Session 1aAA

Architectural Acoustics, Noise, Signal Processing in Acoustics, and Speech Communication: Integrated Approach to Speech Privacy

Kenneth W. Good, Cochair
Armstrong, 2500 Columbia Ave., Lancaster, PA 17601

Eric L. Reuter, Cochair
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Chair's Introduction—9:30

Invited Papers

9:35

1aAA1. Speech privacy measurements and metrics. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

What is the current state of speech privacy measurements and metrics for built environments? This paper will provide an update on findings from the TCAA Speech Privacy Subcommittee. We will explore the uses, pros, cons, and appropriateness of Speech Privacy Potential (SPP) speech Privacy Index (PI) and Speech Privacy Class (SPC).

9:55

1aAA2. Speech privacy comparison of metrics. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

How do different speech privacy metrics compare, what are the correlations and differences, and what might we conclude from them? This paper will start with the noise reduction results from several common architectural assemblies. From those results, we will calculate each of the popular speech privacy metrics. Finally, we will explore the correlations and differences of the results.

10:15

1aAA3. A review of speech privacy terms and methodology in building codes, guidelines, and standards. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Terminology and methodology for the measurement and classification of speech privacy in buildings are globally diverse and cover speech transmission in a relatively broad range of environments from large open spaces to small divided enclosures. This paper is meant to provide a review and summary of terms and methodologies related to speech privacy throughout the global building noise control industry and regulatory organizations. Differences in terminology and speech transmission measurement between standards will be discussed as well as the progressive use of these standards in building codes and guidelines. A more thorough examination of this documentation is intended to provide the building noise control community with both greater insight into the incorporation of speech privacy in regulatory and standard documentation as well as to provide a foundation for greater innovation in building noise control specific to speech transmission.

10:35

1aAA4. Statistical distribution of ambient noise levels compiled from building acoustics measurements. John Loverde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

One of the primary factors affecting speech privacy is the background noise level. During the course of building acoustics testing by the authors' company, the background noise level has been measured in hundreds of locations nationwide in a wide variety of site conditions and building types. This information is compiled and analyzed. The effect of the statistical expectation of the background noise level on the required level of sound reduction is discussed.

10:55

1aAA5. An investigation of the significance of frequency weighting in speech intelligibility calculations. Ric Doedens (K.R. Moeller Assoc. Ltd., 1050 Pachino Court, Burlington, ON L7L 6B9, Canada, rdoedens@logison.com)

Differing approaches to quantifying speech intelligibility exist within ASTM architectural acoustic standards E1130 and E2638. E1130 was written for open office conditions and to cover a wide range of intelligibility levels. E2638 was written for enclosed room scenarios and focuses on speech privacy at confidential levels or higher. Each is currently being evaluated with respect to the possibility of expansion to include both open and closed settings. One difference between the two methodologies pertains to whether background sound level frequencies are weighted or averaged. This paper explores how frequency weighting of background sound levels is significant in the determination of speech intelligibility/privacy/security.

11:15

1aAA6. Privacy in a corporative office. Sergio Beristain (IMA, ESIME, IPN, P.O. Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

A corporative headquarters were installed in three floors of a new ten story building, where all the needed spaces were properly distributed as requested by the customer, which included private offices, meeting rooms, open plan offices, etc., the former two were located in one end of the two highest floors chosen, away from the stairs and elevator. Within weeks after the inauguration of the venue, the company directives noticed that some key design and production information was being leaked somehow and were not able to find out the process, which was a major problem for the company. They requested a quick and efficient solution to this matter, because they could not stop their activities, and they were not willing to lose any more information.

Contributed Paper

11:35

1aAA7. Reverberation analysis and acoustical modeling for improved acoustical conditions within modern workplace phone rooms. Alex Maurer, Jeffrey Fullerton (Intertek, 50 Summer St., Boston, MA 02110, alexander.maurer@intertek.com), and Kimteri Kim (Intertek, New York, NY)

As open office floor plans have grown popular, phone rooms have become important for staff to conduct personal conversations in private. Varying in acoustic and holistic design, these rooms are intended to both isolate the occupant's speech from being heard by others in the office and reduce intrusive noise from disrupting these phone conversations. The acoustical properties within the rooms are often ignored, even though these spaces can have rather reverberant conditions and be subjectively very bothersome and awkward to converse within. Several types of porous absorption can help to control the acoustics of these phone rooms which are commonly implemented today. We have studied a variety of absorptive panel samples, placements, and coverage areas to test the effect of absorption in the phone rooms. We measured the reverberation time and impulse response of the rooms with various acoustical finishes to better understand the effectiveness of porous absorption to control the acoustics of these phone rooms.

Session 1aAO

Acoustical Oceanography: Future Directions in Acoustical Oceanography

Timothy F. Duda, Cochair
Woods Hole Oceanographic Institution, WHOI AOPE Dept. MS 11, Woods Hole, MA 02543

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

Chair's Introduction—8:30

Invited Papers

8:35

1aAO1. Echoes from the ocean's interior: High-frequency observations of ocean phenomena. Thomas C. Weber, Larry Mayer, Anthony P. Lyons, Scott Loranger, Alexandra M. Padilla, and Elizabeth F. Weidner (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Recent technological advances in high-frequency (>10 kHz) sonar transducers, sonar transceivers, and sonar design have been accompanied by increased capabilities for observing ocean phenomena. These advances include the high range resolution and frequency-domain target classification capabilities associated with wideband acoustic echo sounders, the long-range high-resolution synoptic imaging capabilities associated with multibeam echo sounders and synthetic aperture sonar, and an increased focus on sensor calibration for all systems. High-frequency sonars are increasingly being used to quantify ocean phenomena at scales ranging from sub-centimeter (e.g., individual gas bubbles) to 100s of km (e.g., internal waves) to several 10s of km (e.g., thermohaline staircases). In this talk, we highlight some of the ocean processes that we have been investigating using high-frequency sonar systems, typically involving the transport of hydrocarbons, heat, energy, and fresh water into and through the ocean, and some of the (many) acoustic challenges that must be overcome to continue to increase the value of these observations.

8:55

1aAO2. An advanced sensor platform for acoustic quantification of the ocean twilight zone. Andone C. Lavery, Timothy K. Stanton (Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu), J. Michael Jech (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), and Peter Wiebe (Woods Hole Oceanographic Inst., Woods Hole, MA)

The ocean twilight zone (OTZ) is the vast, globe-spanning, layer of water between 200 and 1000 m depth—home to diverse communities of small fishes, cephalopods, crustaceans, and gelatinous organisms. Yet, little is known about the biology, abundance, biomass, and distribution of these organisms. The OTZ is difficult to sample due to a combination of organism patchiness and avoidance, and difficulties capturing fragile species. Recent evidence suggests that the global OTZ biomass may be sufficient to commercially harvest. Furthermore, much of this biomass performs daily vertical migration (DVM) and may play a potentially critical role in regulating Earth's climate through the export of carbon to the deep ocean. *Deep-See*, an advanced senor platform, was developed to fill the technological void for characterizing the OTZ. This towed vehicle integrates wide-band, split-beam acoustics (1–500 kHz) with optical, environmental, and eDNA sensors that can address many of the challenges associated with sampling in the OTZ. Data from the inaugural cruise in August 2018 highlight that (i) a surprisingly high abundance of organisms can be found outside the dense sound scattering layers and (ii) the target strength of many organisms that perform DVM changes with the depth, which is critical to estimate biomass.

9:15

1aAO3. Sound propagation in the surface mixed layer and upper ocean: An overview of relevant ocean dynamics and mode scattering theory. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

The mixed layer and upper ocean are a region of immense interest to physical oceanographers, meteorologists, and climate scientists because this is the boundary through which energy, momentum, buoyancy, and gasses are exchanged between the ocean and the atmosphere. The upper ocean is also a rich ecosystem for a vast array of ocean organisms and marine wildlife as well as a region of great Naval tactical importance. Given these factors, it is rather surprising that ocean acoustics has paid little attention to this significant region, aside from very high-frequency studies of bubble properties and gas entrainment. From the standpoint of transmission loss, the major work on the problem seems to go back to the Acoustic, Meteorological, Oceanographic Survey (AMOS) of the mid-50s. Here, a review is presented of relevant ocean processes that may be important for acoustic propagation at frequencies ranging from several hundreds of hertz to several kilohertz, where the foundation of the analysis is normal mode and transport theory. Processes of interest are surface gravity waves including subsurface currents, internal waves and tides, wind driven inertial oscillations and Langmuir circulations, eddies and submesoscale processes, turbulence and spice, bubbles, and fishes.

9:35

1aAO4. Ocean remote sensing using passive acoustics. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Conventional acoustic remote sensing techniques typically rely on controlled active sources which can be problematic to deploy and operate over the long term, especially if multiple sources are required to fully illuminate the ocean region of interest. Conversely, receiver arrays are becoming increasingly autonomous, miniaturized, and capable of long term deployment thus enabling passive acoustics for ocean remote sensing applications by taking advantage of uncooperative sources of opportunity (e.g., shipping noise) or the ubiquitous ocean ambient noise which have not typically been used in traditional ocean sensing applications. This fully passive approach can also be advantageous when regulations forbid the use of active sound sources or when no active sources are readily available—e.g., at very low frequencies (~10 Hz) or during covert operations. This presentation will discuss the recent development of ocean remote sensing using passive acoustics notably (1) passive acoustic thermometry to estimate deep ocean temperature variations and internal tides using coherent processing of low-frequency ambient noise; (2) localization of drifting sensor networks using ambient noise to enable random volumetric *ad hoc* receiver array for tracking underwater targets; and (3) monitoring of the shallow water sound channel using shipping sources of opportunity.

9:55

1aAO5. Exploiting ambient noise in polar regions to study ice-ocean interactions. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238, gdeane@ucsd.edu)

We explore recent developments and future directions for ambient noise cryology: the study of ice-ocean interactions using their underwater noise signatures. The study of ice-ocean interactions is currently spurred by climatic shifts in polar regions and their implications, which include sea level rise and geopolitical stability. There are many important ice-ocean interactions, and a broad range of observational techniques are used to study them, such as satellite remote sensing, ship-based observations with *in situ* sensors and AUV's, boreholes, ground-penetrating radar, photogrammetry, seismometry, and differential GPS. Despite such an extensive suite of techniques, submarine calving and ice melting—which play a critical role in the mass balance of ice shelves and marine-terminating glaciers—remain difficult to measure. Progress in quantifying these processes with their underwater noise signatures will be discussed along with future directions for the field.

10:15–10:30 Break

10:30

1aAO6. Better together—Combining acoustics and environmental DNA to understand ecosystems. Jennifer L. Miksis-Olds (School of Marine Sci. & Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NC 03824, j.miksisolds@unh.edu) and Alison Watts (Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH)

Acoustic signals have historically been and presently are the state-of-the-art for sensing the ocean at small to large spatial scales. Passive acoustics non-invasively assess sound levels, surface conditions, human activity, and the distribution of vocalizing marine life. Echosounders provide acoustic backscatter information that contribute not only critical information on biology but also physical components of the water column which has supplied invaluable knowledge on the community structure, organism size and distribution, and oceanic microstructure. Advances in DNA methods present an opportunity to harness a new technology and fundamentally improve our capacity to monitor habitats, communities, and individual species. Environmental DNA (eDNA) includes whole microorganisms, tissue fragments, reproductive and waste products, and other cellular material in a sample. eDNA methods, such as acoustics, allow for the identification of species without having to physically capture animals. While eDNA and acoustics are both powerful methods for identifying and describing organisms present in the environment, no single survey method can fully represent the "real" condition. We envision a future where remote platforms can gather and relay data in near-real time and can elicit a targeted response to information, such as launching a drone to survey additional locations or increasing the sampling frequency when key species are present.

