception of pitch directions in both tonal and non-tonal language (Chen *et al.*, 2017; Zhao & Kuhl, 2015). However, little is known about how musical training may shape categorical perception of trilingual

children, and how stimulus duration and vowel quality come into play. In this paper, we investigate these effects on the categorical perception of Mandarin tones based on Cantonese trilingual children. Specifically, we created rising and falling Mandarin continua on high and low vowels with three duration values. Based on some pilot data, the vowel quality plays a vital role in the categorical perception of rising tones but not in the falling tone pitch directions. Children with professional musical training show much more categorical perception than children without musical training. Specifically, musically trained children show more between-category discrimination than within-category discrimination. Moreover, they are more sensitive to stimulus and are faster in terms of reaction time. Our findings will be compared with previous studies on categorical perception by adults and all significant effects will be discussed.

1pSCb24. Mandarin learners' ESL fricatives. Hua Lin and Junyu Wu (Dept. of Linguist., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, hualin@uvic.ca)

Although English interdentals have been much studied in L2 acquisition, other English fricatives have not received comparable attention. The assumption is that the other fricatives, e.g., [s] which often have L1 counterparts do not pose a problem for ESL learners. However, it is now well understood that similarity between L1 and L2 counterparts may be more problematic in perpetuating an L2 accent. To gain insights into the issue, Mandarin speakers' production of both ESL and native fricatives and their perception of the ESL fricatives is collected and analyzed. The results show that the lack of a problem is true for perception which shows ceiling performance. Production-wise, all ESL voiceless fricatives, not just [θ], are problematic for the Mandarin speakers, judged by a native speaker and measured acoustically on spectral moments and peak. Typically, these fricatives are not on target, but are produced more or less in the L1 fashion. A surprise finding is also made that an L1 counter part may not be the choice of a match; e.g., English [s] which has a close Mandarin counterpart is found replaced with [θ] or [ʃ] by some participants. The paper also discusses the production-perception relation over the findings on the fricatives.

1pSCb25. Perception and production of syllable-tone homophones by second language learners of Chinese. Jiang Liu (Lang., Literatures and Cultures, Univ. of South Carolina, 1620 College St., 917 Humanities Office Bldg., Columbia, SC 29208, jiangliu@mailbox.sc.edu) and Seth Wiener (Modern Languages, Carnegie Mellon Univ., Pittsburgh, PA)

This study examined how the relatively high degree of tonal homophony in spoken Mandarin affects learners' perception and production of new syllable-tone words. Seventeen English-speaking-learners of L2 Chinese participated in a 3-day word-learning task in which pictures were paired with spoken syllable-tone words. Stimuli included 32 monosyllabic Chinese words (e.g., yu3 "feather"). 16 words corresponded to previously learned syllable-tone homophones (e.g., yu3 "language"); 16 words were previously unlearned combinations (e.g., yu4 "jade"). Learning was assessed on day 3 using a 4-alternative-forced-choice identification task and a picture-naming task. Twelve Chinese-L1 speakers identified the learners' productions. Learners were overall more accurate at identifying words that had no previously learned homophone (94%) than words that lacked homophones (90%). The opposite trend was observed in production: learners were overall more accurate at producing words that corresponded to previously learned homophones (77%) than words that lacked homophones (72%). Mixed-effects logistic regression models revealed no statistical difference of the log odds of correct identification or production. These results indicate that despite encountering considerable tonal homophony in speech, learners did not gain advantage in perceiving or producing new words with homophones. The phonologically marked T2-T3 contrast was produced with the lowest accuracy relative to other tonal contrasts.

1pSCb26. The perception-production link in sibilant convergence. Yanyu Long (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, yl2535@cornell.edu)

Many theories suggest a link between perception and production, but evidence for such a link is inconsistent. This study addresses this issue by looking at the relationship between perception of sibilants before [u] or [a] and production change when imitating words starting with a lower-frequency [s] before [u] or [a]. Sibilants were measured by center of gravity. The results at first showed no relationship between production change and perception. However, a closer look revealed that some participants converged to the stimuli while others did not. Mixed effect models showed that for people who converged, larger difference between the perception distribution and the stimuli corresponds to more convergence. This is consistent with the observation that people converged more in the [sa] words, since [s] before [a] has higher perception distribution, hence more distant from the stimuli. No difference was observed between the [u] and [a] contexts in terms of the coefficients of perception in predicting convergence. The results show that no matter what the phonetic context is, inputs are recoded in terms of their distance from the perception distribution before influencing production. With larger such distance, the chosen production targets are farer away from the baseline, resulting in more convergence.

1pSCb27. Sub-articulatory interactions in palatalization processes. Malgorzata E. Cavar (Linguist, Indiana Univ., Bloomington, IN) and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Palatalization is a widespread phenomenon across the world's languages and is typically conceived articulatorily as a tongue body raising and/or fronting gesture that is spread from high and/or front vowels to neighboring consonants. Oddly, palatalization of labial consonants (which allow the greatest degree of gestural overlap with neighboring high and/or front vowels) is typologically least common, suggesting that palatalization processes do not primarily result from gestural overlap, but instead arise from the resolution of incompatible lingual "sub-articulations." In particular, articulatory data from Polish and Russian palatalized consonants point to a critical role for tongue root interactions with tongue blade and dorsum gestures. The products of these resolved sub-articulatory interactions can become lexicalized and reconfigured by later generations of language-learners in response to the interaction of phonological pressure to maintain contrasts and new palatalization processes. The etymological history of the Proto-Slavic stems **menk-* and **ment-* illustrates these interactions. The velar consonant in **menk-* was palatalized by a following high, front vowel to a "soft" posterior affricate. Subsequently, the coronal consonant in **ment-* was palatalized as well (Modern Polish *mącić*). To maintain the contrast between the sound categories, the palatalized velar was reconfigured as a post-alveolar "hard" consonant (*mączyć*).

1pSCb28. Gender, the individual, and intelligibility. Daniel McCloy (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98115-7988, drmccloy@uw.edu), Laura Panfili (Dept. of Linguist., Univ. of Washington, Seattle, WA), Cornelia John, Matthew Winn (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and Richard Wright (Dept. of Linguist., Univ. of Washington, Seattle, WA)

In the clinic and in the laboratory, opinions differ on the relative intelligibility of the speech of women and men. However, the effect of gender alone has rarely been studied explicitly. Here we present a study of 30 talkers (15 male) and 32 listeners assessing intelligibility of a 180-sentence subset of the IEEE sentences presented in steady-state speech-shaped noise. Four signal-to-noise ratios (-4, -2, 0, $+2$ dB SNR) were tested with 45 sentences each. Results showed substantial overlap between intelligibility scores for each gender. Although standard statistical approaches show a slight advantage for female talkers at all SNRs, post-hoc analyses indicated that the gender effect is an artifact driven by a few particularly unintelligible males. These results do not address intrinsic gender-related differences in speech intensity, or in the ability to overcome background noise by speaking clearly, but suggest that gender-related differences are negligible when those factors are controlled. More generally, even with a large sample of talkers, the high degree of talker-intrinsic variability in intelligibility can lead to conclusions that do not generalize to the population of interest, an issue that could affect comparisons rooted in gender, dialect, or other social factors.

1pSCb29. Comparing acoustic and articulatory targets in short-term visual feedback training of non-native vowels. Sonya Mehta (The Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, naya@utdallas.edu) and William F. Katz (The Univ. of Texas at Dallas, TX)

Theories of speech production aim to explain how talkers express abstract linguistic forms as audible events that are intelligible to both speaker and listener. The relationship among planned units of speech, their articulatory implementation, and their acoustic consequences is thus a key issue in speech research. The work reported here is part of a larger project designed to investigate the effects of visual acoustic and visual articulatory feedback on second language (L2) learners' production and perception of non-native speech sounds. L2 talkers from a variety of language backgrounds practiced producing an English vowel, /æ/, while receiving visual feedback on either (1) first and second formant frequencies, provided by a real-time spectrographic display, or (2) tongue back position, shown using a talker-driven tongue avatar. Kinematic data were recorded using an electromagnetic articulograph (EMA) system that tracked tongue midline and lateral movement during vowel productions. Pronunciation accuracy was analyzed by calculating acoustic and kinematic Mahalanobis distances between L2 productions and target (native talker) exemplars. Initial analyses of a single subject's data showed that both types of visual feedback training improved pronunciation, suggesting that both acoustic and articulatory information are recruited during vowel production.

1pSCb30. Gender normalization in fricative perception: Generational differences. Benjamin Munson, Mara Logerquist, and Melanie Putman (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Listeners identify fricatives ambiguous between /s/ and /ʃ/ differently depending on whether they believe that the talker is a woman or a man (Strand & Johnson, 1996; Munson, 2011; Winn, Rhone, Chatterjee, & Idsardi, 2013). Recently, Munson and Logerquist (2017, J. Acoust. Soc. Am. 141, 3982) examined the influence of imputed talker gender on fricative perception across four experimental conditions in a relatively large group (n = 99) of young adults. Surprisingly, Munson and Logerquist were unable to replicate the original finding, despite using stimuli that resembled those from earlier studies very closely. In the current study, we test the hypothesis that Munson and Logerquist's failure to find an effect of imputed gender was because of generational changes in the perception of gender in speech. We do this by comparing the performance of younger, college-aged listeners in Munson and Logerquist to the performance of a new cohort of listeners aged 33–48. Data collection for the latter cohort is ongoing. A finding that these listeners show effects of gender on fricative identification similar to those in previous studies will support the idea that Munson and Logerquist's failure to replicate this finding is due to generational changes in the perception of gender through speech.

1pSCb31. Non-native but not native listeners rely on exemplars for comprehending reduced pronunciation variants. Annika Nijveld, Louis ten Bosch, and Mirjam Ernestus (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, PO Box 9103, Nijmegen 6500 HD, Netherlands, a.nijveld@let.ru.nl)

This study investigates the representations of words' pronunciations in the mental lexicon, and how they are used in spoken word recognition by native (L1) and non-native (L2) listeners. We report two long-term repetition priming lexical decision experiments, each conducted with native English listeners, Dutch learners, and Spanish learners of English. We tested whether the listeners recognized word repetitions more quickly and/or more accurately when the first ('prime') and second ('target') occurrence of the word shared surface details (e.g., the voice of the speaker) compared to when they did not. If so, this suggests that listeners retain word tokens with their acoustic details in their memories (in the form of an exemplar). We found that in one of the experiments, L2 listeners relied more on exemplar representations than L1 listeners did. Specifically, larger exemplar effects arose for the non-native listeners in the experiment in which we included reduced pronunciation variants resulting from acoustic schwa reduction in English (e.g., *b^{a}lloon*). Reduced pronunciation variants are highly difficult to process for non-native but not for native listeners. Our finding suggests that exemplars play a role in spoken word recognition under specific circumstances only, such as under challenging listening conditions for L2 learners.

1pSCb32. Effect of native Korean dialects on Chinese learners' perception of lexical tones. Zhen Qin (Shanghai Jiao Tong Univ., 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045, qinzhenquentin@yahoo.com)

Second-language learners (L2ers) weight cues (e.g., pitch) as a function of how the cues are used in the native language. This study investigates the effect of native dialect on the use of Chinese (i.e., Mandarin Chinese) tonal information by native speakers of Seoul Korean (SK) and Kyungsang Korean (KK). While SK does not use pitch to realize lexical prosody, KK uses pitch to realize lexically contrastive words. Intermediate-to-advanced SK-speaking or KK-speaking L2ers of Chinese (at the same Chinese proficiency) completed a forced-choice tone identification task and a speeded AX tone discrimination task. In the identification experiment, participants heard natural tonal stimuli carried by multiple syllables; in the discrimination experiment, the tonal stimuli were resynthesized to model on natural citation forms of two native speakers (one male, one female), and were superimposed on the vowel /y/. Data collection, with 15 SK listeners and 15 KK listeners to be tested, is ongoing in Shanghai. KK listeners are predicted to have a higher accuracy than SK listeners in identifying tones, and to weigh pitch contour more than SK listeners in MDS analyses of RTs when discriminating tones. If our hypothesis is correct, native dialect is suggested to be considered in L2 speech perception models.