Contributed Papers

10:50

1aAO7. Low frequency acoustical scattering from dynamic schools of swim bladder fish. Luis Donoso and Christopher Feuillade (Inst. of Phys., Pontifical Catholic Univ. of Chile, Av. Vicuna Mackenna 4860, Macul, Santiago, Region Metropolitana de Santiago 7820436, Chile, lldonoso@uc.cl)

A time-domain computational model is used to describe the dynamic evolutions of fish schools. The individual and ensemble fish behaviors are governed by three radial parameters, representing attraction, orientation, and repulsion zones between fishes. Different combinations of these radii cause the schools to evolve into various discoid, swarming, parallelized, or toroidal geometric forms. A previously developed model [*J. Acoust. Soc. Am.* **99**, 196–208 (1996)] is applied to study the variations in ensemble scattering from these school types for ensonification frequencies in the swim bladder resonance region, as a function of frequency and time, and predicts distinct characteristic features for the different school geometries. In particular, this work focuses on scattering from disk-shaped schools, by examining the computed resonance response for different school dimensions, packing density, and orientation, in order to identify specific features that are determinative for this type of arrangement. The ultimate goal of this work is to achieve a better understanding of the physical basis for variations in fish school scattering levels and to obtain a detailed statistical description of these objects. [Work supported by ONRG.]

11:05

1aAO8. The measurement of ocean acidity using the depth-dependence of ambient noise. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., P.O. Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca) and Michael J. Buckingham (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA)

The absorption of sound in seawater is due to the viscous and chemical relaxation of different compounds. Over the wind noise band of 1–10 kHz, the frequency dependence of the absorption is due to the mechanisms of chemical relaxation for magnesium sulfate ($f > 3$ kHz) and for boric acid ($f < 3$ kHz), which involve ionic dissociations activated and deactivated by the condensation and rarefaction of the medium by passing sound waves. Concentrations of both chemicals determine the level at which the sound is absorbed, which makes the process dependent on the salinity, temperature, and pressure in the ocean. The concentration of boric acid in the ocean is a direct measure of pH, while the concentration of magnesium sulfate is independent of pH; thus, a measurement of the frequency dependence of sound absorption may be used to determine ocean acidity. When local winds are strong (> 10 m/s), the ambient noise field is dominated by locally generated surface noise and has a depth-independent directionality and a weakly frequency and depth-dependent intensity, due to sound absorption. By comparing measurements with theory, estimates of ocean acidity can be made from the depth profiles of ambient noise. [Work supported by ONR.]

11:20

1aAO9. Three-dimensional multichannel seismic imaging of water columns in the Gulf of Mexico. Likun Zhang (National Ctr. for Physical Acoust., Univ. of MS, Oxford, MS), Zheguang Zou, Parsa Rad, and Leonardo Macelloni (National Ctr. for Physical Acoust., Univ. of Mississipi, University, MS, zou@olemiss.edu)

Seismic reflection profiling technique, previously used to image the sediments beneath the seafloor, is herein used to image the ocean's water columns, namely, seismic oceanography. The imaging has a much higher lateral resolution (~10 m) than traditional oceanographic measurements such as CTD (usually >100 m). Prior work on seismic oceanography was limited on imaging in two dimensional vertical transects. This work develops the three-dimensional (3D) seismic oceanography technique to image the 3D dynamic processes of water columns. 3D multichannel seismic survey data in a seismic volume of 625 km^3 in the Gulf of Mexico are processed and produce images containing detailed 3D water-column structures near the continental slope. Some mesoscale and sub-mesoscale structures are visualized from different viewing angles. Spectral analyses of the seismic images reveal 3D spatial features of the structures, suggesting the potential of 3D seismic oceanography. [Work supported by NOAA.]

11:35

1aAO10. The soud of light: Towards ocean acoustic sensing with an optical breakdown transducer. Athanasios G. Athanassiadis (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Bldg. 3-257c, Cambridge, MA 02139, thanasi@mit.edu)

When a high-power laser is focused to a small spot in a fluid, nonlinear interactions at the focus can excite a plasma that evolves and glows according to the properties of the breakdown medium. This phenomenon—optical breakdown—is commonly used for underwater chemical measurements in a technique called Laser-Induced Breakdown Spectroscopy (LIBS). However, LIBS generally operates at short ranges and only leverages the optical emission of the plasma for sensing. If instead, the laser systems were tuned to amplify the mechanical effects of optical breakdown, then it could be deployed as a versatile source for broadband acoustic sensing. The optical breakdown acoustic source is simultaneously compact (mm-scale), loud (MPa peak pressures), ultra-broadband (10 kHz–4 MHz), and geometrically reconfigurable. Here, I will describe the physics governing optical breakdown transduction and then show how can this be tuned to enable new single-vehicle sensing strategies in the ocean.

MONDAY MORNING, 13 MAY 2019 NUNN, 8:30 A.M. TO 10:30 A.M.

Session 1aBAa

Biomedical Acoustics: Ultrasound Modeling Workshop

Robert McGough, Chair
Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824

A 2-hour hands-on workshop using FOCUS, the 'Fast Object-oriented C++ Ultrasound Simulator' will be held on Monday 13 May, in Session 1aBAa, at 8:30 a.m. in Nunn. Preregistration was required so participation will be on a space-available basis.

Session 1aBAb

Biomedical Acoustics and Signal Processing in Acoustics: Lung Ultrasound and Tissue Stiffness Method I

Xiaoming Zhang, Cochair
Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

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

Chair's Introduction—11:00

Invited Papers

11:05

1aBAb1. The prognostic role of ultrasonographic air bronchogram in the management of community acquired pneumoniae in children. Riccardo Inchingolo (UOC Pneumologia, Fondazione Policlinico Universitario A. Gemelli IRCCS, Largo Gemelli, 8, Roma 00168, Italy, riccardo.inchingolo@policlinicogemelli.it), Andrea Smargiassi (UOC Pneumologia, Fondazione Policlinico Universitario A. Gemelli IRCCS, Rome, Italy), Roberto Copetti (Pronto Soccorso e Medicina d'Urgenza, Ospedale Civile Latisana, Latisana, Italy), and Gino Soldati (Ecografia Clinica, Lucca, Italy)

Chest ultrasound is a non-invasive method for evaluating children with suspected community-acquired pneumonia (CAP), allowing close follow-up and reduction of ionizing radiation. We studied the prognostic role of the change of ultrasonographic (US) air bronchogram in the management of CAP in terms of rate of complicated CAP, change in empiric antibiotic therapy, relationship to defervescence time, and length of hospitalization. Patients with diagnosis of CAP and radiographic evidence of lung consolidation were prospectively enrolled. The first chest US examination was performed within 12 h from admission and after 48 h. A new grading system (USINCHILD score) based on the presence and features of air bronchogram was adopted. Thirty six patients (mean age of 5 years) were stratified into two groups according to the presence of an increase in at least 1 grade of US score (Δ US grade), expression of an improvement of lung consolidation. The US grade after 48 h < 1 was associated with an increased risk of complicated CAP (OR: 160.88, p-value: 0.0109) and a longer defervescence time (70 h, p-value: 0.0047). Moreover, Δ US grade ≥ 1 was predictive of a short hospitalization (7 days, p-value: 0.0061). USINCHILD score could be an innovative biotechnology tool for the management of pediatric CAP.

11:25

1aBAb2. The ultrasound transmissible lung: Impact of acoustic conditions in flooded lung on therapeutic ultrasound applications for lung tumour treatment. Frank Wolfram and Thomas G. Lesser (Clinic of Thoracic Surgery / Lung Cancer Ctr., SRH Waldklinikum Gera, Strasse d Friedens 122, Gera 07548, Germany, frank.wolfram@srh.de)

For minimal invasive treatment of Lung Cancer and Metastases, thermal ablation is a valuable tool in case of in-operability. Such a radiofrequency (RFA) is widely used despite its noticeable complication rate. The use of non-invasive modalities such as High Intensity Focused Ultrasound (HIFU) would be beneficial in lung, if acoustically transmissible. For such purposes, One Lung Filling (OLF) was developed, which replaces air with the saline content in lung. Suppositious, acoustic conditions in such a saline tissue compound might be different than in solid tissue and impact the HIFU ablation process in lung. Therefore, our work was dedicated to develop a valid acoustic model of lung in flooded conditions and validate it with findings on the HIFU interaction during OLF on preclinical models. Acoustic parameters were determined using a broadband transmission technique which showed atypical but superior conditions for ultrasound transmission through lung. Heat induction was simulated based on the derived parameters in an acoustic-thermal solver using KZK and Penne's bioheat equation (BHTE). Results showed good agreement to the measurement where ablative temperatures are induced in central lung cancer, while lung parenchyma stayed unaffected thermally and due to cavitation up to intensities of 9.500 W cm^{-2} (p— 9.1 MPa). Based on those findings, future directions for clinical application of therapeutic ultrasound during OLF in lung will be discussed.

11:45

1aBAb3. Assessment of human diaphragm function by ultrasounds. Andrea Aliverti (Dipartimento di Elettronica, Informazione e Bioingegneria, Politecnico di Milano, Via G. Colombo, 40, TBMLab, Milano 20133, Italy, andrea.aliverti@polimi.it)

Thoracic ultrasound can provide a non-invasive technique for human diaphragm functional assessment, which can be used as an alternative to traditional, more challenging, and uncomfortable methods, such as transdiaphragmatic pressure measurement, fluoroscopic sniff test, nerve conduction studies, and electromyography. The variables that can be assessed using ultrasounds are (1) the static measurement of the end-expiratory diaphragm thickness (Tdi), (2) the dynamic evaluation of the ratio of inspiratory to the expiratory diaphragm thicknesses, reported as the thickening ratio or thickening fraction (TF), and (3) the diaphragmatic excursion. The measurements of Tdi and TF are performed by placing a high-frequency linear probe at the level of the zone of apposition, while diaphragm excursion is measured using a curvilinear probe placed in the subcoastal region and recording diaphragm movements in the M-mode. Intra- and inter-observer reliabilities of the measurement of Tdi and TF are high, and ultrasound estimates of Tdi are correlated to direct anatomical measurements. Tdi can be used to monitor the evolution of diaphragm weakness. The reduced values of TF are associated with diaphragmatic paresis. Diaphragm excursion is sensitive to changes in the respiratory pattern, related to diaphragm's volume generating capacity, and can be used to identify diaphragm weakness. In intubated patients, diaphragm excursion is related to weaning outcome.

MONDAY MORNING, 13 MAY 2019 SEGELL, 9:00 A.M. TO 11:55 A.M.

Session 1aNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Acoustics of Healthcare Facilities

Jay Bliefnick, Chair
Architectural Engineering & Construction, University of Nebraska, 1110 S 67th St., Omaha, NE 68182-0816

Chair's Introduction—9:00

Invited Papers

9:05

1aNS1. Quiet time impacts on the neonatal intensive care unit soundscape and patient outcomes. Jonathan R. Weber, Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jonryanweber@gmail.com), and Ashley Darcy Mahoney (School of Nursing, The George Washington Univ., Washington, DC)

Healthcare is currently transitioning from prioritizing survival to prioritizing patient care with the expectation of survival. In response, current research intends to explore and ultimately identify an optimal hospital environment. Intensive care units are often susceptible to noisier environments resulting from the requirements of urgent care. Administrative interventions such as Quiet Time are a strategy to reduce noise levels without sacrificing patient care. Despite gaining popularity, there is limited published research that rigorously evaluates the effectiveness of Quiet Time from both acoustical and medical perspectives. The presented work includes a longitudinal study of Quiet Time in multiple neonatal intensive care units. Both acoustical and patient physiological measures were taken with the intention of (1) characterizing the relationship between Quiet Time and the measured soundscape, (2) investigating potential relationships between Quiet Time and infant health, and (3) exploring potential relationships between the measured soundscape and infant health. Results including detailed acoustical analysis of traditional and newly developed metrics and statistical models relating both Quiet Time and soundscape to patient physiological response will be presented. Taken as a whole, the research provides insight into the effectiveness of Quiet Time interventions and the relationships between hospital noise and infant health.

9:25

1aNS2. Subjective and objective assessments of pediatric and neonatal hospital soundscapes. Yoshimi Hasegawa and Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska Lincoln, 6185 Walnut St., Omaha, NE 68106-2125, yhasegawa@unomaha.edu)

Existing literature reveals that overall noise levels in pediatric and neonatal hospital units are exceeding acceptable ranges and exhibiting persistent noise level fluctuations over time and location. Previous studies have also revealed insights into the subjective perception of noise and explored existing noise sources and their measured acoustical properties. However, less has been done regarding clear identification of hospital noise sources related to staff annoyance in addition to detailed, informative representations of the acoustical characteristics of hospital environments. This study utilizes two methods of unsupervised learning techniques— factor analysis and clustering analysis— to assess occupant perception alongside detailed noise level measurements. Data were collected in two pediatric and neonatal hospital units to provide informative representations of the existing acoustical environments. The factor analysis results show three inherent noise categories among the various noise sources in the hospitals. The subsequent multiple linear regression demonstrates potential negative impacts of those noise categories on occupants' psychological responses. The clustering analysis results show some relevant parameters for classifying the noise levels into a number of distinct conditions related to the activity level. This study presents new methods for screening hospital acoustical environments and allows us to gain insight into problematic hospital noise and corresponding perception.

9:45

1aNS3. Hospital design features related to patient experience. Kenton Hummel, Erica E. Ryherd, and Jay Bliefnick (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, hummelkenton@gmail.com)

Noise can be problematic in hospitals due to concerns over impacts on patients and staff members. Noise in particular is problematic and consistently rated low on patient experience surveys nationwide. Furthermore, previous studies have linked hospital noise to negative reactions for patients including reduced sleep and increased incidence of re-hospitalization. These safety risks—coupled with reduced Hospital Value-Based Purchasing program reimbursements due to low patient experience ratings—result in strong incentives for healthcare providers to provide quieter, safer environments. In this talk, we will explore what is known about hospital design features and patient experience, including a particular focus on noise. Results from ongoing and previous studies reveal new insights into how various features of the built environment are related to patient experience and therefore should be considered in the design and renovation of healthcare facilities.

10:05

1aNS4. Evaluating patient and staff perceptions of soundscape conditions within three hospitals. Jay Bliefnick and Erica E. Ryherd (Architectural Eng. & Construction, Univ. of Nebraska, 1110 S 67th St., Omaha, NE 68182-0816, jbliefnick@gmail.com)

Hospitals can present challenging soundscapes due to the continuous activity found within these environments. Routinely, this leads to poor perceptions of acoustical conditions from both patients and staff, such as in nationally reported HCAHPS patient surveys or hospital-administered staff surveys. In fact, it has been found that patient satisfaction of in-room soundscape conditions is highly related to the overall hospital rating and that staff job performance and satisfaction can be negatively affected by the acoustical environments in which they work. This research addressed these issues by collecting acoustical measurements within 38 patient rooms from 11 units of three individual hospitals, comparing results with patient and staff survey information. Data collected included 24-h occupied sound monitoring of patient rooms and nursing stations, unoccupied BNL sound levels from 20 patient rooms, and impulse response measurements of hospital unit hallways. Results from statistical analyses between measured acoustical data and patient/staff surveys will be reported, along with lessons learned applicable across all three hospitals. Taken as a whole, this study provided new insights into patient and staff perceptions of hospital noise, and could ultimately aid in the design process of new hospitals to improve patient and staff satisfaction.

10:25–10:40 Break

Contributed Papers

10:40

1aNS5. Impacts of noise on staff cognitive performance in a hospital emergency department. Khaleela Zaman, Peter Dodds, Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, khaleela25@yahoo.com), and Paul Barach (Wayne State Univ. School of Medicine, Lincoln, MA)

The soundscape of a typical modern hospital emergency department is undoubtedly noisy. Noise-related stress can contribute to human error, adverse medication events, and physician burnout and may negatively affect physicians' mental health and limit clinicians' ability to provide high-quality patient care. Previous studies have revealed average sound pressure levels for hospitals worldwide to be in significant excess of the World Health Organization firm guidelines. However, neither sound pressure level measurements nor loudness evaluations provide enough insight into the problem or potential solutions. Noise sources in hospitals include monitor alarms, overhead paging, echogenic surfaces, trash bins, ring binders, and patient crying out, among others. These sounds can be abrupt, yet not sustained. In order to evaluate the impact of hospital staff distraction and propensity for human error due to noise, the authors have conducted research at a busy, urban emergency department. The effects of various sonic occurrences on the cognitive load and working memory of physicians operating in the emergency department were assessed using binaural augmented acoustic environments as the backdrop for cognitive executive function evaluations. This paper discusses the methods for the binaural augmentation, cognitive testing, and initial results and offers interpretation and potential solutions to address these results.

10:55

1aNS6. Acoustic challenges of senior care facilities. Emily Schilb and Edward Dugger (Edward Dugger + Assoc., 1239 SE Indian St., Ste. 103, Stuart, FL 34997, emily@edplusa.com)

The population of the US, Europe, and Asia is aging, and the rate at which individuals are developing neurodegenerative diseases is also increasing. As a result, we are seeing a rise in the number of senior living facilities, ranging from independent living for active seniors to memory care facilities aiding those suffering from Alzheimer's and Dementia. Often these mixed levels of care are housed in one building, with the result being a dynamic facility with uses not limited to residential units and medical clinics but including fitness centers, pools, theaters, auditoriums, recreation rooms, dining rooms, and bars. With such a diverse range of often acoustically incompatible activities and uses, the challenges in designing these facilities quickly become apparent. During this presentation, we will explore the challenges of integrating residential suites with various amenities and the potential solutions for maintaining acoustic isolation without compromising architectural design or functionality.

11:10

1aNS7. Soundscape in dementia care environment. Arezoo Talebzadeh (Design for Health, OCAD Univ., 8 The Esplanade, Unit 2403, Toronto, ON M5E 0A6, Canada, arezoo.talebzadeh@gmail.com), Ramin Behboudi (Professional Engineers of ON, Toronto, ON, Canada), and Andrea Iaboni (Toronto Rehabilitation Inst., Univ. Health Network, Toronto, ON, Canada)

Noise is an important sensory stimulus in any environment, especially in unfamiliar settings. Noise is impossible to ignore, and any disturbing noise or constant sound can be agitating, disturbing, and confusing for people who cannot escape the environment. People with dementia may already feel disoriented, isolated, and confined inside care facilities; uncontrolled sound can add to their anxiety and distress. Soundscape refers to the human perception of the auditory environment in context; it relies not only on the subjective quality of sound by quantifying the sound level but also the objective quality of the auditory environment based on people's perception. The aim of this study is to describe the soundscape of the Specialized Dementia Unit at the Toronto Rehabilitation Institute, through data collection and observation, and to evaluate the quality of soundscape. Results show that the overall sound level (dB) of the unit is higher than recommendations, and also, the observation study shows that higher sound level not necessarily results in negative atmosphere, such as chaos or agitation. The findings prove a need for further study on the relation between the sound level and the perception of sound in dementia care units, which can foster improvement in quality of life for residents and staff.

11:25

1aNS8. Experimental study on effect of background noise on deep sleep in bedroom. Xiang Yan (Tsinghua Univ., Rm. 104, Main Academic Bldg., Beijing 100084, China, xiang.yan72@yahoo.com), Jianghua Wang, Hui Li (Beijing Deshang Jingjie Technol. Development Co. Ltd., Beijing, China), and Yuxiao Chen (Tsinghua Univ., Beijing, China)

One of the most important external factors for sleep quality is noise. Previous studies show that deep sleep is disturbed strongly by noise, resulting in insufficient cerebral cortex deep resting, in delaying the growth and development, and in reducing immunity and brain functions. In the past, all the research studies added artificial noise into the bedroom, and how background noise affects sleep was still unknown. In this paper, the difference in the deep sleep length between the normal bedroom and the 0-dB(A) silence room which excludes the background noise during subject's sleep was compared. Continuously two night sleeps of 35 random subjects, wearing the EEG brainwave Zeo headband, were recorded, one in the silence room and another in subject's self-home bedroom. The result shows that comparing 29–35 dBA with the 0 dB extreme silence condition, deep sleep length increased. The best value locates about 31 dBA, which may be related to the masking of breathing sounds. Below 31 dBA, with noise reducing, the deep sleep length is reduced because of the excessive quietness reduces the masking effect, highlighting the effects of incidental noise. Between 31 and 48 dBA, as the noise increases, the length of deep sleep would decrease, by 4% for each additional 1 dB noise.

11:40

1aNS9. Does noise sensitivity or attentional capacity predict cardiovascular responses to distracting sound? Jordan N. Oliver (Purdue Univ., West Lafayette, IN), Lea Sollmann, Annika Niehl (Systems Neurosci. & NeuroTechnol. Unit, HTW Saar, Saarbruecken, Germany), and Alexander L. Francis (Purdue Univ., SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu)

Workplace noise may cause stress that has long-term implications for health. Employees in open-plan offices typically identify intermittently occurring background sounds as significant sources of stress, possibly because they distract attention away from intended tasks. Thus, individual differences in attention and noise sensitivity may explain individual differences in physiological responses to workplace noise. In previous research [Oliver, *et al.*, in 176th Meeting of the Acoustical Society of America (2018)], we reported results from a preliminary analysis of 19 participants' affective and electrodermal responses to working in background noise that included intermittent environmental sounds. Electrodermal responses were used as a measure of distraction because they constitute a primary component of the physiological orienting response, a multi-component response reflecting exogenous capture of attention. Our results suggested that individuals who are more sensitive to noise experienced greater frustration with the primary task, but there was no relationship between frustration and distraction as indexed by electrodermal responses. Here, we present additional analyses of an expanded dataset, focusing on attentional and cardiovascular measures (heart rate and pulse volume amplitude) previously linked to cognitive demand and noise annoyance (Francis *et al.*, 2016) and relate these results to an overall characterization of the auditory orienting response.