1pSCb33. Productive pitfalls in the phonetic pursuit of psycholinguistic questions. Melissa Redford (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu), Jeffrey Kallay (Univ. of Oregon, Eugene, OH), and Jill Potratz (Univ. of Oregon, Eugene, OR)

Coarticulation in adult speech spans multiple segments, including across syllable and word boundaries. Little in the way of comparable data is available for child speech, where the focus has been on within-syllable anticipatory effects. Still, the developmental studies indicate that anticipatory coarticulation within a syllable is as strong in child speech as in adult speech, and perhaps stronger. This finding is consistent with the hypothesis that words are fundamental units of production under the long-held assumption that anticipatory coarticulation reflects speech motor planning. Our current work leverages this same assumption to investigate age-related changes in the structure of the speech plan. A related goal is to characterize the development of long-distance anticipatory coarticulation. For example, we are testing the hypothesis that younger children's plan consists of more fully individuated word-sized production units than the plan guiding older children's and adults' speech in sentence elicitation experiments, where the initial boundary of a production unit is operationalized as the onset of lip rounding. The work has propelled us to develop a novel psycholinguistic measure of coarticulation, and is now leading us to question the concept of a speech plan and the nature of anticipatory coarticulation. [Work supported by NIH award number R01HD087452.]

1pSCb34. Lexically dependent estimation of acoustic information in speech. Charles Redmon and Allard Jongman (Linguist, Univ. of Kansas, 1541 Lilac Ln., Rm 427, Lawrence, KS 66046, redmon@ku.edu)

Current estimates of the relevant acoustic information in speech are typically inventory-based. They assume a set of phonemes or featural/gestural decompositions of phonemes exists in a language, and proceed to identify and rank/weight the set of acoustic parameters sufficient to distinguish a phonemic contrast based on data from instantiations of the sounds/categories comprising those contrasts in either a controlled set of real words in the language or in nonword syllables. Acknowledging that such contrasts ultimately derive from lexical oppositions, we present an alternative, complementary approach to the quantification of acoustic information in speech: one that is lexically based. Using acoustic data from the Massive Auditory Lexical Decision project (Tucker *et al.*, 2018)—over 26,000 unique words produced by a single speaker of Western Canadian English—and listener identification data on a representative 240-word sample of that lexicon, we define a weighted network of the phonological lexicon (cf. Vitevitch, 2008), where the weights on links between minimal pairs correspond to the predicted acoustic similarity from a model fit to the listener error distributions. From this network the distributed, "global" information contributed by individual parameters operating in an ensemble of lexical oppositions can be estimated from changes in network entropy under perturbations of those parameters.

1pSCb35. Effects of cochlear-implant simulation on processing of vowel sequences by young normal-hearing listeners. Catherine L. Rogers, Jenna Vallario (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu), and Gail Donaldson (Commun. Sci. and Disord., Appalachian State Univ., Tampa, FL)

To better understand the effects of listening environment on efficiency of phonetic processing, the present study examined the effects of signal degradation on phonetic processing of two-syllable sequences by normal-hearing listeners. Auditory temporal-order processing of American-English vowel sequences was compared across two listening conditions, each presented to a separate group of young, normal-hearing listeners: 1) unprocessed resynthesized stimuli and 2) stimuli that had been processed to simulate the signal produced by a 16-channel cochlear implant (CI). Using the methods of Fogerty, Humes and Kewley-Port [2010, *J. Acoust. Soc. Am.*, **127**, 2509-2520], 70-ms resynthesized versions of the syllables "pit, pet, put," and "pot" were presented in a two-syllable temporal-order processing task. Task difficulty was increased by decreasing syllable-onset asynchrony (SOA), i.e., the duration between syllable onsets. SOA thresholds for accuracy of syllable-sequence identification were estimated using the method of constant stimuli on each of four 72-trial blocks. Data analyzed to date show a threshold difference of approximately 20 ms between the unprocessed and CI-processed listener groups, or a difference in threshold of a factor of two or greater. Results will be discussed with regard to implications for phonetic processing of speech in challenging listening environments and practical implications for CI users.

1pSCb36. Effect of distributional shape on learning a target sound. Emily Sadlier-Brown and Carla L. Hudson Kam (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, e.sadlier-brown@alumni.ubc.ca)

Vowel pronunciations (measured in Hz) form distributions that vary from normally distributed to quite skewed. Labov (2001) noted that vowels undergoing change tend to be skewed, while stable ones are not. We ask if the different distributions are causally related to change. We exposed participants (n = 238) to a positively skewed, negatively skewed, or normal distribution of pure tones varying in pitch (to mimic how vowels vary in quality). Participants were told they were listening to notes played by amateur musicians who'd been aiming for the same note. In each trial, participants listened to 20 tones then played the note they thought was the target. Output pitches were compared to the input. The question was whether learners would play a tone corresponding to the mean or the mode of the set or would instead "shift" the note. In the two conditions analyzed so far (normal and positively skewed), participants output pitches that were slightly higher than the mean of the input set, indicating shift in both conditions; however, the difference between conditions is not significant. There is also an effect of the final note in the set, evoking the well-established recency effect in memory. Analysis of the third condition is ongoing.

1pSCb37. Realtime integration of acoustic input and semantic expectations in speech processing: evidence from electroencephalography. McCall E. Sarrett (Psychol. and Brain Sci., Univ. of Iowa, Spence Labs. of Psych., 308 Iowa Ave., Iowa City, IA 52240, mccall-sarrett@uiowa.edu), Efthymia Kapnoula (Basque Ctr. on Cognition, Brain, and Lang., San Sebastian, Spain), and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

A critical question in speech perception is the relative independence of perceptual and semantic processing. Answering this requires addressing two issues. First, does perceptual processing complete before semantic processing (discrete stages vs. continuous cascades)? Second, do semantic expectations affect perceptual processing (feedback)? These questions are difficult to address as there are few measures of early perceptual processing for speech. We extend a recent electroencephalography (EEG) paradigm which has shown sensitivity to pre-categorical encoding of Voice-Onset Time (VOT; Toscano *et al*., 2010). By measuring the timecourse over which perceptual and semantic factors affect the neural signal, we quantify how these processes interact. Participants (N = 31) heard sentences (*Good dogs also sometimes*—) which biased them to expect a target word (*bark* rather than *park*). We manipulated VOT of the target word and coarticulation leading to it. A component-independent analysis determined when each cue affects the continuous EEG signal every 2 msec. This revealed an early window (125–225 msec) sensitive exclusively to bottom-up information, a late window (400–575 msec) sensitive to semantic information, and a critical intermediate window (225–350 msec) during which VOT and coarticulation are processed simultaneously with semantic expectations. This suggests continuous cascades and early interactions between perceptual and semantic processes.

1pSCb38. Developing new pre-lexical processing skills in adults. Anna M. Schmidt (Speech Path. & Aud., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

Studies of adults learning to decode words may provide a window on understanding of the pre-lexical processing of words as learning progresses. Second language (L2) learners (for example) have been found to assimilate new phones to first language (L1) phonemes when asked for the closest sound in their languages (e.g., Schmidt, 1996). Ratings of fit with L1 phonemes was related to details of acoustics of stimuli, L1 acoustics, and vowel context suggesting that L1 phonetic details rather than phoneme categories were compared. What happens after attention has been focused on acoustic/articulatory characteristics of these new sounds (as in training studies) and learning progresses? Do adults establish new phoneme categories to aid in word processing or is it more likely that native categories are expanded based upon developing within L1 category perceptual skills and orthographic word prediction skills. Evidence will be reviewed indicating that learned production of L2 sounds can be become native-like but L2 perception outcomes remain non-native like suggesting that new phoneme categories are not formed. Thus, new learned words can be accurately produced but success in pre-lexical processing of new heard words will depend upon linguistic factors such as context.

1pSCb39. Recalibration dominated by the right ear. Mark Scott (Dept. of English Lit. and Linguist, Qatar Univ., Doha 27419, Qatar, mark.a.j.scott@gmail.com)

Recalibration is a phonetic learning process by which the perceptual boundaries between speech sounds are adjusted. In a typical recalibration experiment, ambiguous sounds are paired with videos that disambiguate the ambiguous sounds to one category or another, the perceiver learns the association and, when the video is absent, continues to perceive the ambiguous sounds as the learned categories. The current experiment tests whether recalibration is stronger in the right or left ear. The right ear has a direct link to the left hemisphere, where such perceptual learning likely occurs, and so we might expect the right ear to be *more* susceptible to recalibration. However, the right ear is less influenced by visual information so we might expect it to be *less* susceptible to such visual-based recalibration. In this experiment, participants heard an ambiguous speech sound (between /b/ and /d/) which was rendered unambiguous by accompanying video of a face pronouncing one of these sounds. In each block, the audio was presented exclusively to the left or right ear. After multiple exposures, participants' perception of the ambiguous sound (without accompanying video) was tested. Data collection is ongoing, but currently there is a strong trend towards more recalibration in the right ear.

1pSCb40. Linking production and perception of clear speech. Joan A. Sereno, Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS), Yue Wang (Linguist, Simon Fraser Univ., 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada, yuew@sfu.ca), Ghassan Hamarneh, Lisa Tang (Comput. Sci., Simon Fraser Univ., Vancouver, BC, Canada), Saurabh Garg (Univ. of Br. Columbia, Vancouver, BC, Canada), Paul Tupper (Mathematics, Simon Fraser Univ., Burnaby, BC, Canada), Bob McMurray (Psych., Univ. of Iowa, Iowa City, IA), Charles Redmon, Yuyu Zeng (Linguist, Univ. of Kansas, Lawrence, KS), Beverly Hannah, Keith K. W. Leung, and Sylvia Cho (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Speech communication can adopt different styles as a function of speaking environments and communicative needs. In auditorily or visually challenging contexts, speakers often alter their speech production using a clarified, hyper-articulated speech style with the intention of enhancing speech intelligibility. Such modifications may result in perceptible articulatory and acoustic changes. Questions thus arise as to whether and what clear-speech modifications facilitate perception. This presentation surveys recent research conducted in our labs, investigating clear-speech production and its associated effects on perception. In a series of three-stream studies, this research relates analyses of visible articulatory features using computer image-processing techniques, measurements of acoustic properties, and perceptual patterns of clear-speech segments and suprasegmentals by native and non-native perceivers. Results reveal that clear (relative to plain) speech modulates different and compensatory articulatory-acoustic cues within each sound category to enhance intelligibility. However, our results also show that clear-speech modifications that reduce phonemic category distinctiveness inhibit intelligibility. These findings indicate that clear-speech effects are governed by the collateral principles of within-category cue enhancement and maintenance of category distinctiveness.