Session 1aPA

Physical Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, Noise, Psychological and Physiological Acoustics, and Speech Communication: Battlefield Acoustics I

W. C. Kirkpatrick Alberts, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20723

Gregory W. Lyons, Cochair
Construction Engineering Research Laboratory, U.S. Army Engineer Research and Development Center, 2902 Newmark Dr., Champaign, IL 61822

Chair's Introduction—8:40

Invited Papers

8:45

1aPA1. Artillery location: Battlefield acoustics in the First World War. R. D. Costley (U.S. Army Engineer Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, casa.costley@gmail.com)

Although acoustics has played a role in warfare for hundreds, if not thousands, of years, the First World War came after the technological advances of the late 19th and early 20th centuries that enabled more quantifiable and less subjective observations. In 1914, after the outbreak of the war, Frenchman Charles Nordmann, a Professor of Astronomy at the Paris Observatory at Meudon, conceived and developed systems for locating artillery by measuring the differences in the times of arrival of the sound from the artillery to different observation positions. In one system, humans made "subjective" observations; alternatively, an "objective" system recorded these sounds with carbon microphones. In July 1915, Colonel Coote Hedley, head of the Geographical Section of the General Staff in London, learned of Nordmann's work on a visit to France and recruited Second Lieutenant William Lawrence Bragg to evaluate it. Bragg led the allied effort to improve and deploy sound-ranging systems. In developing the system, they encountered problems in sensor design, array design, wind noise, and outdoor sound propagation. The technical achievements gained and obstacles encountered will be described, along with the impact they made to the war effort. Permission to publish was granted by Director, Geotechnical and Structures Laboratory.

9:05

1aPA2. Ambient infrasound noise in urban environments. Sarah McComas (U.S. Army Res. and Development Ctr., 3909 Halls Ferry Rd. ATTN: CEERD-GS-S, Vicksburg, MS 39180, sarah.mccomas@usace.army.mil), Chris Hayward (Roy M. Huffington Dept. of Earth Sci., Southern Methodist Univ., Dallas, TX), Christopher Simpson (U.S. Army Res. and Development Ctr., Vicksburg, MS), Brian W. Stump (Roy M. Huffington Dept. of Earth Sci., Southern Methodist Univ., Dallas, TX), and Mihan H. McKenna (U.S. Army Res. and Development Ctr., Vicksburg, MS)

Associated with the emerging use of infrasound at tactical ranges (less than 50 km in complicated terrain), there is an increased need to deploy infrasound arrays in urban environments near sources of interest in order to record small signals. Initial urban deployment research demonstrated successful monitoring of a single source within 1 km utilizing rooftop infrasound arrays. In order to understand the applicability of this technique for a broader range of sources, the impact of the urban array design on noise reduction and detection of signals of interest must be assessed. These source signals, recorded within an urban scenario, are embedded in a complicated ambient noise environment with multiple coherent clutter sources and an overall decrease in the signal-to-noise ratio. Understanding these ambient acoustic field characteristics requires deconvolving observed coherent signals from time-varying atmospheric effects for different levels of urbanization. This paper presents an initial approach to instrumenting the urban environment with consideration of wind filter effects and data analysis to estimate total ambient acoustic field for a subset of urban zones. This work then provides a definition of urbanization that can be related to the infrasound wavefield. Permission to publish was granted by the Director, Geotechnical and Structures Laboratory.

9:25

1aPA3. Integrating acoustic shooter detection into a hearing protection device. Sébastien Hengy, Marcus Christoph, Pascal Duffner, Pascal Hamery, and Veronique Zimpfer (French-German Res. Inst. of Saint-Louis (ISL), 5 rue du Général Cassagnou - BP 70034, Saint-Louis 68300, France, sebastien.hengy@isl.eu)

Recent advances in the development of hearing protection devices open new fields of applications on the battlefield. While the TCAPS (Tactical Communication And Protective Systems) protect against acute acoustic traumas, they maintain information relative to the acoustic environment of the soldiers. Today, these systems are efficient for both hearing protection and communication. We propose to use the microphones equipping the TCAPS headsets in order to detect and localize shooters on the battlefield. The microphone underneath the hearing protection is used in order to detect the shock and muzzle waves generated by supersonic shots. A meshed network between the TCAPS deployed on the field allows transmitting asynchronous information relative to the detected waves to data fusion nodes that allow estimating the shooter's position. Solutions are proposed in order to compensate the effects of the presence of the head between the microphones underneath the hearing protection. Results concerning the estimation of the time difference of arrival of a transient wave in free field and in the presence of an artificial head are presented. The data fusion process is tested thanks to simulations in various deployment configurations.

9:45

1aPA4. Characterization of digital MEMS microphone elements for usage in a hostile fire detection system on a multi-rotor drone. Wayne E. Prather, William G. Frazier, and Xiao Di (NCPA, Univ. of MS, 145 Hill Dr., P.O. Box 1848, MS 38677-1848, wayne@olemiss.edu)

Acoustic based Hostile Fire Detection Systems (HFDS) depend on multi-element arrays of microphones which are simultaneously sampled at sample rates often in the 10–50 ksps range. A driving component of the electronic hardware complexity of the system is the signal buffering, filtering, and analog to digital conversion required for each microphone channel. Digital MEMS microphones consolidate these functions to within the microphone element giving a digital output representative of the measured acoustic signal. Digital MEMS microphones allow for a completely digital hardware solution which reduces complexity and allows more flexibility in utilizing higher element number arrays if desired. The performance parameters relevant to HFDSs of current best-in-class digital MEMS microphones were measured and compared to current analog microphones commonly utilized in these systems. The purpose of these tests was to determine the viability of digital MEMS microphones as an alternate acoustic sensing element for a HFDS system being implemented on a multi-rotor drone.

10:05–10:20 Break

Contributed Papers

10:20

1aPA5. Overview of acoustic detection for counter unmanned aircraft systems. Brian B. Bishop and Justin Tufariello (The MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, bishopb@mitre.org)

Integration of unmanned aircraft systems (UAS) into the National Airspace System (NAS) is currently a major problem for government and industry. The detection of unauthorized UAS activity is a concern. There is a need for effective low-cost sensor technology to detect and track unauthorized UAS operations. Acoustic sensor systems have potential to improve counter UAS (C-UAS) sensing capabilities at greatly reduced cost relative to traditional sensors such as radar and optical systems, but the commercially available acoustic sensing technology is still lacking. The MITRE Corporation has been a leader in advising the United States' Federal Government on the issues around integration of UAS into the NAS and C-UAS technologies. Research done by MITRE on acoustic sensing of UAS will be presented. The current state of commercially available acoustic sensor systems will be briefly reviewed. Recommendations that MITRE has developed on how to design acoustic sensor systems for C-UAS will be discussed. Future research directions in topics such as acoustic beamforming, array design, and outdoor sound propagation will be recommended. Approved for Public Release; Distribution Unlimited 18-4035

10:35

1aPA6. Experimental studies of sound propagation over vertical and slanted paths in a turbulent atmosphere. Vladimir E. Ostashev, D. Keith Wilson, Matthew Kamrath, Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), Michael J. White, Michelle E. Swearingen, and Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Sound propagation over vertical and slanted paths differs from horizontal paths due to the height dependence of turbulence parameters and atmospheric stratification. Vertical and slanted paths are important for several applications, such as localization of unattended aerial systems (UASs) with ground-based acoustic microphone arrays, detection of ground-based sources with elevated arrays, and the effect of atmospheric turbulence on sonic booms. This paper describes a comprehensive experiment on near-vertical sound propagation conducted over five days in September 2018 at the National Wind Technology Center (NWTC), located near Boulder, CO. The experiment involved the NWTC 135-m meteorological tower with meteorological instruments installed at 17 heights. Nine microphones were located on three horizontal booms attached to the tower at different heights, and a speaker was placed on the ground. Two reference microphones where located near the speaker. About 65% of the time, the speaker transmitted nine or twelve tones. During the remaining time, 6-ms chirps were transmitted. A preliminary analysis of the signal statistics is presented and compared with theoretical predictions.

10:50

1aPA7. Broadband acoustic vector sensing in a forested environment.
Sandra L. Collier, Max Denis, John Noble, Latasha Solomon, David Ligon, Deryck D. James, W. C. Kirkpatrick Alberts, Christian G. Reiff, and Leng Sim (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil)

The horizontal acoustic particle velocity and acoustic pressure, concurrent with atmospheric data, were collected during a series of outdoor field tests. Here, we present the effects of a forested environment on acoustic vector sensing for broadband sources. The sources are representative of battlefield signals of interest. The complex distributions are studied in the unsaturated, partially saturated, and fully saturated regimes. Distinct path-dependent, frequency-dependent propagation effects are observed.

11:05

1aPA8. Beamforming and range-migration localization algorithms using the retrieved Green's function.
Max Denis (U.S. Army Res. Lab., 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com), Sandra L. Collier, John Noble, W. C. Kirkpatrick Alberts, David Ligon, Leng Sim, Deryck D. James, and Christian G. Reiff (U.S. Army Res. Lab., Adelphi, MD)

In this work, the localization of acoustic noise sources and scatterers in an outdoor environment using Green's function retrieval methods is presented. Of particular interest is the implementation of cross-correlation and multidimensional deconvolution into localization algorithms. Conventional beamforming and range-migration algorithms, implemented with Green's function retrieval methods, are compared as localization techniques for an array system. The accuracy of both algorithms to reconstruct the position of targets (noise sources and scatterers) is investigated. Additionally, the added localization enhancements using the retrieved Green's functions are discussed.

11:20

1aPA9. Developing extrapolated pressure level maps from firearms.
Reese D. Rasband, Kent L. Gee (BYU Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, r.rasband18@gmail.com), Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH), Caleb M. Wagner (Human Systems Program Office, Air Force Res. Lab., Dayton, OH), and William J. Murphy (Hearing Loss Prevention Team, National Inst. for Occupational Safety and Health, Cincinnati, OH)

This paper describes the development of spatial maps for both peak and A-weighted equivalent sound levels from various firearms, as measured with different spatial resolutions. Two outdoor datasets that adhered to the MIL-STD-1474E weapon noise measurement standard are considered. First is an extensive measurement of the M16A4 rifle at U.S. Marine Corps Base Quantico that included several shooter configurations and a circular measurement arc with 15-deg spacing and 3.67 m radius [R. D. Rasband *et al.*, *J. Acoust. Soc. Am.* **143**, 1935 (2018)]. Second is a measurement of many different firearms using an array with 30-deg spacing and 3.0 m radius, performed in Rudyard Michigan [W.J. Murphy *et al.*, *J. Acoust. Soc. Am.* **132**, 1905 (2012)]. Using these levels and conservative estimates based on spherical spreading and symmetry, level maps are created based on shooter position, weapon used, and number of shots fired. The fidelity of the extrapolation procedure and possible future improvements is discussed. [Work supported by ONR, WTB Quantico, and AFRL through ORISE.]