1pSCb41. Weak adaptation to foreign-accented voice-onset-time distribution. Tifani Biro (Penn State Univ., University Park, PA), Seulgi Shin, Yuyu Zeng, and Annie Tremblay (Linguist, Univ. of Kansas, 541 Lilac Ln. Blake Hall, Rm. 427, Lawrence, KS 60045-3129, atrembla.ku@gmail.com)

Listeners quickly adapt to foreign-accented speech in transcriptions and word/sentence judgments (Bradlow & Bent, 2008; Clarke & Garrett, 2004; Sidaras, Alexander, & Nygaard, 2009). However, continuous spoken-word recognition measures such as eye tracking have revealed limitations on foreign-accent adaptation. For example, Trude, Tremblay, and Brown-Schmidt (2013) found that English listeners' adaptation to a second-order constraint in a Quebec French talker's accent (/i/ = [ɪ] before coda consonants except voiced fricatives) was limited. This study investigates whether foreign-accent adaptation is less limited when the mapping between the accented and underlying words does not require higher-level inferencing. English listeners completed an eye-tracking experiment in which they heard a (female) French talker and a (male) English talker. Target and competitor words began with a voiced or voiceless stop and were otherwise temporarily ambiguous (e.g., *timber*and *dimple*). All stops were resynthesized: The French talker's stops were prevoiced (voiced) and short-lag (voiceless) and the English talker's stops were short-lag (voiced) and long-lag (voiceless). Preliminary results suggest an effect of voicing only for the French talker, with voiceless-stop targets generating more competition than voiced-stop targets. Importantly, this effect decreased only slightly from the first to the second half of the experiment, suggesting weak foreign-accent adaptation.

1pSCb42. Perception of word boundaries in spontaneous speech. Jiaer Tao, Francisco Torreira, and Meghan Clayards (McGill Univ., 1085 Ave Dr. Penfield, Montreal, QC H3A 1A7, Canada, meghan.clayards@mcgill.ca)

Studies of lab speech have identified several acoustic variables that listeners use to identify word boundaries (e.g. allophonic variation, segment duration), however spontaneous speech may not contain the same acoustic signals. Using data from a recent study we tested whether disambiguating information is also available to listeners under more typical conditions. 15 pairs of target phrases only differing in the placement of word boundaries, e.g., *grey#day vs. grade#A* were elicited under read (146 tokens) and spontaneous speech conditions (316 tokens) from multiple talkers who were not aware of the ambiguities. Phrases were played in isolation to 30 Native English listeners in a 2AFC segmentation task (e.g. *grey day* vs. *grade A*). Accuracy was above chance for both read (80.9%) and spontaneous speech (73.1%). Accuracy varied according to the critical consonant type and was highest for voiceless stops followed by nasals, voiced stops, fricatives and clusters. Only two items (*pierced ears* vs. *peer steers* and *beef eater* vs. *bee feeder*) were at chance for spontaneous speech. Measures of word-initial lengthening were good predictors of performance but listeners outperformed predictions from these measures indicating they weren't the only cues.

1pSCb43. Use of tonal information in French and English listeners' segmentation of Korean speech. Annie Tremblay, Seulgi Shin (Linguist, Univ. of Kansas, 541 Lilac Ln. Blake Hall, Rm. 427, Lawrence, KS 60045-3129, atrembla.ku@gmail.com), Sahyang Kim (English Education, Hongik Univ., Seoul, South Korea), and Taehong Cho (English Lang. & Lit., Hanyang Univ., Seoul, South Korea)

(Seoul) Korean has been analyzed as having an Accentual Phrase (AP) with a L(HL)H tonal pattern for APs beginning with a lenis segment (Jun, 1998). This tonal information modulates Korean listeners' speech segmentation (Kim & Cho, 2009; Tremblay, Shin, Kim, & Cho, 2018). This study investigates whether French- and English-speaking late learners of Korean can also use this tonal information. French has similarly been analyzed as having an AP with a L(HL)H tonal pattern (Jun & Fougeron, 2002). In contrast, English words are often stressed initially (Cutler & Carter, 1987), with nuclear-pitch-accented words beginning with a H tone (Beckman, 1986). French, English, and Korean listeners completed an eye-tracking experiment. They heard sentences containing a temporary ambiguity between an AP-initial target (*saesinbu-ga]$_{AP}$[masul-eul* 'the-new-bride-subj magic-obj') and a disyllabic competitor spanning the AP boundary (*gama*"palanquin"). Stimuli were resynthesized to orthogonally manipulate the AP-final and AP-initial tones (H/L). Results show that learners eventually used the AP-final and AP-initial tones like Korean listeners, but French listeners made earlier use of both tones than English listeners, even if the AP-initial tone is less crucial to French speech segmentation (Tremblay, Broersma, Coughlin, & Choi, 2016). Speech segmentation is thus very adaptive, even among late learners.

1pSCb44. Identifying the distinctive acoustic cues of Mandarin tones. Paul Tupper (Dept. of Mathematics, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, paul_tupper@math.sfu.ca), Keith K. W. Leung, Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman, and Joan A. Sereno (Univ. of Kansas, Lawrence, KS)

Using mathematical modeling, this study aims to characterize distinctive acoustic features of Mandarin tones based on a corpus of 1013 monosyllabic words produced by 21 native Mandarin speakers. For each tone, 22 acoustic cues were extracted. Besides standard F0, duration, and intensity measures, further cues were determined by fitting two mathematical models to the pitch contours. The first is a broken-line model, which models the contour as a continuous curve consisting of two lines with a single breakpoint. The second model is a parabola, which gives three parameters: a mean F0, an F0 slope, and an F0 curvature. Using Cohen's d, we identify which of the 22 cues are important for distinguishing each tone from the others for all speakers, as well as identifying cues that are used idiosyncratically by particular speakers. Although the specific cues that best characterize each tone differ, we show that the three cues obtained by fitting a parabola to the tone contour are an effective small set of cues such that any pair of tones is well distinguished by at least one of them. We propose using these three cues as a canonical choice for defining tone characteristics.

1pSCb45. The role of distributional information in cross-talker generalization of phonetic category retuning. Christina Tzeng and Lynne C. Nygaard (Emory Univ., 36 Eagle Row, Atlanta, GA 30322, ctzeng@emory.edu)

Listeners use lexical information to retune phoneme categories. Previous findings on the specificity of such perceptual adjustments suggest that cross-talker generalization occurs when talkers are acoustically similar. An alternative mechanism implies that listeners are sensitive to the statistical distribution of phonetic variation across speakers and can use this sensitivity to track both group- and talker-specific phoneme boundaries. Our objective is to disambiguate between these two possibilities. In the experiment, listeners hear words with an ambiguous fricative produced by multiple speakers of American English. Utterances across talkers vary in two ways: 1) the similarity of particular talkers' categorization curves and 2) the statistical distribution of sounds along the phonetic continuum (e.g., unimodal vs. bimodal) across talkers. We assess phonetic category retuning using novel words by novel talkers whose utterances either match or mismatch talker-specific properties or the distribution of phonetic realizations heard during exposure. If cross-talker generalization is similarity-based, listeners should generalize retuned phonetic representations to specific similar-sounding talkers at test. Alternatively, if listeners capitalize on the statistical distribution of exposure sounds, listeners should generalize to test talkers whose productions are consistent with the learned distribution. Results inform the conditions under which listeners form talker-independent representations of speech sound categories.

1pSCb46. Perceptual asymmetries in lexical tone perception. Ratree Wayland (Linguist, Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu) and Si Chen (Dept. of Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

Perceptual asymmetric relations between two stimuli such as perceptual confusions where stimulus A is more frequently confused as stimulus B than the reverse has previously been accounted for in terms of response bias due to the prevalent assumption that perceptual space is Euclidean (Polka & Bohn, 2003). Perceptual asymmetries in lexical perception have been reported for both infants and adults. Tsao (2008) found that a stimulus change from the background mandarin T1 (55) to the target Mandarin T3 (213) was easier than the reverse among one-year-old Mandarin learning infants. Politzer-Ahles *et al*. (2016) found that mismatch negativity (MMN) was attenuated among both native and non-native Mandarin listeners when Mandarin T3 was the standard and another deviant in comparison to the reverse. This study examines the effects of memory load and first language (L1 background) on perceptual asymmetry patterns in adult lexical tone perception. All possible pairings in both orders of Mandarin tones produced on two open syllables [ma] and [ni] will be presented to native Mandarin and American English listeners in an AX categorial discrimination task. To examine the effects of auditory memory load, two different inter-stimulus-interval (ISI), 250 ms and 1,000 ms, will be used.

1pSCb47. Non-native production of Mandarin T2/T3 pairs in disyllabic real and pseudo words. Xianghua Wu (California State Univ., Sacramento, 6000 J St., Sacramento, CA 95819, wu@csus.edu)

Mandarin rising (T2) and falling rising (T3) tones have been found to be the most confusing tone pair for native and non-native production due to similar F0 trajectory and equivalent lexical function known as T3 sandhi. This ongoing preliminary study investigates non-native production of T2 and T3 combinations in disyllabic real and pseudo words, aiming to examine whether T3 sandhi is realized solely at the real word level. Another goal of this study is to explore the effects of two distinctive orthographic systems in Mandarin (i.e., the sound-based pinyin and the meaning-based Chinese characters) on tone production. Ten native English speakers with 8-12 months experience in learning Mandarin participated in the production task. The stimuli were combinations of T2-T2, T2-T3, T3-T3, and T3-T2 grouped into four semantic and orthographic contexts: real and pseudo words in pinyin only, and those in both pinyin and Chinese characters. In the perception task, five native listeners of Mandarin identified the tone category for the first and second syllables. The results are expected to indicate the effects of tonal context, word meaning, and/or orthographic input on T2/T3 productions, and will be compared with earlier findings (e.g., Mok *et al*., 2018; Zhang *et al*., 2015).

1pSCb48. The acquisition of English stress by Kazakh-Russian bilinguals: The role of dominant language. Mahire Yakup (Lang., Linguist and Literatures, Nazarbayev Univ., Block 38-1104, 53 Kabanbay Batyr Ave., Astana, Aqmola 010000, Kazakhstan, yakefu.mayila@nu.edu.kz)

This study systematically investigates the cross-linguistic influence of the acquisition of English lexical stress by Kazakh-Russian bilinguals. Experiment 1 examined the Kazakh accent/stress pattern. Even though there is an argument about Kazakh stress, we used near minimal pairs in Kazakh such as balaDAR "children" vs. balaDAY "like a child, childish." By measuring duration, intensity, and pitch on 20 Kazakh-Russian bilinguals with strong Kazakh language level, we compared the stressed syllable versus unstressed syllables in the near minimal pairs and found duration as a stronger cue for the stress/accent in the syllables. In experiment 2 we investigated the roles of Russian as the dominant language in cross-linguistic influences. We used ten minimal pairs in Russian produced by 20 Russian bilinguals who claimed their native language is Russian. The results showed Kazakh-Russian bilinguals produced Russian stress using duration. Experiment 3 focuses on the acquisition of stress pattern in English by Kazakh-Russian bilinguals. We recruited 40 participants with (IELTS= 7.0) producing 10 English minimal pairs in three conditions. We found that duration and intensity are stronger cues than F0. None of the group used F0 as a stress cue in English lexical stress. The result will be discussed regarding dominant Russian language.

1pSCb49. Priming the representation of left-dominant sandhi words: A Shanghai dialect case study. Hanbo Yan (School of Chinese Studies and Exchange, Shanghai International Studies Univ., 550 West Dalian Rd., Bldg. 2, Rm. 418, Shanghai, Shanghai 200083, China, yanhanbo@shisu.edu.cn), Yu-Fu Chien (Dept. of Chinese Lang. and Lit., Fudan Univ., Shanghai, Shanghai, China), and Jie Zhang (Dept. of Linguist, Univ. of Kansas, Lawrence, KS)

This study examines how the acoustic input (the surface form) and the abstract linguistic representation (the underlying representation) interact during spoken word recognition by investigating left-dominant tone sandhi, a tonal alternation in which the underlying tone of the first syllable spreads to the sandhi domain. We conducted an auditory-auditory priming lexical decision experiment on Shanghai left-dominant sandhi words, in which each disyllabic target ([tɕi55 dɛ31] "egg") was preceded by monosyllabic primes either sharing the same underlying tone ([tɕi55]), surface tone ([tɕi53] "machine"), or being unrelated to the tone of the first syllable of the sandhi targets ([tɕi24] "to remember"). Results showed a surface priming effect, but not an underlying priming effect. Moreover, the surface priming did not interact with speakers' familiarity ratings to the sandhi targets. The results are discussed in the context of how phonological opacity, productivity, and the directionality of tone sandhi patterns influence the representation of tone sandhi words as well as how the lexicality of the primes and the participants' usage pattern of Shanghai may have influenced the results.

1pSCb50. Effect of proficiency on perception of English nasal stops by native speakers of Brazilian Portuguese. Steven Alcorn, Irene Smith, Luis De La Cruz, and Rajka Smiljanic (The Univ. of Texas at Austin, 150 W. 21st St., Stop B3700, Austin, TX 78712, steven.alcorn@utexas.edu)

Brazilian Portuguese (BP) does not contrast /m/ and /n/ in coda position, leading to perceptual difficulties with minimal pairs like *lime*/*line* in L2 English. This study examined the effect of acoustic-phonetic and semantic enhancements on word recognition in noise for native English (n = 26) and L1 BP/L2 English (n = 21) listeners. They heard 120 high- and low-predictability sentences produced in clear and conversational speaking styles and mixed with speech-shaped noise. They typed the last word ending in either /m/ or /n/. The L2 listeners were separated into high- and low-proficiency (HP and LP) groups based on the perceived accentedness ratings of speech samples they provided. The results showed that native listeners benefited more from semantic than acoustic-phonetic enhancements alone, and the accuracy was highest when both enhancements were combined. The HP L2 listeners recognized more target words overall than the LP group, and their benefit from semantic over acoustic-phonetic enhancements mirrored that of the native listeners. In contrast, LP listeners could not utilize the higher-level semantic information to the same degree in the absence of signal clarity. These findings show that L2 listeners benefited from acoustic-phonetic, semantic, and combined enhancements when recognizing /m/-/n/ coda pairs, but this effect was mediated by proficiency.

1pSCb51. Recall of clearly spoken sentences. Sandie Keerstock and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, 305 E. 23rd St. CLA 4.400 E9 Mail Code: B5100, Austin, TX 78712, keerstock@utexas.edu)

Recognition memory (i.e., identifying items as old or new) is higher for sentences spoken clearly than for sentences spoken casually. The clear speech benefit on recognition memory was observed for both native and non-native English listeners. The current study investigated the effect of clear speech on sentence recall, a more complex and effortful type of declarative memory that requires processing at phonological, lexical-semantic, morphosyntactic, and syntactic levels. Thirty native and 30 non-native English listeners heard 72 meaningful sentences (e.g., "The grandfather drank the dark coffee") produced in conversational and clear speaking styles by a female native English speaker. Participants were presented with 12 sentences blocked by speaking style, and instructed to memorize them. After each block of 12 sentences, participants wrote down what they remembered. Blocks were counterbalanced for speaking style. Responses were scored by number of keywords correctly recalled (verbatim memory) and sentence-level comprehension (gist memory). Preliminary results show enhanced verbatim memory for clear speech (i.e., higher rate of key words correctly recalled for sentences produced in clear speech) for both listener groups suggesting that hyper-articulated clear speech provides cognitive release and aids retention of surface form.

MONDAY AFTERNOON, 5 NOVEMBER 2018 SHAUGHNESSY (FE), 1:00 P.M. TO 5:25 P.M.

Session 1pSP

Signal Processing in Acoustics, Acoustical Oceanography, Architectural Acoustics, Musical Acoustics, Underwater Acoustics, and Noise: Machine Learning for Acoustic Applications II

Peter Gerstoft, Cochair
SIO Marine Phys Lab MC0238, Univ. of California San Diego, 9500 Gillman Drive, La Jolla, CA 92093-0238

Weichang Li, Cochair
Aramco Research Center - Houston, 16300 Park Row, Houston, TX 77084

Chair's Introduction—1:00

Invited Papers

1:05

1pSP1. Correlation of biological and acoustical measurements of the endangered Golden Cheeked Warbler with applications of machine learning. David P. Knobles (KSA LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept., The Univ. of Texas at Austin, Austin, TX), Lisa O'Donnell, Darrell Hutchinson, and William Reiner (Balcones Canyonlands Preserve, City of Austin, Austin, TX)

Acoustic measurements on *Setophaga chrysoparia* (Golden-cheeked warbler) have been made within the 40,000 acre Balcones Canyonland Preserve (BCP) in Austin, TX, where the long-term goal is to understand the potential effects of anthropogenic noise on breeding success. The anthropogenic sources of noise include road traffic, jet aircraft, helicopters, and urban development and utilization. During the breeding season from March through May, acoustical recordings were made in dense Ash Juniper forests for 12 hours per day at 12 locations within the BCP. The evolution of the number of songs of Type A and B and their variants are signatures of breeding success. The study attempts to apply principle component analysis for feature selection input into a feedforward neural network classifier of the A and B signatures, their variants, and the ability to automate the identification of individual birds. A feedforward neural network attempts to parameterize a mapping $y = f(x,\theta)$ for input features and category pairs of the A and B signatures and parameterization θ. Explored is the sensitivity of the number of hidden layers required to learn the optimal parameterization θ approximating f. [Work supported by City of Austin.]

1:25

1pSP2. Travel time tomography with local sparse modeling and dictionary learning. Michael J. Bianco and Peter Gerstoft (Marine Physical Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92037, mbianco@ucsd.edu)

In this talk, we present a machine learning-based approach to 2D travel time tomography. Travel time tomography methods image slowness structures (e.g., Earth geology) from acoustic and seismic wave travel times across sensor arrays. Typically, slowness is obtained via an ill-posed linear inverse problem, which requires regularization to obtain physically plausible solutions. We propose to regularize this inversion by modeling rectangular groups of slowness pixels from the image, called patches, as sparse linear combinations of atoms from a dictionary. In this locally-sparse travel time tomography (LST) method, the dictionary, which represents elemental slowness features, is initially unknown and is learned from the travel time data using an unsupervised machine learning task called dictionary learning. This local model constrains small-scale slowness features, and is combined with a global model, which constrains larger-scale features with L2 regularization. In contrast to conventional regularization, which allows only for smoothness or discontinuous slowness, LST permits increased resolution where warranted by data. A *maximum a posteriori formulation* of LST is derived, which is solved as an iterative algorithm. LST performance is evaluated on both synthetic and real data in the context of ambient noise tomography.

1:45

1pSP3. The use of context in machine learning for bioacoustics. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, marie.roch@sdsu.edu), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA), Danielle Cholewiak (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), Erica Fleishman (Colorado State Univ., Fort Collins, CO), Kaitlin E. Frasier (San Diego State Univ., La Jolla, CA), Herve Glotin (Université de Toulon, La Garde, France), Tyler A. Helble (SPAWAR Systems Ctr. Pacific, San Diego, CA), John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA), Holger Klinck (Lab of Ornithology, Cornell Univ., Ithaca, NY), Scott Lindeneau, Xiaobai Liu (San Diego State Univ., San Diego, CA), Eva-Marie Nosal (Univ. of Hawai'i, Honolulu, HI), Kaitlin Palmer (San Diego State Univ., San Diego, CA), Yu Shiu (Lab of Ornithology, Cornell Univ., Ithaca, NY), and Gurisht Singh (San Diego State Univ., San Diego, CA)

Biological acoustic signals are often produced in context. Contextual information includes things such as timing between calls, conspecific or interspecies cues, and physical environmental cues such as sunrise or sunset. We show how some forms of contextual information can be used to improve the results of detection and classification tasks for biological acoustic signals. We examine how context can be used to improve labeled data, resulting in more-accurate classification results, as well as how learners can exploit context directly. We demonstrate these improvements via two bioacoustic detection/classification tasks. The first algorithm detects odontocete echolocation clicks. We used a decision support system that allowed analysts to label echolocation clicks using between call timing cues as well as other measurements and found that deep learners trained with these high quality data are able to detect clicks in adverse environments. The second algorithm applies contextual information surrounding North Atlantic right whale upcalls to improve precision and recall. [This work was supported by ONR Grant Nos. N00014-17-1-2867 and N00014-15-1-2299.]

Contributed Papers

2:05

1pSP4. Bayesian association of multiple infrasound events using long-range propagation models. Christophe Millet (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr), Michael Bertin (CMLA, ENS, Cachan, France), and Pierrick Mialle (CTBTO, Vienna, Austria)

The International Monitoring System (IMS) is a worldwide network of monitoring stations that helps to verify compliance with the Comprehensive Nuclear Test-Ban Treaty by detecting events that might indicate violations of the treaty. The IMS uses a combination of four technologies: seismological, radionuclide, hydroacoustic, and infrasound. An important limitation of these technologies is related to the fact that the structure of the propagation medium is partially unknown. This is especially true for the infrasound technology, and indeed, a current trend is to undertake the impact of atmospheric variability on waveforms using computational models. Simulation-based predictions, however, are inherently limited by large uncertainties. Further, many thousands of detections are recorded per week and thus, the problem of calculating plausible waveforms for subsets of detections often leads to computational demands that exceed available resources. In this paper, we present a new powerful statistical model for analyzing and interpreting large-scale IMS data. The method is based on a parallel Markov chain Monte Carlo algorithm and full-wave modeling. The method can detect association when multiple, interacting events are present in the data. The posterior probability of no association can be estimated, thereby providing a way to reduce the false alarm rate in operational-like environments.

2:20

1pSP5. A sensitivity-based Bayesian hierarchical process for calibrating infrasound propagation models. Christophe Millet (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr)

Model calibration is viewed in the sense of adapting the full set of model parameters in order to get better resemblance between observations and major end-predictions. In this paper, we present a new probabilistic method to calibrate normal-mode-based propagation models using some observed data and sources of uncertainties. The unknown parameters are estimated using a multiple parallel Markov Chain Monte-Carlo method, for which convergence diagnostics are available. Using a few normal modes allows to rapidly estimate the statistical distributions of the arrival characteristics, on a mode-by-mode basis. In a sense, the unknown inputs "propagate" through the plausible waveguides with each mode and alters its amplitude and phase structure. The resulting waveform is obtained as a combination of individual wavepackets which depends continuously of the input parameters. Further, once the maximum likelihood has been identified, the reduced model can be extended to higher dimensions (with a larger number of modes) to better refine the calibration process. Numerical results are obtained using the FLOWS platform (*Fast Low Order Wave Simulation*), that integrate advanced spectral numerical methods and realistic representations of atmospheric disturbances. The method is used to revisit the infrasound signals recorded at I37NO during campaigns of ammunition destruction explosions at Hukkakero.**2:35**

1pSP6. Detecting infrasound signals and sources with dense network data. Michael A. Hedlin and Catherine de Groot-Hedlin (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0225, hedlin@ucsd.edu)

New, massive, datasets can be used to examine atmospheric phenomena in more detail than before but require analytical methods that are both efficient and capable of extracting useful information from faint signals immersed in noise. We have developed the AELUMA (Automated Event Location Using a Mesh of Arrays) method that recasts any dense network of sensors as a distributed mesh of triangular arrays. Each array provides a local estimate of signal properties. This information from arrays across the network is combined to estimate the source origin time and location. The process is repeated without oversight to catalog events. A key challenge in attributing signals to their source occurs when a large number of signals are detected nearly concurrently from different sources. We apply a cluster (decision tree) analysis that takes the results of array processing at all arrays to iteratively parse out subsets of detections from distinct sources. We used recordings of infrasound signals made at an extensive network of sensors to build a catalog of infrasonic activity across the continental United States. The accuracy of AELUMA is assessed using events for which the origin time and location are well known.