Session 1aPP

Psychological and Physiological Acoustics: Physiology and Modeling

Anahita H. Mehta, Cochair
University of Minnesota, N640, Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455

Axel Ahrens, Cochair
Department of Electrical Engineering, Hearing Systems group, Technical University of Denmark, Ørsteds Plads, Building 352, Kgs. Lyngby 2800, Denmark

Contributed Papers

10:10

1aPP1. The role of cue enhancement and frequency fine-tuning in hearing impaired phone recognition. Ali Abavisani and Jont Allen (ECE, Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Rm. 2137, Urbana, IL 61801, aliabavi@illinois.edu)

A speech-based hearing test is designed to identify the susceptible error-prone phones for individual hearing impaired (HI) ear. Only robust tokens in the experiments' noise levels had been chosen for the test. The noise-robustness of tokens is measured as SNR_{90}s of the token, which is the signal to the speech-weighted noise ratio where an NH listener would recognize the token with an accuracy of 90% on average. Two sets of tokens T_1 and T_2 having the same consonant-vowels but different talkers with distinct SNR_{90}s had been presented at flat gain at listeners' most comfortable level. We studied the effects of frequency fine-tuning of the primary cue by presenting tokens of the same consonant but different vowels with similar SNR_{90}s. Additionally, we investigated the role of changing the intensity of primary cue in HI phone recognition, by presenting tokens from both sets T_1 and T_2. On average, 85% of tokens are improved when we replaced the CV with the same CV but with a more robust talker. Additionally, using CVs with similar SNR_{90}, on average, tokens are improved by 28%, 28%, 25%, and 19%, when we replaced the vowel with ɑ, ɛ, ə, and ɪ. The confusion pattern in each case provides insight into how these changes affect the phone recognition in each HI ear. We propose to prescribe hearing aid amplification tailored to individual HI ears, based on the confusion pattern, the response from cue enhancement, and the response from frequency fine-tuning of the cue.

10:25

1aPP2. Cortical effects on spatial tuning to speech in background babble. Erol J. Ozmeral, Katherine Palandrani, David A. Eddins, and Ann C. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210, Tampa, FL 33612, eozmeral@usf.edu)

The ability to understand speech in complex backgrounds often relies on spatial factors that contribute to forming discernible auditory objects. From stimulus-evoked onset responses in normal hearing listeners using electroencephalography (EEG), we have shown measurable spatial tuning to moving noise bursts in quiet, revealing a potential window into cortical object formation. However, it is still unknown whether comparable effects are observed with speech stimuli, and whether and how much the presence of noise disrupts EEG responses to moving speech. To test whether the presence of noise has deleterious effects on object formation and potential selective auditory attention, we measured cortical responses to moving speech in the free field with and without background babble (+6 dB SNR) during both passive and active conditions. Active conditions required listeners to respond to the onset of the speech when it occurred at a new location, while indicating yes or no to whether the stimulus occurred at a block-specific location. We discuss in detail the effect of noise and attention on spatial tuning to speech stimuli as measured by the magnitude and latencies of the N1, P2, and P3 components at primary and secondary auditory regions of interest.

10:40

1aPP3. Physiological assays of suprathreshold hearing are consistent with widespread deafferentation of the human auditory periphery. Alexandra Mai, Brooke Flesher, Kelsey Dougherty, Anna Hagedorn, Jennifer M. Simpson, Michael G. Heinz, and Hari M. Bharadwaj (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, mai10@purdue.edu)

Multiple animal models have robustly shown the effects of noise-exposure and aging can have on the afferent synapses between the cochlea and the auditory nerve. This cochlear synaptopathy can affect responses to suprathreshold stimuli while leaving audiometric thresholds intact. However, currently, there is much debate on whether these same changes occur in humans with significant noise-exposure or with middle age. Our study examined two different physiological responses in which these afferent synapses are a crucial component, the auditory brainstem response (ABR) and the middle ear muscle reflex (MEMR). Versions of these measures were completed in both a clinical setting and a laboratory. Responses to both measures in both testing environments demonstrated significant age and noise-exposure effects. Moreover, these effects remained significant even after statistically accounting for variability in audiometric sensitivity and otoacoustic emissions, suggesting that despite clinically normal audiograms, cochlear synaptopathy may be a widespread occurrence in humans with both acoustic-overexposure and normal aging. Finally, our results suggest that a battery combining ABR and MEMR measures may be viable as a non-invasive assay of synaptopathy and can help examine the perceptual sequelae of such damage.

10:55

1aPP4. Combining psychophysics and modeling to probe suprathreshold auditory processing efficiently. Emmanuel Ponsot (LSP, CNRS/ENS, 1 Pl. Igor Stravinsky, Paris 75004, France, ponsot@ircam.fr)

The ability to understand speech in noise differs considerably among people. Even "normal-hearing" individuals, i.e., with normal audiograms, exhibit a variety of performances in speech-in-noise listening tests. The number of studies investigating the underlying causes of this variability, not captured by current audiological measures, has exploded over the last few years. However, we still lack an overall computational account of suprathreshold auditory processing that would incorporate enough flexibility to capture these idiosyncratic differences. We propose to uncover the psychophysical mechanisms of suprathreshold processing trough its variability among individuals, by combining noise-based psychophysical methods and computational tools. This approach is illustrated in the case of spectrotemporal modulation processing in normal and impaired hearing. Experimental results collected show how listeners modulate their tuning characteristics depending on the task and reveal that hearing-impaired listeners with similar

audiometric losses exhibit a large variety of computational strategies for such tasks, which our modeling approach have the potential to connect with specific computational elements. Overall, this shows how a signal-detection theory framework combined with efficient experimental methods and modeling tools should be considered as a powerful approach to further understand the respective contributions of sensory coding and read-out inefficiencies in human suprathreshold processing.

11:10

1aPP5. Midbrain sensitivity to spectral cues for sound-source localization: A modeling study. Yuxiang Wang and Laurel H. Carney (Elec. Eng., Univ. of Rochester, Rochester, NY, ywang310@ur.rochester.edu)

The features of head-related transfer functions (HRTFs) are a key topic in the field of spatial auditory displays. Neural encoding of HRTFs is not well understood; their sharp spectral peaks and notches are not well represented by average rates of relatively widely tuned auditory-nerve (AN) fibers, and phase-locking to the temporal fine structure is not adequate to encode features at high-frequencies. However, low-frequency fluctuations in cochlear responses to wideband stimuli are encoded in time-varying rates of AN fibers across all frequency channels. Here, we focus on modeling the profile of amplitude fluctuations in response to spectral cues across the population of AN channels. The fluctuation profile across AN responses sets up an average-rate profile across inferior colliculus (IC) neurons, which are sensitive to envelope-related low-frequency fluctuations. Both unilateral and bilateral IC model responses indicate that rather than responding to spectral peaks in a given HRTF, IC responses are more sensitive to fluctuations in frequency channels near steep spectral slopes. The influence of the spatial location and stimulus level on model IC responses was examined. Using statistical methods, psychoacoustical thresholds of discrimination of sounds that differ in the source location were estimated and compared to trends in the existing perceptual results.

11:25

1aPP6. Comparison of the predictive accuracy of different computational models of auditory perception. Evelyn E. Davies-Venn (SLHS, Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55445, venn@umn.edu), Nursadul Mamun (Dept. of Elec. Eng., Univ. of Texas-Dallas, Dallas, TX), Md. Hossain (Faculty of Eng. & IT, The Univ. of Sydney, Sydney, NSW, Australia), Timothy Kwan (Dept. of Biomedical Eng., Univ. of Malaya, Kuala Lumpur, Malaysia), Melanie Putman (SLHS, Univ. of Minnesota, Minneapolis, MN), and M. S. A. Zilany (Dept. of Comput. Eng., Univ. of Hail, KSA, Saudi Arabia)

Computational models of auditory perception offer a time-efficient method of assessing the effects of distortion on speech perception. Several objective metrics have been proposed to predict speech intelligibility, especially when speech is obscured by the presence of background noise. Novel approaches to full-reference and reference-free speech intelligibility metrics have emerged in recent years, but deciphering the best metric for predicting speech intelligibility still requires investigation. This study assessed the predictive accuracy of several reliable, full, and reference-free speech intelligibility metrics. Speech perception scores were measured on listeners with normal hearing and hearing loss in quiet and noise. Acoustic recordings were made of the presented speech stimuli and combined with a computational model of the auditory nerve to simulate behavioral scores using several established metrics such as the STOI, NSIM, SRMR, SII, SNRloss, and BSIM. The estimated speech scores were correlated with behavioral speech recognition scores to assess predictive accuracy of the model simulations. Several of the predicted scores correlated well with behavioral scores. Evaluation of individual phonemes revealed differential sensitivity of the metrics across different phonemic classifications.

11:40

1aPP7. Neural correlates of auditory enhancement. Anahita H. Mehta and Andrew J. Oxenham (Univ. of Minnesota, N640, Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, mehta@umn.edu)

Auditory enhancement is the increase in salience of a target embedded in a simultaneous masker that occurs when a copy of the masker, termed the precursor, is presented first. The effect reflects the general principle of contrast enhancement and may help in the perceptual constancy of speech under varying acoustic conditions. The physiological mechanisms underlying auditory enhancement remain unknown. This study investigated EEG responses under conditions that elicited perceptual enhancement. The target tone was amplitude-modulated at two modulation frequencies to target cortical (~40 Hz) and subcortical (100–200 Hz) responses. Measurements were made in either passive conditions or under active tasks to examine the potential effects of attention on the neural correlates of enhancement. Robust effects of enhancement were observed at the cortical level, replicating our earlier findings. Preliminary data under passive conditions also suggest a trend towards increased neural response to the enhanced target tone at frequencies exceeding 200 Hz, suggesting a subcortical contribution. The results suggest that this paradigm can be used to tap into the neural correlates of auditory enhancement at both cortical and subcortical levels simultaneously and show the potential for tapping into attentional modulation of auditory enhancement. [Work supported by NIH grant R01DC012262.]

Session 1aSAa

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Architectural Acoustics: Smart Materials for Acoustics and Vibration I

Kathryn H. Matlack, Cochair
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Bogdan Ioan Popa, Cochair
Mechanical Engineering, University of Michigan, University of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Invited Papers

8:00

1aSAa1. Design and additive manufacturing of electro-mechanical metamaterials with designed anisotropy and self-sensing. Xiaoyu (Rayne) Zheng (Mech. Eng., Virginia Tech, 635 Prices Fork Rd., 445 Goodwin Hall, Blacksburg, VA 24061, raynexzheng@vt.edu)

In this talk, I will present our research on design, additive manufacturing, and performances of new classes of multi-functional materials that transcend the common electro-mechanical coupling limitations. These materials are as light as carbon aerogels, but with orders of magnitude higher stiffness and strength, they possess multi-functionalities. They are composed of interconnected 3D hierarchical micro-structures as designed "atoms" and "molecules" as in natural materials to reach uncharted white space in the material selection charts. Attention is focused on our development of a suite of novel additive manufacturing and processing techniques to synthesis traditionally unprocessable inorganic/organic highly responsive building blocks and architect them into scalable form factors with precisely defined active three-dimensional micro- and nano-scale features. We will discuss the possiblities of coupling and decoupling of the piezoelectric coefficients, density stiffness scaling, and selective electro-mechanical amplifications through our metamaterial design and printing approach. These insights shed light on a next generation of metamaterials, with designed-in structural and smart functionalities, including self-sensing and actuation, vector sensing and detection as well as simultaneous impact wave absorptions and self-monitoring with only a fraction of solid, as opposed to relying on multiple components.

8:20

1aSAa2. Dynamic reconfiguration of two-dimensional lattices with bistable springs. Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, julien.meaud@me.gatech.edu)

This study focuses on wave propagation and reconfiguration in two-dimensional square mass-spring lattices that include bistable springs. Due to the presence of directional and/or omnidirectional bandgaps in stable deformed configurations in which some of the bistable springs are in a deformed equilibrium, these lattices have the ability to serve as reconfigurable wave filters and wave guides. In this work, we study non-linear wave propagation in response to stimulus of large amplitude. Mechanical stimuli of large amplitudes have the capability of causing a dynamic reconfiguration of the lattice to a configuration of lower potential energy, which dramatically affect wave propagation. Using numerical simulations, the influence of the stimulus amplitude, duration, and location of the applied stimulus on the reconfiguration is analyzed. The ability to alter in a predictable manner wave propagation in these lattices offers new opportunities for tunable phononic crystals.