2:50

1pSP7. Graph clustering for localization within a sensor array. Peter Gerstoft (Noise Lab, Univ. of California San Diego, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu)

We develop a model-free technique to identify weak sources within dense sensor arrays using graph clustering. No knowledge about the propagation medium is needed except that signal strengths decay to insignificant levels within a scale that is shorter than the aperture. We then reinterpret the spatial coherence matrix of a wave field as a matrix whose support is a connectivity matrix of a graph with sensors as vertices. In a dense network, well-separated sources induce clusters in this graph. The support of the covariance matrix is estimated from limited-time data using a hypothesis test with a robust phase-only coherence test statistic combined with a physical distance criterion. The method is applied to a dense 5200 element geophone array that blanketed 7 km $\times$10 km of the city of Long Beach (CA). The analysis exposes a helicopter traversing the array.

3:05–3:25 Break

Invited Papers

3:25

1pSP8. Blind equalization and automatic modulation classification of underwater acoustic signals. Caitlyn N. Marcoux, Bindu Chandna, and Ballard J. Blair (The MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, cmarcoux@mitre.org)

It is useful to passively characterize the underwater acoustic communications environment for a variety of purposes, including interference avoidance and enforcement of restrictions protecting marine mammals. Automatically determining the modulation of a received waveform can permit sonar or communications operations within the same bandwidth with a minimum of collisions, and it can identify a particular system operating outside its permitted regime. The characterization system needs to both determine the modulation and an unknown, time-varying channel impulse response since the transmitter and receiver are not coordinating. In this work, we demonstrate the use of blind equalization along with Convolutional Neural Networks for automatic classification of underwater signals. Our current research focuses on classification of constant modulus signals and demonstrates an approximate 30 percent improvement in modulation classification, compared to approaches without equalization, and a significant reduction in the amount of data needed for training. We considered BPSK, QPSK, MSK, FSK and 8-PSK modulations using simplified synthetic channels simulated via MATLAB to demonstrate our results. Future work is aimed at demonstrating classification improvement using realistic channel models simulated via the Sonar Simulation Toolkit, real underwater channels gathered from data collects, and additional underwater acoustic signal types.

3:45

1pSP9. Acoustic seafloor classification using machine learning and simulations of Helmholtz equations. Christina Frederick and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Blvd., Newark, NJ 07102, christin@njit.edu)

We discuss numerical methods for inverse problems in high frequency underwater acoustics, aiming to recover detailed characteristics of the seafloor from measured backscatter data generated from SONAR systems. The key to successful inversion is the use of accurate forward modeling capturing the dependence of backscatter on seafloor properties, such as sediment type, roughness, and thickness. Prediction models are often statistical models formed using years of collected ground-truth data or physical models that require costly simulations or unrealistic environmental assumptions. Sophisticated tools are needed to accommodate complicated scenarios, i.e., uncharted seafloor landscapes. To enable a rapid, remote, and accurate seafloor parameter recovery, we propose a combination of machine learning and high fidelity forward modeling and simulation of the physical wave propagation and scattering process. The idea is to partition environments on the order of kilometers in width into much smaller "template" domains, a few meters in width, where the sediment layer can be described using a few parameters. Machine learning is used to train a classifier using a reference library of simulations of Helmholtz equations on the domains. We investigate the potential of multilayer perceptrons and generalized regression in sediment identification using the reference library. [Work supported by NSF and ONR.]

4:05

1pSP10. Big data based ocean acoustic source localization via deep learning. Haiqiang Niu, ZaiXiao Gong, Haibin Wang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, nhq@mail.ioa.ac.cn), Emma Reeves Ozanich, and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Machine learning has shown its potential in ocean acoustic source localization when real data are available to construct the training data set (Niu *et al*., JASA 142, 1176–1188 (2017); Niu *et al*., JASA 142, EL455–460 (2017)). This paper investigates the cases suffering from deficient real observations, where big data generated from the acoustic propagation model are used as the training set and therefore deep learning models could be exploited. To handle the environmental uncertainty, the multi-frequency acoustic data are generated using an acoustic propagation model (KRAKEN) based on tens of thousands of environments. A 50-layer residual neural network model (ResNet-50) is used to address this challenging problem. The performance on simulated and experimental data demonstrates that the proposed approach is able to adapt to a variety of environments in source localization problems.

Contributed Papers

4:25

1pSP11. Clustering analysis of crowd noise from collegiate basketball games. Brooks A. Butler, Mylan R. Cook, Kent L. Gee, Mark K. Transtrum, Sean Warnick, Eric Todd, and Harald Larsen (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com)

The relationship between crowd noise and crowd behavioral dynamics is a relatively unexplored field of research. Signal processing and machine learning (ML) may be useful in classifying and predicting crowd emotional state. As a precursor to performing ML, it is instructive to identify which crowd acoustic events an unsupervised ML algorithm would classify as unique. An initial set of audio features have been extracted from high-fidelity noise recordings of crowds at Brigham Young University men's and women's basketball games. A K-Means clustering analysis was conducted on half-second segments of the recordings using extracted features consisting of numerous statistical and spectral characteristics. For example, a clustering analysis performed on a one-twelfth-octave spectrogram of crowd noise recordings reveals there are approximately six unique events that occur during a game. The clusters for different audio feature sets are compared with human labeling of the different acoustical events. Implications for further ML algorithm development based on various sets of audio features are discussed.

4:40

1pSP12. Clustering analysis of inputs to a geospatial model of outdoor ambient sound. Brooks A. Butler, Kent L. Gee, Mark K. Transtrum, Katrina Pedersen (Phys. and Astronomy, Brigham Young Univ., 35 E 800 N, Provo, UT 84604, brooks.butler93@gmail.com), Michael M. James, and Alexandria R. Salton (Blue Ridge Res. and Consulting LLC, Asheville, NC)

Outdoor ambient acoustical environments may be predicted through supervised machine learning using geospatial features as inputs. However, collecting sufficient training data is an expensive process, particularly when attempting to improve the accuracy of models based on supervised learning methods over large, geospatially diverse regions. Unsupervised machine learning methods, such as K-Means clustering analysis, enable a statistical comparison between the geospatial diversity represented in the current training dataset versus the predictor locations. In this case, the geospatial features that represent the regions of western North Carolina and Utah have been simultaneously clustered to examine the common clusters between the two locations. Initial results show that most geospatial clusters group themselves according to a relatively small number of prominent geospatial features, and that Utah requires appreciably more clusters to represent its geospace. Additionally, the training dataset has a relatively low geospatial diversity because most of the current training data sites reside in a small number of clusters. This analysis informs a choice of new site locations for data acquisition that maximize the statistical similarity of the training and input datasets. [Work funded by an Army SBIR.]

4:55

1pSP13. Optimal experimental design for machine learning using the Fisher information matrix. Tracianne B. Neilsen, Mark K. Transtrum, David F. Van Komen (Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), and David P. Knobles (KSA, LLC, Austin, TX)

Optimal experimental design (OED) refers to a class of methods for selecting new data collection conditions that minimize the statistical uncertainty in the inferred parameter values of a model. The Fisher information matrix (FIM) gives an estimate of the relative uncertainty in and correlation among the model parameters based on the local curvature of the cost function. FIM-based approaches to OED allow for rapid assessment of many different experimental conditions (e.g., input data type, parameterizations, etc.). In machine learning models, accurate parameter estimates are often not a priority (nor even desirable) as they have no direct physical meaning. Instead, one would like to minimize the uncertainty in the model predictions for several quantities of interest. FIM approaches to OED can be generalized to minimize statistical variance, not in parameters, but in predictions of the quantities of interest. This approach has been applied, for example, to systems biology models of biochemical reaction networks [Transtrum and Qiu, BMC Bioinformatics 13(1), 181 (2012)]. Preliminary application of the FIM to optimize experimental design for source localization in an uncertain ocean environment is a first step towards an efficient machine learning algorithm that produces results with the least uncertainty in the quantities of interest.

5:10

1pSP14. Curvature and distance amplitude correction for ultrasonic testing of a component with curved surface using deep learning method. Yujian Mei (Zhejiang Univ., 38 Zheda Rd., Hangzhou, Zhejiang Province 310027, China, meiyj1019@zju.edu.cn), Haoran Jin (Nanyang Technolog. Univ., Hangzhou, ZheJiang, China), Bei Yu, Eryong Wu, and Keji Yang (Zhejiang Univ., Hangzhou City, Zhejiang Province, China)

Surface curvature and distances from the transducer will have an effect on the amplitude of ultrasonic wave. When using the amplitude of ultrasonic signal to quantitatively size the flaws, a correction is necessary to assure all ultrasonic signal amplitude is based on the echo signal from flat surface. Hence, a deep learning method is proposed to adjust the signal amplitude which enters arbitrary curved surface. A deep learning model will adjust the signal amplitude as the radius of surface, distance from the transducer and attenuation coefficient of material. Then, the ultrasonic signal is imaged on the basis of corrected ultrasonic data. Compared to the ultrasonic imaging based on the raw data, the quantitative evaluation of flaw size is more precise by using the deep learning method. The deep learning model is trained by the CIVA simulation data. Ultimately, the model is verified by the comparisons between prediction and experimental data. The proposed method is adaptable to various curvature, distance, and material, which avoids manufacturing multiple standard specimens to satisfy lower costs and shorter cycle.

Session 1pUWa

Underwater Acoustics: Noise and the Environment

Gerald L. D'Spain, Chair
Marine Physical Lab, Scripps Institution of Oceanography, 291 Rosecrans St., San Diego, CA 92106

Contributed Papers

1:00

1pUWa1. Green's function emergence through cross-correlation of shipping noise. Emmanuel Skarsoulis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, N. Plastira 100, Heraklion GR-70013, Greece, eskars@iacm.forth.gr) and Bruce Cornuelle (Marine Physical Lab., Scripps Inst. of Oceanogr. UCSD, La Jolla, CA)

The cross-correlation of the underwater acoustic noise field at two locations is related to the time-domain Green's function (TDGF) between the two locations. The propagation conditions, the distribution of noise sources, and the hydrophone locations have a significant influence on the part of the TDGF that can be reliably reconstructed through the cross-correlation procedure. In the low-frequency part of the acoustic spectrum the dominating noise source is shipping, and for that case the effect of propagation conditions on the result of the cross-correlation procedure is studied by considering typical sound velocity profiles and hydrophone setups. The aim is to understand the effect that propagation conditions have on the cross-correlation procedure and point to the potential advantages of certain areas for the conduct of passive tomography experiments. Further, the effect of mooring motion on the cross-correlation result is addressed and ways to remove this effect, e.g., with the help of long-baseline navigation systems will be discussed. [Work supported by ONR.]

1:15

1pUWa2. Reductions in underwater radiated noise from shipping during the 2017 Haro Strait vessel slowdown trial. Alexander O. MacGillivray and Zizheng Li (JASCO Appl. Sci., 2305–4464 Markham St, Victoria, BC V8Z7X8, Canada, alex@jasco.com)

During 2017, the Vancouver Fraser Port Authority's Enhancing Cetacean Habitat and Observation (ECHO) program carried out a voluntary slowdown trial in Haro Strait (British Columbia) to investigate whether limiting vessel speeds to 11 knots would decrease noise in Southern Resident Killer Whale habitat. During the trial, JASCO collected source levels measurements on two underwater listening stations situated adjacent to the Haro Strait traffic lanes, while a third listening station in Georgia Strait measured noise from vessels outside the slowdown zone. Acoustic data from these three listening stations were analyzed using JASCO's PortListen® system, which tracks vessels using the Automated Identification System (AIS) and automatically measures the source levels of passing vessels, according to the ANSI standard for ship noise measurement (12.64-2009 R2014). The effects of voluntary slowdowns on vessel noise emissions were investigated, on a per-class basis, by comparing measurements of participating vessels with measurements obtained during control periods before and after the trial. Analysis of the trial data showed that speed reductions were an effective method for reducing broadband source levels for five categories of piloted commercial vessels: containerships, cruise vessels, vehicle carriers, tankers, and bulk carriers.

1:30

1pUWa3. An experiment of air-to-water sound propagation in deep water. GuangXu Wang (National Lab. of Acoust., Inst. of Acoust.,Chinese Academy of Sci., No.21st of Bei Si Huan Xi Rd., Beijing 100190, China, wgx@mail.ioa.ac.cn), LingShan Zhang, and ZhaoHui Peng (National Lab. of Acoust., Inst. of Acoust.,Chinese Academy of Sci., Beijing, China)

The underwater sound propagation from an airborne source has been paid much attention recently. To explore the underwater sound propagation from an airborne source in deep water, an experiment was conducted in South China Sea. A vertical line array of hydrophones with depth from 100 meters to 3400 meters was anchored on the seabed to detect the sound of a loudspeaker. Two microphones were placed under the loudspeaker to measure the source level. Both continuous and linear frequency modulation wave signals with frequency from 100 Hz to 2000 Hz were transmitted by the loudspeaker. The distances between the loudspeaker and the vertical line array were from 5km to 14.5km. Comparing signal detected by underwater hydrophones with signal detected by microphones in air, transmission loss of is figured out. The transmission loss predicted by a wave number integration model OAST is compared with the experimental results.

1:45

1pUWa4. Stochastic matched-field processing using polynomial chaos expansions to model environmental uncertainty. Roger M. Oba, Peter C. Mignerey, and Steven I. Finette (Acoust. Div., Code 7167, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, roger.oba@nrl.navy.mil)

A stochastic generalization of matched-field processing (MFP) is presented that incorporates stochastic steering *matrices* rather than steering *vectors*. This provides a natural framework for including environmental uncertainty associated with incompletely known environmental knowledge. Parametric probabilities allow use of polynomial chaos (PC) expansions to describe environmental uncertainty in a rigorous manner, yielding efficient representations of the stochastic steering matrices through the application of sparse sampling methods. In particular, PC methods apply to the uncertainty of sediment variability that can be modeled via horizontal spectral methods. With moderate variability over scales justified by data in the New Jersey shelf region, the sediment uncertainty can be modeled with relatively few PC expansion coefficients, which depend only on the mean amplitude of spectral components. The coefficients express the nonlinear dependence of the acoustic field on the sediment variability, essentially orthonormalizing higher moments. The coefficients lead immediately to stochastic steering matrices, which are then compared via various MFP processors against acoustic data for MFP source localization. [Work supported by the U.S. Office of Naval Research.]

2:00

1pUWa5. Underwater ambient noise spatial and temporal coherence at basin scale. G. Bazile Kinda, Florent Le Courtois, Yann Stéphan, Jean-Michel Boutonnier (Shom, HOM, 13, rue du Chatellier, Brest CS 92803, 29228, France, bazile.kinda@shom.fr), Jean-Yves Royer (Laboratoire Domaines Oceaniques, CNRS and Univ. of Brest, Plouzané, France), and Guilhem Barruol (Institut de Physique du Globe de Paris, Paris, France)

Underwater soundscape may be seen as an intrinsic feature of the environment in which it is recorded. In fact, underwater soundscape is a complex combination of sounds from a variety of acoustic sources, whether from natural or from human origin. When the distances between the sources and the receivers are sufficiently large, the received signals are strongly modified by the acoustic response of a continuously fluctuating marine environment. Then, spatio-temporal characteristics of the environment can be inferred from the measurements of soundscapes from hydrophone networks. The presentation aims to assess the spatial and temporal coherences of low frequency underwater ambient noise in the Indian Ocean basin. The data under consideration combine year round acoustic recordings from two recoding systems: on the one hand, data were from ~50 Ocean Bottom Seismometers (OBS) arranged in a network as large as 2000 km x 2000 km (http://www.rhum-rum.net/); and on the other hand acoustic data from hydrophones in the SOFAR channel.

2:15

1pUWa6. Sparsity-based shipping noise analysis and cancellation in underwater acoustic communication. Ruoyu Zhang (Commun. Res. Ctr., Harbin Inst. of Technol., 2 Yikuang St., Harbin 150080, China, 15B905028@hit.edu.cn), Lutz Lampe (Dept. of Elec. and Comput. Eng., The Univ. of Br. Columbia, Vancouver, BC, Canada), and Honglin Zhao (Commun. Res. Ctr., Harbin Inst. of Technol., Harbin, China)

We are interested in achieving reliable underwater acoustic communication (UWAC) in the presence of noise generated by vessel traffic, i.e., shipping noise. Our general approach is to determine an underlying structure in shipping noise and to exploit this structure to remove it from desired UWAC signals. Since a well-established statistical characterization of shipping noise is not available, in this paper, we discuss our evaluation of shipping noise data sets obtained from labeled recordings from an underwater acoustic hydrophone array deployed in the Strait of Georgia by Ocean Networks Canada. Motivated by the general observation that shipping noise exhibits a structure akin to impulse noise, we study to what extent shipping noise can be described as a sparse signal. Specifically, we apply the expectation-maximization method to train a Gaussian-mixture model with two components, in which the large variance component represents the impulse-noise part. In a second step, we use the model to perform noise cancellation in a multicarrier UWAC scenario. For this purpose, we apply a sparse-signal recovery formulation and a message passing algorithm for detection, i.e., a form of compressed sensing. Numerical results show the moderate level of mitigation of shipping noise interference achieved with the proposed method.

2:30

1pUWa7. Toward efficient shipping noise probability density function estimation using sea-lane source decomposition and probability theory. Florian J. Aulanier (Institut Maurice-Lamontagne, Fisheries and Oceans Canada, Pêches et Océans Canada 850, Rte. de la Mer, C. P. 1000, Mont-Joli, QC G5H 3Z4, Canada, florian.aulanier@dfo-mpo.gc.ca)

One goal of the Canadian Ocean Protection Plan (OPP) is to understand the potential effects of shipping noise on endangered whale species in order to mitigate them. Shipping noise environmental impact risk assessment requires the understanding of large scale, high-resolution time-space shipping noise distributions. The computation of such shipping noise probability density functions (pdf) requests considerable computing resources, especially when propagation occurs in complex and varying environments like shallow waters, canyons, or fjords. Besides, input parameters variability and uncertainties analyses require multiple hindcast, nowcast or forecast scenarios to be run when using a direct Monte-Carlo approach. In order to reduce the computation effort, sea-lane shipping traffic decomposition and probability theory are jointly used to derive shipping noise probability density functions with a logarithmic complexity algorithm, as opposed to the linear complexity of direct Monte-Carlo methods. First, a theoretical model is derived for straight shipping routes using simplified logarithmic propagation, validated with numerical examples, and used to perform a sensitivity analysis of shipping noise pdf to speed and route closest point of approach. The improvement in numerical efficiency is shown on a more realistic four sea-lane case scenario mimicking part of the summertime St. Lawrence estuary traffic. Eventually, *in situ* measurements will be available for comparison.

2:45

1pUWa8. Parallel processing and machine learning for long-timescale ambient noise measurements—Illustration with data from the Neptune Ocean Observatory offshore British Columbia. Amelie Klein (Mathematics, Univ. of Bath, University of Bath, Claverton Down, Bath, Avon and NE Somerset BA2 7AY, United Kingdom, ammk22@bath.ac.uk), Philippe Blondel (Phys., Univ. of Bath, Bath, Avon and NE Somerset, United Kingdom), and Kari Heine (Mathematics, Univ. of Bath, Bath, and NE Somerset, United Kingdom)

Seafloor observatories are a rapidly maturing technological approach, enabling the monitoring of marine soundscapes over large areas (basin scale) and long timescales (10+ years). However, the time needed to process broadband measurements, especially over large periods, often acts as a bottleneck. This is particularly true when combining multi-resolution analyses with assessing the impacts of relatively short transients. We are using parallel processing to enable machine learning approaches. To accelerate the computation of spectrograms, we have implemented a parallel processing method that uses the FFT algorithm FFTW3 (http://www.fftw.org/fftw3.pdf), using MPI/C++ on the High Performance Computing facilities at the University of Bath, and compared with spectrogram calculations from well-established software PAMGuide (Merchant *et al.*, Meth. Ecol. Evol. 2015), with Matlab's Parallel Computing Toolbox. This approach was tested on 1 month of broadband (96 kHz) measurements from the NEPTUNE node at Folger Deep. One month of data can be processed in <3 hours, to a dB accuracy even on short time segments, and that performance increases with the number of parallel processing units. Stability of the parallel approach has been tested with synthetic signals (e.g., chirps) and increasing signal-to-noise ratios. This enables much faster monitoring of long-term trends of important sound metrics.

3:00–3:15 Break

3:15

1pUWa9. Soundscape measurements and modeling of the Atlantic Outer Continental Shelf. James J. Murray (OASIS, Inc., 5 Militia Dr., Lexington, MA 02421, jjmurray@gmail.com), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA), and Richard L. Campbell (OASIS, Inc., Lexington, MA)

The impact of anthropogenic sound on the marine ecosystem has become an area of concern and subsequently increased research. As part of the Atlantic Deep Water Ecosystem Observatory Network (ADEON, adeon.unh.edu), a five-year collection effort of acoustic, oceanographic, marine ecosystem, and remote sensing satellite observations is underway. The centerpiece of the network is 6 long-term landers deployed off the coast of Jacksonville up past the mouth of the Chesapeake Bay. In this paper we present results from the first (of four) horizontal line array measurement cruises (deployed from a sailboat) between two of the moorings. The scientific objective is to evaluate the horizontal directionality of the ambient sound field and to measure the spatial correlation of the noise between the lander off of Wilmington NC and the lander off of the Virginia. A soundscape model, including environmental and source level variability will be presented. The acoustic sources will include ships, wind, rain, and marine mammals.

3:30

1pUWa10. Spatio-temporal modelling of underwater noise related to an arctic exploration drilling program. Jorge E. Quijano, David E. Hannay (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z 7X8, Canada, jorge.quijano@jasco.com), and Melanie Austin (JASCO Appl. Sci., Anchorage, AK)

One concern related to oil and gas industry activities in the arctic is the potential impact of anthropogenic noise on marine mammals. To gain understanding of the temporal evolution of the acoustic noise footprint from such activities, and to quantify the geographic extent at which sound levels remain within regulatory thresholds for marine mammal injury and behavioral disturbance, this work presents a comprehensive monitoring and modeling study of anthropogenic noise resulting from an exploration drilling project in the northeastern Chukchi Sea, conducted from July to October, 2015. Measured acoustic source levels of multiple vessels and drilling equipment were used to generate time-dependent wide-area noise fields. Model validation was carried out by comparing simulated results with high-resolution acoustic data acquired at multiple stations in the area. The study shows that noise radiating from up to 19 supporting vessels is usually dominant, but noise from certain drilling-related activities occasionally rises above vessel noise within a radius of 8 km from drillsites. Modeled scenarios also show that by constraining vessel positions to within a few kilometers of the drilling location, simultaneous construction of two oil well top holes results in minor increment of the aggregate acoustic footprint's extent relative to single-well drilling.