8:40

1aSAa3. Optimal vibration suppression in adaptable acoustic metamaterials by a complex wavenumber. Aaron J. Stearns (Mech. Eng., Appl. Res. Lab., Penn State Univ., P.O. Box 30 Burrowes St., State College, PA 16804-0030, ajs6037@psu.edu) and Benjamin Beck (Acoust., Appl. Res. Lab., Penn State Univ., State College, PA)

Acoustic metamaterials are composite materials exhibiting effective properties and acoustic behavior not found in traditional materials. Through periodic subwavelength resonant inclusions, acoustic metamaterials enable steering, cloaking, lensing, and frequency band control of acoustic waves. A common drawback of acoustic metamaterials is that the properties are limited to narrow frequency bands. Investigation of practical active and adaptable acoustic metamaterials is valuable in achieving wider operation frequency bands. Here, a one dimensional metamaterial is analyzed using a finite element method. The purpose of the metamaterial is to minimize vibration. From the finite element method formulation, a complex wavenumber is calculated. The unit cell considered is active and adaptable via piezoelectric actuators attached to negative capacitance shunt circuits. Therefore, the stiffness of the unit cell is tunable through selection of the circuit parameters. Choosing the negative capacitance shunt circuit elements to maximize the attenuative part of the complex wavenumber leads to maximum vibration suppression. So far, it has been found that maximizing the attenuative part of the wavenumber gives unrealizable specifications for the shunt circuit. To prevent this, the goal of this work is to constrain the optimization problem using experimentally obtained circuit stability bounds. The experimental stability results and optimization results will be presented.

9:00–9:15 Break

Contributed Papers

9:15

1aSAa4. Reconfigurable valley acoustic phononic crystals based on a fluidic system. Chen Shen (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC 27708, chen.shen4@duke.edu), Zhenhua Tian (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Junfei Li (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC), Eric Reit, Tony Jun Huang (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), and Steven Cummer (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC)

Valley serves as a new degree of freedom in controlling wave dynamics. Here, we present a design of valley acoustic phononic crystals (PCs) composed of a hybrid channel-cavity structure. Valley states for both waveguide and surface acoustic modes can be realized, and the mode transition is enabled by adjusting the channel height. Reconfigurable valley Hall phase transition in a wide range of frequencies is allowed by tuning the cavity sizes based on a fluidic system. By injecting/withdrawing fluid into/out of the cavities, the dispersion relation, phase transition, and edge states can be controlled conveniently in the two-dimensional PCs. Frequency-dependent acoustic routing, tunable refraction, topological switching, wave splitting, and reconfigurable acoustic pathways with suppressed backscattering are demonstrated both numerically and experimentally through acoustic field scanning. The reconfigurable valley PCs can serve as a versatile platform for exploring valley-related physics and achieving tunable wideband acoustic devices.

9:30

1aSAa5. Anomalous wave polarization through a 3D periodic auxetic lattice. Ganesh U. Patil and Kathryn H. Matlack (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801, gupatil2@illinois.edu)

Solid media supports both longitudinal and shear wave polarizations, providing a rich platform for designing phononic materials with prescribed wave filtering, engineered mode conversions, negative refraction, and other unique properties. While longitudinal waves almost always propagate at a faster velocity than shear waves in natural materials, tailoring the polarization of the faster wave velocity could enable unique control over wave propagation properties. Here, we present a three-dimensional periodic "bowtie" lattice that exhibits a shift in the faster wave polarization from the quasi-longitudinal to quasi-transverse wave in an anisotropic plane. We observe that this shift, termed "anomalous wave polarization", is possible when the lattice behaves auxetically. Using the finite element method, we show that the wave polarizations in the bowtie lattice depend on certain geometric parameters and the wave propagation direction. We use wave velocity information to evaluate the lattice effective properties using the Bloch-wave homogenization approach and confirm the required elastic condition for the polarization anomaly in an anisotropic plane. We also emphasize the importance of identifying the wave polarization along with the wave velocity classification for lattice effective property evaluation. This lattice behavior will be useful for designing mode-converting metamaterials that have potential application in non-destructive testing and bio-medical ultrasounds.

Session 1aSAb

Structural Acoustics and Vibration and Architectural Acoustics: Vibration Reduction for Extraordinarily Sensitive Applications

Mohammad Afrough, Cochair
Mei Wu Acoustics, 329 Estrella Way, San Mateo, CA 94403

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

Chair's Introduction—9:10

Invited Papers

9:15

1aSAb1. Ultra-low frequency vertical vibration isolators for absolute gravimeters. Kang Wu, Jiamin Yao, Guan Wang, Gang Li, and Lijun Wang (Precision Instruments, Tsinghua Univ., Bldg. 9003, Qinghuayuan 1, Haidian District, Beijing 100084, China, kangwu@mail.tsinghua.edu.cn)

High-precision absolute determinations of gravitational acceleration g provide important data for many fields such as metrology, geophysics, and geological exploration. In absolute gravimetry, vibrational noise from seismic and other environmental disturbances is one of the limiting factors. Several types of ultra-low vertical vibration isolators have been developed in Tsinghua University. The first one is a passive isolator based on LaCoste spring linkage and can achieve a natural period up to 32 s. The second one is an active isolator employing a two-stage beam structure. The upper beam is suspended from the frame with a hex spring, and the lower beam is suspended from the upper one using a zero-length spring. A feedback circuit is equipped to keep the angle between the two beams at a fixed value. The isolator can achieve a natural period of 100 s. The last one is an active isolator based on a two-stage structure, in which geometric anti-springs are used to support the proof mass. The volume of the isolator is greatly decreased, and the allowable load is increased while maintaining a natural period more than 15 s.

9:40

1aSAb2. Listening to the songs of the universe: How vibration control for the laser interferometer gravitational-wave observatory (LIGO) allows us to measure ripples in the fabric of space. Brian Lantz (Ginzton Lab, Stanford Univ., Spilker Bldg., 348 Via Pueblo Mall, Stanford, CA 94305, BLantz@stanford.edu)

In 2015, humanity made the first detection of gravitational waves from the violent collision of two black holes. As Einstein predicted, this collision sent waves through the fabric of space-time, but nearly 100 years passed between Einstein's prediction and the first measurement of these waves by the advanced LIGO detectors. The detection was made possible by many advances in the precision measurement. I will describe the detectors and one of the key technologies, the vibration isolation for the optics; at 10 Hz, the motion of the LIGO mirrors is at least 1,000,000,000 times less than the motion of the ground. By creating one of the quietest places on Earth, we have created a new way to listen to the stars.

10:05

1aSAb3. An overview of isolation controls at LIGO. Arnaud Pele (Caltech, 19100 LIGO Ln., Livingston, LA 70754, apele@ligo-la.caltech.edu)

The Laser Interferometer Gravitational-Wave Observatory (LIGO) is a large-scale physics experiment that aims at measuring gravitational waves emitted by astrophysical sources. The detection of black hole mergers and neutron star collision was made possible by the extreme level of isolation required to hold the optics still from external ground disturbances in a large band of the spectrum. From the low-frequencies (earthquakes, wind, microseism, below 1 Hz) to the higher frequencies (anthropogenic noise, above 1 Hz), the controls system must be tuned to meet the requirements for lock acquisition, lock stability, and sensitivity of the instrument. In this talk, I will describe the overall control scheme of the LIGO isolation platforms and mirror suspensions, and the challenges met to design the many feedback and feedforward control loops.

10:30–10:45 Break

10:45

1aSAb4. Seismic isolation in advanced Virgo gravitational wave detector. Valerio Boschi (European Gravitational Observatory, via Amaldi, Cascina (PI) 56021, Italy, valerio.boschi@ligo.org)

We will present an overview of the seismic isolation systems used in an AdVirgo gravitational wave interferometer. We will concentrate on the the so-called super-attenuator, the seismic isolator used for all the detector main optical components. This complex mechanical device is able to provide more than 12 orders of magnitude of attenuation above a few Hz. We will also describe its high-performance digital control system and the control algorithms implemented with it.

11:10

1aSAb5. Vibration reduction for gravitational wave detectors. Carlos Frajuca (Mech., IFSP, Rua Medina, 82, Carapicuiba, SP 06355140, Brazil, frajuca@gmail.com), Fabio D. Bortoli (Mech., IFSP, Sao Paulo, SP, Brazil), and Nadja S. Magalhaes (Phys., Unifesp, Carapicuiba, Brazil)

ultimately vibration detectors. They transform the vibrations in space-time into electrical vibrations in the electronic components of the detector. In this work, it is shown that how undesirable vibration is reduced in some of these detectors, even when it is necessary, in order to increase the general sensitivity, to incorporate in such detectors devices that vibrate alone and may compromise the overall performance.

MONDAY MORNING, 13 MAY 2019 BECKHAM, 9:00 A.M. TO 11:35 A.M.

Session 1aSP

Signal Processing in Acoustics, Engineering Acoustics, Physical Acoustics, Underwater Acoustics, and Structural Acoustics and Vibration: Reconfigurable Arrays for Adaptive Wave Guiding

Ryan L. Harne, Cochair
Mechanical and Aerospace Engineering, The Ohio State University, 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210

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

Invited Papers

9:00

1aSP1. Topology optimization of origami-inspired reconfigurable frequency selective surfaces. Kazuko Fuchi (Univ. of Dayton Res. Inst., 300 College Park, Dayton, OH 45469, kfuchi1@udayton.edu), Andrew Gillman (UES, Inc., Beavercreek, OH), Philip Buskohl (Mater. and Manufacturing Directorate, Air Force Res. Lab., WPAFB, OH), and Alexander Pankonien (Aerosp. Systems Directorate, Air Force Res. Lab., WPAFB, OH)

Fold-driven reconfigurable devices have a potential to expand functional spaces beyond traditional adaptive wave propagation strategies. In particular, designs inspired by the art of origami leverage the mathematics of origami to map designs defined on two-dimensional surfaces to complex three-dimensional shapes. The design space in this paradigm is vast, so a systematic method is needed to design a device that achieves its goal. In our initial effort, we surveyed electromagnetic wave propagation properties of foldable frequency selective surfaces (FSS) and foldable and deployable antennas based on various known origami designs to identify a number of working principles of functional tuning. We incorporated these findings in the implementation of a design method that finds an origami FSS pattern that achieves the desired frequency tuning. This method is adopted from density-based topology optimization, with a notion that anything functional could be described through a distribution of an effective density of the relevant material property. A substrate is "patterned" with foldable segments parameterized through torsional springs; electromagnetically relevant conductive patterns are described as predefined surfaces that remain unchanged. This talk will discuss the lessons learned from our investigations and remaining challenges of designing fold-driven reconfigurable devices for wave propagation control.