3:45

1pUWa11. Upslope propagation of low frequency deep ocean signals. Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Kenneth Houston, Robert Tingley, Terry Nawara (Draper Lab., Cambridge, MA), Daniel Lawrence (Riptide Autonomous Solutions, Pembroke, MA), and Thomas Brovarone (Hydroacoustics, Inc., Henrietta, NY)

Over the period 18–21 March, 2017, 1-hour waveforms in the 60–120 Hz band were transmitted from a HLF-6A source deployed at 300 m depth in the deep northeast Pacific Ocean. These transmissions were recorded by two vertical hydrophone line arrays at various ranges in the deep ocean, GPS-equipped sonobuoys deployed on the continental shelf, and a bottom-mounted hydrophone in 900-m water at the western edge of Monterey Bay operated by the Monterey Bay Aquarium and Research Institute (MBARI). Although the source-receiver ranges to the sonobuoys and the MBARI hydrophone were approximately the same, the bathymetry profile up the continental slope to the sonobuoy location was significantly steeper than to the MBARI hydrophone. Only the upper part of the frequency band, above 90–95 Hz, is received with good signal-to-noise ratio, illustrating the high-pass temporal filtering of upslope propagation. Upslope propagation also acts as a low-pass spatial filter, allowing only lower-order modes to propagate onto the shelf. Numerical modeling is used to examine the predictability of the measured travel times and multipath arrival structure.

4:00

1pUWa12. Compressive sound speed profile estimation with direction-of-arrivals. Youngmin Choo (Defense System Eng., Sejong Univ., 209 Neungdong-ro, Gwangjin-gu, Seoul 05006, South Korea, ychoo@sejong.ac.kr) and Woojae Seong (Naval Architecture and Ocean Eng., Seoul National Univ., Seoul, South Korea)

Sound speed profile (SSP) predominates acoustic propagation in the ocean. In this work, the SSP is inverted using compressive sensing (CS) combined with direction of arrivals (DOAs) from beamforming. The travel times and the positions of the arrivals can be approximately linearized using their Taylor expansion with the shape function coefficients that parameterize the SSP. The linear relation between the travel times/positions and the shape function coefficients enables CS to reconstruct the SSP. The conventional objective function in CS is modified to simultaneously exploit the information from the travel times and positions. The proposed scheme is examined in the SWellEx-96 experimental environment.

4:15

1pUWa13. Ship speed reduction is an effective mitigation for underwater noise effects. Jason Wood (SMRU Consulting, PO Box 764, Friday Harbor, WA 98250, jw@smruconsulting.com), Dominic Tollit, and Ruth Joy (SMRU Consulting, Vancouver, Br. Columbia, Canada)

The Vancouver Fraser Port Authority's ECHO Program led a voluntary 2-month ship slowdown trial during which 56% of piloted ships slowed down to <13 knots in a 16 nautical mile corridor. The goal was to determine if a slowdown could be used as a mitigation measure to reduce ship related underwater noise effects in core Southern Resident killer whale habitat. A calibrated hydrophone system was used to measure ambient noise levels from 10 Hz to 100 kHz during the trial and a representative 2-month baseline period. The hydrophone was located 2.3 (inbound) and 5 km (outbound) from the center of the shipping lanes at 23 m depth. Analyses of data with ships present showed a median broadband noise reduction of 2.5 dB. This reduction was highest in the 10–100 Hz decade band (3.1 dB) and lowest in the 10,000–100,000 Hz band (0.3 dB). A statistical model found that received noise levels were best described by the distance to ships, the presence of small boats, water velocity, slowdown period, ship speed through water, and wind speed. This highlights that appropriate temporal scales and the inclusion of covariate data are needed to adequately measure ship related changes in underwater ambient noise levels.

4:30

1pUWa14. The preliminary study on spatial correlation of ocean sound field. Jin-bao Weng and Yan-ming Yang (Third Inst. of Oceanogr., State Oceanic Administration, No. 178 Daxue Rd., Siming District, Xiamen 361005, China, wengjinbao@tio.org.cn)

The sound field temporal correlation and spatial correlation, which are the foundation of the investigation of underwater signal space-time high-order characteristics, have important value in the underwater acoustic application. The spatial correlation is studied based on the shallow water acoustic propagation experiment data acquired in the northern South China Sea in 2017, and the deep water acoustic propagation experiment data acquired in the western Pacific in 2013. As for the explosive sound signals in shallow water, time domain waveform cross-correlation coefficients between signals from different propagation distance are calculated. In contrast, the linear frequency modulated signals in deep water need additional matched filtering. The signal processing results shows that, the overall spatial correlation is poor and the correlation radius is relatively small in shallow water, the convergence zone has an obviously better spatial correlation than the shadow zone for the deep water situation. The processing result is verified by simulation and analysis.

4:45

1pUWa15. The preliminary study on fluctuation of ocean sound field. Jin-bao Weng and Yan-ming Yang (Third Inst. of Oceanogr., State Oceanic Administration, No. 178 Daxue Rd., Siming District, Xiamen 361005, China, wengjinbao@tio.org.cn)

The ocean environmental variation causes the amplitude and phase fluctuations of acoustic signals that travel in the ocean acoustic waveguide. Sound propagation fluctuation is studied based on the deep water fixed-point sound propagation experiment data acquired in the northern South China Sea in 2016. In this experiment, two sets of submerged buoy systems, which integrate the emission and receiving of sound wave, have continuously and steadily worked as long as three months. Considering the Doppler frequency shift, the amplitude and phase of acoustic signals are calculated after matched filtering. The sound intensity scintillation index computed for the entire record is 0.8, which is less than the saturation value of one. In contrast, the scintillation index computed for the 2001 ASIAEX South China Sea Experiment is 2.6 and 1.7. The ocean environmental variation and its relation with sound propagation fluctuation are analyzed.

5:00

1pUWa16. Multi-shipping sources classification method based on the line spectral components of shipping radiated noise. Hongjian Jia and Xiukun Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, jiahongjian_hrbeu@outlook.com)

The passive target detection based on shipping radiated noise is the key technology for underwater operations today. It has high research value both in civil and military applications. In this paper, the feature extraction of the underwater channels in the cepstrum domain is performed by using the line spectral components of the shipping radiated noise. The channel impulse response (CIR) function is used as the classification basis for the multi-shipping sources, and the line spectral components belonging to the same shipping source in the sonar received signal are searched to solve the classification problem of the multi-shipping radiated noise sources in the complex ocean background.

MONDAY AFTERNOON, 5 NOVEMBER 2018 OAK BAY 1/2 (VCC), 1:00 P.M. TO 4:45 P.M.

Session 1pUWb

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Variability in Shallow Water Propagation and Reverberation II

Brian T. Hefner, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

David R. Dall'Osto, Cochair
Acoustics, Applied Physics Laboratory at University of Washington, 1013 N 40th St., Seattle, WA 98105

Contributed Papers

1:00

1pUWb1. Excess attenuation caused by subducted bubbles at the ebb plume front in the Connecticut River. D. Benjamin Reeder, John E. Joseph, and Tarry Rago (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu)

A field experiment was carried out in the Connecticut River in June 2017, one goal of which was to investigate the low-to-mid-frequency acoustic propagation characteristics of the fresh water plume front outside the river mouth during ebb. Linear frequency-modulated (LFM) acoustic signals in the 500-2000 Hz band were collected during several tidal cycles. Transmission loss in excess of that which is expected by sound speed gradient-driven refraction is explored via acoustic propagation modeling. The excess attenuation at the ebb plume front is attributed to large clouds of subducted bubbles by downwelling at the ebb plume front.

1:15

1pUWb2. Temporal variability of propagation and reverberation during TREX13. Brian T. Hefner, Jie Yang, Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), and William S. Hodgkiss (UC San Diego, La Jolla, CA)

During the Target and Reverberation Experiment in 2013, a bottom-mounted, mid-frequency source and horizontal line array were deployed on the seafloor to measure reverberation over the course of the experiment. Extending from the source and parallel to shore was the "main reverberation track," a 7-km long and 1.5 km wide region where a majority of the acoustic and environmental measurements were made. Along this track, two vertical line arrays were deployed to continuously collect propagation data. The combination of the HLA and VLAs made it possible to make contemporaneous measurements of both transmission loss (TL) and reverberation level (RL) over a 3-week timespan. During this time, a storm passed through the site and the variability in the TL is found to be driven by different mechanisms. Prior to the storm, the variability was primarily driven by changes in the sound speed profile; after the storm, the water was well-mixed and the variability was a steady decay of TL while the rough sea surface decayed. This talk examines the variability in TL during these time periods and the associated changes in the measured RL. [Work supported by ONR.]

1:30

1pUWb3. Variability of target echo and clutter in the 2013 Target and Reverberation Experiment. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com), Jie Yang, Brian T. Hefner, and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Target and Reverberation Experiment was conducted in the Gulf of Mexico, off Panama City, Florida. Reverberation, noise, and target echo measurements were carried out for more than a month during April and May 2013, using a fixed source and fixed horizontal array receiver deployed in about 20 m of water. To support modeling efforts, a considerable number of ancillary environmental and transmission loss measurements were made. Various pulses in the 1.8–3.6 kHz frequency band were transmitted day and night, and have produced a rich data set. Initial results by various authors were published in the recent TREX13 Special Issue of the IEEE Journal of Oceanic Engineering. This work extends the analysis, concentrating on the variability of the echoes from various targets: a vertical hose, vertical arrays, a towed echo repeater, the hull of the towing ship, and other persistent bottom scatters, such as shipwrecks. In addition to the ping-to-ping variability of the echoes there are consistent trends, which are not understood, though probably due to oceanographic effects. An attempt will be made to relate the echo variability to measured and modeled reverberation and transmission loss. [Work supported by ONR, Ocean Acoustics, Code 22.]

1:45

1pUWb4. Characterizing acoustic propagation within the approach channel to Brandon Road Lock and Dam. Samuel L. Denes, David Zeddies (JASCO Appl. Sci., 8630 Fenton St., Ste 218, Silver Spring, MD 20910, sam.denes@jasco.com), and Christa Woodley (Engineer Res. and Development Ctr., US Army Corps of Engineers, Vicksburg, MS)

A propagation experiment was conducted in the approach channel to the Brandon Road Lock and Dam on the Des Plaines River in Joliet, Illinois, over the course of ten days in 2017. A sequence of frequency sweeps, continuous tones, and white noise was broadcast from an underwater transducer deployed from a shallow draft workboat at nine locations within the channel. A vertical array of one tri-axial particle motion sensor and three hydrophones was deployed from a movable platform and recorded signals throughout the approach channel. An additional bottom mounted, autonomous recorder with one hydrophone and one particle motion sensor was deployed outside of the navigable channel for the duration of the experiment. The engineered channel includes cement walls and floors near the steel lock doors, transitioning to natural surface shoreline down channel. For sources or receivers near the lock, the highly reverberant interfaces dominate propagation effects. This presentation will compare results of the measurement experiment with propagation models.

2:00

1pUWb5. Measured variability of acoustic propagation loss in the Salish Sea, British Columbia, Canada. Caitlin O'Neill (Fisheries and Oceans Canada, 9860 West Saanich Rd., Sidney, BC V8L 4B2, Canada, caitlin.v.oneill@gmail.com), Harald Yurk (Fisheries and Oceans Canada, West Vancouver, BC, Canada), and Svein Vagle (Fisheries and Oceans Canada, Sidney, BC, Canada)

The Salish Sea encompasses the waters between southern British Columbia mainland coast, Vancouver Island, and Washington State. Its waters are used by commercial and recreational fishers, host shipping traffic to the major ports of Vancouver and Seattle, and is travelled frequently by local ferries. It is also an important habitat for many marine mammal species, including the endangered Southern Resident Killer Whale (*Orcinus orca*). Southern Resident Killer Whales use calls, whistles, and clicks to communicate and find prey. Human activities on the water may affect the marine acoustic environment by masking whale vocalizations, which in turn will negatively impact the ability of the whales to forage and socialize. A field study was conducted to measure acoustic transmission loss at various places in the Salish Sea. Sound signals simulating killer whale call elements were broadcast from different angles to a calibrated hydrophone. Results from the study showed that acoustic transmission loss is highly spatially variable in this area. We suggest that these observations are due to changes in oceanography conditions and ambient sound levels, which are important factors to consider when assessing impact on Southern Resident Killer Whales. [Work supported by the Oceans Protection Plan of the Government of Canada.]