9:20

1aSP2. Partial activation of modular and foldable tessellated acoustic arrays for wave focusing. Ningxiner Zhao and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, zhao.2684@osu.edu)

Tessellated acoustic arrays inspired by origami structures are suggested to enable wave focusing by exploiting curvatures realized by folded configurations of the array transducer elements. The use of origami-inspired folding patterns also cultivates great portability for space-limited applications. Yet, maintaining curvatures may prohibit feasible implementation of a tessellated been studied for the acoustic energy focusing usage. This research proposes an alternative technique to achieve wave focusing with tessellated arrays that do not realize curvatures upon folding. Here, the partial activation of a tessellated array is exploited to result in constructive interference that realizes a nearfield focal region. The modeling approach to examine partially activated arrays is presented and verified against numerical simulations. Then, the changes in nearfield focusing characteristics are correlated with corresponding changes in partial activation and array folding extent. The opportunities to exploit the partial activation of relatively simple origami-inspired array structures are examined to identify strategies to simplify implementation of portable, folding acoustic arrays in applications.

9:40

1aSP3. A performance metric for screen selection with the acoustic single pixel imager. Juan Ramirez (U.S. Air Force Res. Lab., 329 East 1st St., Apt 306, Dayton, OH 45402, juan.ramirez.ee@gmail.com), Jeffrey S. Rogers, and Geoffrey F. Edelmann (Naval Res. Lab., Washington, DC)

In the recent literature, an Acoustic Single-Pixel Imager has been successfully developed for source localization in a two-dimensional waveguide. Source bearing angle estimation was carried out by applying sparse recovery techniques on sensor measurements taken over dierent imaging screens. In this paper, we show that the mutual coherence of the sensing matrix can be used as a metric to predict the source localization capability of the single-pixel imaging system. In particular, our analysis focuses on the sparsity of open cells within the imaging screen and the number of imaging screens used to maximize the probability of correct detection over varying levels of source sparsity. In this work, we develop a simulation environment to demonstrate how the mutual coherence of the sensing matrix correlates with source localization performance over source sparsity, sparsity of open screen cells, and number of measurements used for sparse recovery. Our analysis shows that the leading factor in source localization performance gains is primarily from the number of imaging screens used to measure the acoustic wave-field.

10:00

1aSP4. Mechanics and dynamics of reconfigurable curved creased origami arrays. Evgueni T. Filipov and Steven R. Woodruff (Dept. of Civil and Environ. Eng., Univ. of Michigan, 2350 Hayward, 2144 GG Brown Bldg., Ann Arbor, MI 48109, filipov@umich.edu)

The principles of origami have allowed for novel deployable and reconfigurable structures with applications from micro-robotics to adaptable architecture. Most origami patterns use rigorous mathematical definitions to achieve their desired folding kinematics; however, these geometries result in discrete and segmented structures that have sharp edges. In contrast, curved creased origami uses a more arbitrary placement of folds and can thus achieve smooth surfaces. These smooth surfaces can be useful for problems in acoustics, fluid flow, electromagnetics, wave-propagation, and more. In this talk, we first explore the geometry of the curved crease origami, where continuous curvatures and elastic bending occur over the entire surface of the thin sheet. We use a simplified bar and hinge model to approximate the folding sequence and model the behaviors of the curved crease origami. The model captures stretching and shearing of the origami, bending along principle curvature directions, and bending at the prescribed curved creases. We explore the stiffness, large deformation mechanics, buckling, and dynamic behaviors of the curved crease origami. Our results show that the curved folding can enable highly tuneable properties for these novel thin-sheet arrays.

10:20–10:40 Break

10:40

1aSP5. Deployable tessellated acoustic array with a curved Miura-ori pattern for ultrasound focusing in multilayered media. Chengzhe Zou and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, zou.258@osu.edu)

High intensity focused ultrasound (HIFU) has been successfully applied to treat cancers in clinical settings. In such treatment, the ultrasonic transducer projects focused ultrasound to the diseased tissues for thermal ablation. Nevertheless, the absorption, diffusion, and reflection that occur on the propagation path of ultrasound may reduce the effectiveness of HIFU. In order to overcome this challenge, a concept of origami-inspired deployable tessellated acoustic arrays may be leveraged. Fueled by large portability, the array may be compacted for insertion to the body and guidance to the point of care where the deployed array is then used for treatment. Such a vision requires understanding how wave propagation behaviors from reconfigurable tessellated transducer arrays are tailored in a multilayer environment. Here, the curved Miura-ori tessellation is used to approximate the arc shape for focusing. An analytical modeling framework is extended to investigate the new wave propagation behaviors encountered in biological-like media. Using the analytical tool, the tessellated acoustic array is compared with the ideal arc transducer, and the results indicate that the proposed concept may be comparable with an ideal case in focusing. In addition, the deployability of the tessellated acoustic array is confirmed through experimental efforts in a controlled multilayer environment.

11:00

1aSP6. Experimental observation of valley-Hall edge states in elastic waveguides based on diatomic-graphene-like phononic crystals. Hongfei Zhu (Univ. of Notre Dame, Notre Dame, IN), Ting-Wei Liu, and Fabio Semperlotti (Purdue Univ., 177 S Russell St., West Lafayette, IN 47907, liu2041@purdue.edu)

Inspired by recent discoveries of topological phases of matter in quantum physics, there has been a rapidly growing research effort in creating their analogs in other classical wave systems, including acoustics. Achieving robust wave transmission even in the presence of disorder and defects could have a profound impact on many practical applications and devices. In this study, we report on the design and experimental validation of a fully continuous and load-bearing phononic structural waveguide capable of one-directional guided modes along the walls of topologically distinct domains. The lattice structure of the waveguide is inspired by diatomic graphene which allows realizing an elastodynamic analog of the quantum valley Hall effect (QVHE). Despite the fact that individual bulk properties are topologically trivial (i.e., associated with a zero Chern number), the dynamic behavior acquires topological significance in the neighborhood of the high symmetry points in momentum space; the so-called valleys. Our theoretical and experimental results confirm the existence of protected edge states traveling along the walls of domains having broken space inversion symmetry.

Contributed Paper

11:20

1aSP7. Continuous scan beamforming using a rotating microphone array for mapping of acoustic sources in a soundproof chamber. Abe H. Lee, Andrew White, and Parthiv Shah (ATA Eng., Inc., 13290 Evening Creek Dr. South, Ste. 250, San Diego, CA 92128, abe.lee@ata-e.com)

Continuous scan beamforming (CSBF) is a novel approach that can improve the dynamic range of a microphone array used for source localization. In the conventional beamforming approach in which spatially fixed sensors are used, the number of sensors employed determines the dynamic range of the array. Whereas, in the CSBF approach, by employing moving sensors in a prescribed motion, the effective number of sensors (so-called virtual sensors) used for the beamforming process can be greatly increased, and therefore, it can provide enhanced dynamic ranges close to the theoretical limit. In the CSBF process, for reconstruction of the time data acquired by moving sensors, stationary microphones are used for phase referencing. At ATA Engineering, Inc., a rotating, planar configuration array of 60 microphones was built and tested in a soundproof chamber with spatially distributed acoustic sources inside. The results showed that CSBF has much better performance than the conventional beamforming that employs fixed sensors; CSBF is able to discriminate tested sources that are almost 20 dB apart, which is not possible with the conventional approach. Due to the enhanced dynamic range, CSBF provides much cleaner mapping of the distribution of sources without physically increasing the number of sensors.

MONDAY AFTERNOON, 13 MAY 2019 WILLIS, 1:00 P.M. TO 3:30 P.M.

Session 1pAO

Acoustical Oceanography: Topics in Acoustical Oceanography

Gopu R. Potty, Cochair
Dept. of Ocean Engineering, University of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882

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

Contributed Papers

1:00

1pAO1. Variations of acoustic noise intensity accompanying internal wave solitons. Boris Katsnelson (Marine Geosci., Univ. of Haifa, Mt. Carmel, Haifa 3498839, Israel, bkatsnels@univ.haifa.ac.il), Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA), and Qianchu Zhang (Marine Geosci., Univ. of Haifa, Haifa, Israel)

In this paper, acoustical noise intensity fluctuations recorded by single hydrophones (SHRUs) in the Shallow Water 2006 experiment are studied. The area of experiment (New Jersey Atlantic shelf) is characterized by a remarkable activity of internal waves, in particular, approximately twice per day trains of nonlinear internal waves (NIW) consisting of up to ten separate peaks with the amplitudes about 10–15 m and wave front parallel to the coastal line move toward the beach. During its motion trains of NIW cross positions of five SHRUs, located about 5–8 km to each other along line perpendicular to the coast as well as thermistor's strings which allow us to estimate the shape and evolution of trains while they are propagating. It is shown that the appearance of irregular wideband sound field (amplitude by up to 20–40 dB greater than noise background) takes place when the train of NIW is passing through the location of the corresponding SHRUs. The nature of these signals is discussed, and the spectrum and specific temporal variations as well as other characteristics are analyzed. [Work was supported by ONRG, NSF, and BSF.]

1:15

1pAO2. Variability of the sound field in the presence of internal Kelvin waves in a stratified lake: The Sea of Galilee as a case study. Ernest Uzhansky, Boris Katsnelson (Marine Geosci., Univ. of Haifa, 199 Abba Khouchy Ave., Haifa 3498838, Israel, ericheg@inbox.ru), Andrey Lunkov (Prokhorov General Phys. Inst., Moscow, Russian Federation), and Ilia Ostrovsky (IOLR, I.Allon Kinneret Lab, Migdal, Israel)

The spatiotemporal variability of low- and mid-frequency sound field in the presence of internal Kelvin waves (IKWs) was studied in the Sea of Galilee. Experimental measurements of the sound field were carried out using a vertical line array (VLA) consisting of ten hydrophones with 3 m spacing. The VLA was deployed in the deepest (37 m) part of the lake. Signals were transmitted from the source deployed at the peripheral lake location at a distance of 5.5 km from the VLA at 8-m depth. Linear frequency modulation pulses (300–2000 Hz) were transmitted with 5 sec intervals during >24 h (the period of the IKWs). IKWs were registered using three thermistor chains (TCs) positioned along an offshore transect at 10-m, 20-m, and 37-m station depths. This setting allowed us to characterize the variations of the thermal structure and the corresponding sound speed profile along transect. The vertical structure of the sound field registered with the VLA shows connection with temporal variability of IKWs. The modeling of sound propagation was done using a Parabolic Equation (PE) method, taking into account the parameters of bottom and lake bathymetry. The PE results showed close agreement with our experimental measurements. [Work supported by Israel Science Foundation.]

1:30

1pAO3. Using ambient noise to evaluate the acoustic Green's function for a passive geoacoustic inversion in a dynamic shallow-water environment. Tsu Wei Tan, Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, ttan1@nps.edu), Boris Katsnelson, and Marina Yarina (Univ. of Haifa, Haifa, Israel)

Cross-correlation functions (CCFs) of ambient and shipping noise recorded by two hydrophones approximate the deterministic Green's function and contain information about the propagation environment. This paper employs the data collected in the Shallow Water 2006 experiment on the New Jersey continental shelf to investigate the factors that affect the emergence of approximate Green's functions from noise CCF estimates and accuracy of the approximation. One month-long continuous records of noise obtained by moored single-hydrophone receivers are analyzed. Hydrophones are located in 80–100-m deep water at distances of several kilometers from each other. Rapid variations of the water sound speed profile, which are primarily due to propagation of trains of strong nonlinear internal gravity waves, limit useful noise-averaging time. Available water temperature data are used to guide the selection of time windows for noise averaging and improve CCF evaluation and retrieval of information on seafloor properties. Various approaches to coherent stacking of CCFs are compared. Time warping transform is applied to the resultant noise CCF to extract dispersion curves of acoustic normal modes. The results of the geoacoustic inversion based on the passively measured dispersion curves are compared with the earlier results obtained using controlled sound sources. [Work supported by NSF and BSF.]