2:15–2:30 Break

2:30

1pUWb6. Confined space acoustic studies in Weymouth Fore River. Timothy F. Duda, Arthur Newhall, Peter Traykovski (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Michael B. Porter (Heat, Light and Sound Res., Inc., San Diego, CA), and Laurel Henderson (Heat, Light and Sound Res., Inc., La Jolla, CA)

An acoustic propagation study with two moving sources and three receiver positions was conducted in May 2018 in Weymouth Fore River, Massachusetts. Detailed bathymetric and geometric boundary information was previously collected. The data show strong inter-ping variability at ping rates of four Hz and one Hz. Signals were linear sweeps from 8-16 kHz and 21-34 kHz. The variation of the matched-filter envelope output, thought to be robust to phase alterations from small movements, was highly variable. This implies interfering multipath surpassing slowly varying response to boundary-interaction changes for the slowly varying source. The signatures of boundary interaction in models are compared to data to evaluate modeling procedures in the BELLHOP3D program. Interactions with concrete boundaries and steel ship and barge hulls are included in the analysis.

2:45

1pUWb7. A full-field time-domain model for reverberation in complex environments and waveguides with rough interfaces. Anatoliy Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A full-field perturbation approach [Ivakin 2016] has been used for modeling reverberation at given frequency in spatially varying layered environments and waveguides with rough interfaces. Average reverberation intensity was shown to be related with a local Bragg-wavenumber power spectrum of roughness through integration over the reference (unperturbed or smoothed) interfaces. The integrand includes a factor, a two-way propagation/interaction kernel, defined in terms of Green's function and a local contrast of acoustic parameters at the interfaces. This work extends the approach to time domain and includes additional integration over frequencies with another factor in the integrand, the radiated energy time-frequency distribution (ambiguity function). However, the integration within the signal frequency bandwidth can be significantly simplified using the frequency-range interference invariant. As a result, extremely fast estimations of reverberation time series (codas) can be made which require calculations of Green's function of the reference medium in the vicinity of the interfaces at only one (central) frequency. Numerical examples for time dependence of narrow- and broad-band reverberation in layered environments and waveguides with rough interfaces using PE-approximation for Green's function are presented. A possibility for remote monitoring of temporal and spatial variability of ocean interfaces based on propagation and reverberation measurements is discussed. [Work supported by ONR.]

3:00

1pUWb8. A study of underwater propagation model performance in complex environments with inherent uncertainty. Mark Langhirt (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mal83@psu.edu), Sheri Martinelli, and Charles W. Holland (Marine & Physical Acoust., ARL Penn State, State College, PA)

A number of computational models have been developed for purposes of modeling and simulation of acoustic wave propagation in underwater environments. Almost all of these models make approximations to the wave equation; furthermore, the method of implementation has a direct impact on the trade-off between accuracy and computational efficiency. Sonar technologies also vary greatly in range and frequency and one model is generally not suited for all applications; there are often subtle differences between them that can be highly dependent on the particulars of the environment. We investigate the performance of various models (ray tracing, level sets, normal modes, parabolic equation, and energy flux) using at-sea data from real world environments to assess their robustness of in the presence of the inherent uncertainty involved in reconstructing an underwater environment. In particular, we focus on the effect of perturbations in the sound speed profile. The sound speed profile, as a model input parameter, is subject to measurement error, as well as to undersampling of what, in many geographical areas of interest, is a complex spatial-temporal process. Ultimately, our objective is to establish some guidelines for practical use of the models in applications.

3:15

1pUWb9. Perturbation approximations for bathymetrically controlled ocean acoustic mode coupling. Brendan J. DeCourcy and Timothy F. Duda (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Iselin 3 (MS# 55), Falmouth, MA 02543, bdecourcy@whoi.edu)

A perturbation approximation of mode coupling equations for acoustic propagation in a submarine canyon environment is presented. The vertical modes and mode coefficients are represented by nearly adiabatic dominant mode functions and correction terms that are sensitive to the horizontal gradient of the variable bathymetry. This approach allows for a more accurate representation of the bottom boundary condition, and eliminates

computationally expensive implicit equations for the mode coefficients, replacing them with explicit equations for the nearly adiabatic contributions and the bathymetrically sensitive corrections. The presented method is applied to sloping bottom environments based on real bathymetric data to examine in detail the role of bottom variations in coupled mode propagation scenarios. [Work supported by the ONR.]

3:30

1pUWb10. Stage-wise orthogonal matching pursuit for estimation of time delay and Doppler doubly spreading of underwater acoustic channels. Feiyun Wu, Kunde Yang, Quan Sun, and Yunchao Zhu (Northwestern PolyTech. Univ., Youyi Rd. 127, Xi'an, Shaanxi 710072, China, wfy@nwpu.edu.cn)

The model of time delay and Doppler (DD) doubly spread is usually used for estimation of underwater acoustic channel (UAC) of time-varying and multi path. This study introduces the doubly spread model, then forms the optimization problem of channel estimations. By exploiting the sparsity of UAC, we propose the stage-wise orthogonal matching pursuit by inserting the orthogonal projection strategy on two domains. The traditional methods including least square (LS) based method, matching pursuit (MP), and orthogonal matching pursuit (OMP) methods are used for comparisons based on doubly spread models. Finally, the superiority of the proposed method is verified by experimental results.

3:45

1pUWb11. Geoacoustic inversion for bottom parameters in the deep-water area of the South China Sea. Shuanglin Wu, Zhenglin Li, and Jixing Qin (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring West Rd., Haidian District, Beijing 100190, China, wushuanglin@mail.ioa.ac.cn)

The properties of bottom acoustic parameters are greatly influenced on sound field prediction. Acoustic parameters in deep water are not well understood. We will discuss experience with geoacoustic inversion of transmission-loss (TL) data from 100 to 600 Hz from deep water regions. Bottom acoustic parameters are sensitive to the TL data in the shadow zone of deep water. Multiple steps TL inversion method is proposed to invertsound speed, density and attenuation in deep water. Both a uniform liquid half-space and a two-layer geoacoustic profile are assumed. Sound speeds of bottom are inverted by using TL data in the shadow zones at low frequency obtained in an acoustic propagation experiment conducted in the South China Sea (SCS) in summer 2014. Meanwhile, bottom densities are estimated combining with the Hamilton sediment empirical relationship. Attenuation coefficients at different frequencies are then estimated from the long range TL data by using the known sound speeds and densities as a constraint condition. The nonlinear relationship between attenuation coefficient and frequency is given in the end. The inverted bottom parameters can be used to forecast the transmission loss in deep water area of SCS very well.

4:00

1pUWb12. Extracting the intrinsic geo-acoustic parameters from the effective geo-acoustic parameters. Juan Zeng, Z. D. Zhao, Li Ma, and E. C. Shang (Inst. of Acoust., No. 21 Bei Si Huan Xi Lu, Beijing 100190, China, zengjuan@mail.ioa.ac.cn)

So far, most of the geoacoustic (GA) parameters are inverted through model-based approaches. The inverted GA parameters are not the intrinsic ones but the so-called "effective" ones. The "effective" GA parameters are model and frequency dependent and will be distorted due to model-mismatching. The real physical meaning of the term "effective" has not been fully addressed yet. In this paper we proposed to interpret the term of "effective" based on the Brekhovskikh's reflection principle, which reveals how the GA parameters influence the Green's function through the only channel, i.e. the bottom reflective coefficient. Based upon this point, we discuss the relationship between the "effective" GA parameters and the intrinsic GA parameters. First of all, it is shown that the real essential physical parameters, the reflective parameters, P(f) and Q(f) can be extracted from the "effective" GA parameters. Finally, with the help of P(f) and Q(f) some of the intrinsic GA parameters, the frequency dependence of the seabed attenuation and seabed sound speed profile can be retrieved.

4:15

1pUWb13. Range-dependent geoacoustic inversion for sound propagation in two seamounts environment. Lichen Pang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, lichenpang@163.com), Tao Hu, Licheng Lu, Qunyan Ren, and Li Ma (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper presents a work of range-dependent geoacoustic inversion for two seamounts environment with negative gradient sound speed profiles based on the transmission losses (TLs), collected in an experiment in the South China Sea, 2016. Bombs of 200 m and the mooring vertical line array (VLA) is applied for recording. Typical characteristics of shadowing and reflecting by seamounts are confirmed by the fluctuations of TLs with respect to range, indicating the complexity of sound propagation within seamounts environment. In the framework of matched field inversion (MFI), the geoacoustic model of two layers and the range-dependent acoustic model (RAM) is applied to produce the replica TLs, which is used to compute the cost function in combination with the collected TLs in terms of the least mean variance. The simulated annealing (SA) is expected to augment the searching efficiency of geoacoustic parameters. Results of four models with different parameters within 120km are obtained, with which the simulated TLs are in good accordance with experiment results for several frequencies and receiving depths. Conclusion is drawn that the effects of range-dependence of geoacoustic parameters in seamounts environment are as important as those of bathymetry on sound propagation, which should be seriously considered especially with negative gradient sound speed profiles.

4:30

1pUWb14. Comparison of velocity and attenuation measurements to the corrected effective density fluid model for gassy sediments. Fei Wang and Yiwang Huang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, wangfei0618@hrbeu.edu.cn)

To validate the corrected effective density fluid model for gassy sediments, an acoustic experiment was undertaken to measure the velocity and attenuation over the frequency range of 8 and 120 kHz in an indoor water tank. The velocity and attenuation obtained by different estimation methods have good consistency respectively. Both of them have four distinct peaks and there is one-to-one correspondence between velocity peaks and attenuation peaks. Through numerical simulation of the acoustic characteristics attributed to the resonance of gas bubbles with different radii, the existence of small bubbles causes the velocity to decrease in the vicinity of resonance frequency of large bubbles. Five modified Gaussian functions are introduced to model the bubble size distribution inside the gassy sediment. A best bubble size distribution is obtained by inversion of the velocity and attenuation data simultaneously, and the predictions of the corrected model are broadly in good agreement with the measurements.

Payment of an additional fee is required to attend this session.

MONDAY EVENING, 5 NOVEMBER 2018 SALON A (VCC), 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on An Introduction to Sound in the Sea

James P. Cottingham, Chair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. An introduction to sound in the sea. Thomas Dakin (Ocean Networks Canada, Univ. of Victoria, TEF-128A 2300 McKenzie Ave.,Victoria, BC V8W2Y2, Canada, tdakin@uvic.ca)

This introductory talk will address why underwater acoustics are important, its uses, how sound propagates so far in the ocean, man-made sounds, including shipping noise, and the impact of noise in the ocean. A short overview of organizations working on the issue of underwater noise will be given including those around the Salish Sea where the conference is being held. The way sound is measured and analyzed will be shown with local examples of marine life, earthquake and anthropogenic noise. A short exercise to show the effects of acoustic masking will be given, followed by an explanation of echolocation use by the Southern Resident Killer Whales and the implications of acoustic masking on finding food for this local and endangered species.

1p MON. PM