1:45

1pAO4. Implementation of random forest in geo-acoustic study. Zhengyu Hou (CAS Key Lab. of Ocean and Marginal Sea Geology, South China Sea Inst. of Oceanology, Chinese Acad. of Sci., 164 Xingang West Rd., Haizhu District, Guangzhou 510301, China, zyhou@scsio.ac.cn)

The correlation between sediment sound velocity (V) and physical properties has been studied for 60 years using empirical formulas and found to be difficult to predict V accurately. Random forest (RF) is a scientific discipline and a method of data analysis that automates analytical model building. Here, we present the implementation of the RF algorithm in V prediction and sediment classification. The goal of this study is to establish a predictive model based on RF using multiple physical parameters (mean grain size, porosity, wet bulk density, and water content). Compared to empirical formulas, the average error of RF velocity is only 0.95%, ranging from 0.03% to 2.73%, which has improved the accuracy of V prediction. We also used Mean Decrease Impurity importance to evaluate the importance of a variable and found that the most important feature in the predictive model is the mean grain size. The classification model based on RF reaching up to 75% accuracy in the dataset. Multiple features, such as physical properties, sedimentary environment, and sediment source, affect the geo-acoustic properties of sediments. The next goal is to use multiple features to improve the model and further improve the accuracy of sound velocity prediction and sediment classification.

2:00–2:15 Break

2:15

1pAO5. The contribution of 12 kHz multibeam sonar to a southern California marine soundscape. Hilary Kates Varghese (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, hkatesvarghese@ccom.unh.edu), Michael J. Smith (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Dover, NH), Jennifer L. Miksis-Olds (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NC), and Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

An ocean mapping survey was conducted over the Southern California Antisubmarine Warfare Range, a hydrophone range to characterize the radiation pattern of the *R/V* Sally Ride's EM 122 (12 kHz) Kongsberg multibeam echosounder. Spanning a 2000 km^2 area, the 89-hydrophones in the range, combined with the mapping survey, provided the opportunity to study the contribution of this anthropogenic noise to the marine soundscape. The soundscape was characterized and compared at selected hydrophones across the range before, during, and after the multibeam survey. One minute averages of the sound level were calculated over the data collection period. Sound level percentiles (P1, P10, P50, P90, and P99) were calculated for the full spectrum (1 Hz–48 kHz) and select frequency bands, and spectral probability density plots were generated for each time period. Frequency correlation matrices for each time period were produced and compared using difference matrices to identify changes in the soundscape. The results are placed in the context of the auditory scene of Cuvier's beaked whales resident on the range by applying a mid-frequency marine mammal weighting function. [Work supported by NOAA, ONR, and Scripps Institute of Oceanography.]

2:30

1pAO6. Estimation of environmental parameters with machine learning using a compact tetrahedral array and sources of opportunity. Jesse T. Moore, James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Middleton 14, Narragansett, RI 02882, jesse_moore@my.uri.edu), Aditi Tripathy (Ocean Eng., Univ. of Rhode Island, Kingston, RI), Makio Tazawa (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jennifer Amaral, Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Arthur Newhall, and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

In a previous paper, we showed that we could localize sound sources using a compact tetrahedral hydrophone array in a continental shelf environment south of Block Island, Rhode Island. The tetrahedral array of phones, 0.5 m on a side, was deployed to monitor the construction and operation of the first offshore wind farm in the United States. Directions of arrival (DOAs) for a number of ships were computed using a time difference of arrival technique. Given the DOAs, ranges are estimated using supervised machine learning techniques. Here, we extend this work to estimate a number of environmental parameters including water depth and sediment composition. Training sets of range-dependent ocean waveguides and sediment sound speeds were generated using a propagation model for a neural network. Data from the tetrahedral array were processed by the neural network, which provided estimates of the water depth and sediment parameters such as sound speed and density. These estimates are compared to bathymetric data and core data collected as part of the site characterization for the wind farm. [Work supported by the Office of Naval Research and the Bureau of Ocean Energy Management.]

2:45

1pAO7. Estimation of shipping noise from sparse measurements via generative adversarial networks. Johnny L. Chen (Appl. Res. in Acoust., LLC, 209 N. Commerce St., Ste. 300, Culpeper, VA 22701-2780, johnny.chen@ariacoustics.com) and Jason E. Summers (Appl. Res. in Acoust., LLC, Washington, District of Columbia)

There is growing interest in prediction of anthropogenic noise levels in the ocean. Evidence suggests that sources of ambient noise such as shipping traffic may be destructive to marine organisms that rely on acoustics for communication. Characterizing and predicting ambient noise are also critical to effective naval operations. Understanding ocean noise is constrained by the limited ability to directly measure the spatial distribution of noise levels. In this work, we present a deep-learning method to estimate the spatial distribution of ambient noise due to shipping from a small number of measurements. Noise levels are typically estimated using forward models based on statistical information about shipping routes and source levels and predictions or measurements of environmental variables including bathymetry and sound-speed profile. Inverse methods are sometimes used to estimate input parameters from *in situ* data. In contrast, we demonstrate the robust estimation of shipping noise using a pretrained generative adversarial network (GAN) as a prior. By using a context and prior loss, our algorithm is able to accurately predict the entire spatial distribution of noise from sparsely sampled measurements. This can yield greater accuracy than a forward model based on environmental parameters taken from archival databases or inferred *in situ*.

3:00

1pAO8. Broadband acoustic propagation in a seagrass meadow throughout a diurnal cycle. Kevin M. Lee, Megan S. Ballard, Jason D. Sagers, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Gabriel R. Venegas, Jay R. Johnson, Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Abdullah F. Rahman (School of Earth, Environ., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX)

Acoustic propagation in seagrass meadows is sensitive to gas produced by photosynthesis and respiration. In addition to gas volumes within the seagrass, bubbles are introduced into the water as oxygen diffuses through the plant tissue, leading to dispersion, absorption, and scattering of sound. Because the oxygen production cycle is largely driven by sunlight, these acoustical effects have a diurnal dependence. Previous work has examined the use of acoustics as a remote sensing tool for monitoring the seagrass photosynthetic activity (Hermand, 2004). In the present paper, we describe an acoustic propagation experiment conducted in a *Thalassia testudinum* meadow in the Lower Laguna Madre, a shallow bay on the Texas Gulf of Mexico coast. A spherical omnidirectional source transmitted frequency-modulated chirps (0.1 kHz to 100 kHz) every 10 min for a 24-h period, during which oceanographic probes measured water temperature, salinity, and dissolved oxygen. The received acoustic signals were match-filtered to obtain band-limited impulse responses, enabling identification of various propagation paths within the waveguide. The dependence of the received acoustic amplitude and frequency content on time-of-day, dissolved oxygen, and other environmental parameters will be discussed with the goal of using acoustics to study seagrass photosynthesis and productivity. [Work supported by ARL:UT IR&D and ONR.]

3:15

1pAO9. Effect of a soft sediment layer over elastic basement on normal mode dispersion. Gopu R. Potty, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), and Julien Bonnel (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The effect of shear on dispersion of acoustic normal modes was investigated in a previous study (Potty and Miller, 2010). Modal dispersion was calculated using a bottom model consisting of a liquid layer over an elastic basement. The modal travel times corresponding to the Airy Phase regions were found to be extremely sensitive to shear. Simple inversion schemes were developed to estimate the shear speed in the sediment by comparing theoretical predictions with experimental data. Modal dispersion characteristics of broadband data collected during experiments conducted in Middle Atlantic Bight and New England Mud patch were analyzed, and bottom shear speeds were estimated. The estimated shear speeds were also compared with shear speeds calculated from core data. In this study, the bottom model will be revised to include a soft sediment layer over the elastic basement. The modal dispersion will be calculated using this bottom model corresponding to the New England Mud patch environment using propagation models such as ORCA and KRAKENC. The effect of the addition of the soft mud layer on dispersion of lower order modes will be investigated and presented. [Work supported by the Office of Naval Research.]

Session 1pBA

Biomedical Acoustics and Signal Processing in Acoustics: Lung Ultrasound and Tissue Stiffness Method II

Xiaoming Zhang, Cochair
Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

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

Chair's Introduction—1:20

Invited Papers

1:25

1pBA1. Machine learning assisted evaluation of interstitial lung diseases. Chi Wan Koo (Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, koo.chiwan@mayo.edu)

Interstitial lung disease (ILD) is an inflammatory condition encompassing greater than 200 different chronic disorders that often lead to pulmonary fibrosis and the attendant morbidity and mortality. Distinguishing one ILD from another can be challenging even for experts given similar clinical manifestations. However, firm diagnosis is important for management, counseling, and surveillance. Computer-Aided Lung Informatics for Pathology Evaluation and Ratings (CALIPER, Mayo Clinic, Rochester, MN, USA), a machine learned image analysis tool for characterizing and quantifying diffused lung diseases on CT, has been shown to correlate with ILD mortality such as for idiopathic pulmonary fibrosis. The CALIPER assessment of ILD is not influenced by confounding conditions such as emphysema or pulmonary hypertension that may affect pulmonary function measurements. Moreover, CALIPER can provide more reproducible results with less inter- and intraobserver variability compared to the conventional subjective visual assessment of CT scans by radiologists. Herein, we aim to provide an overview of ILD and discuss CALIPER evaluation of ILD.

1:45

1pBA2. Lung ultrasound surface wave elastography for assessing interstitial lung disease. Xiaoming Zhang, Boran Zhou, Jinling Zhou, Brian Bartholmai, Thomas Osborn, and Sanjay Kalra (Mayo Clinic , 200 1st St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Many lung diseases including interstitial lung disease (ILD) are associated with changes in the lung's biomechanical properties. ILD comprises a number of serious diseases in which fibrosis stiffens and damages lung tissue. Most ILDs are typically distributed in the lung's peripheral and subpleural regions. Pulmonary function test (PFT) and high-resolution computed tomography (HRCT) are used to assess ILD. Ultrasonography is not widely used for lung assessment because ultrasound cannot image deep lung tissue. We have developed lung ultrasound surface wave elastography (LUSWE) for measuring superficial lung wave speed. In LUSWE, a 0.1-s 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 six intercostal spaces. Significant differences of surface wave speed between patients and controls were found in 6 lung regions and for 3 excitation frequencies. A positive correlation between LUSWE and clinical tests including HRCT and PTF was found. LUSWE may complement the clinical standard HRCT for assessing ILD.

2:05

1pBA3. Application of lung ultrasound surface wave elastography for assessment of extravascular lung water in patients hospitalized with congestive heart failure. Brandon M. Wiley (Cardiovascular Medicine, Mayo Clinic, 1216 Second St. SW, Rochester, MN 55905, wiley.brandon@mayo.edu), Boran Zhou (Radiology, Mayo Clinic, Rochester, MN), Govind Pandompatam (Cardiovascular Medicine, Mayo Clinic, Rochester, MN), Jinling Zhou (Radiology, Mayo Clinic, Rochester, MN), Hilal Olgun Kucuk (Cardiovascular Medicine, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Lung ultrasound (LUS) detects the presence of extravascular lung water (EVLW) through the visualization of B-Line artifacts. However, the qualitative nature of LUS limits its effectiveness in serial or longitudinal studies such as evaluating changes in EVLW at different time points in patients undergoing diuretic therapy for congestive heart failure. Lung ultrasound surface wave elastography (LUSWE) is a novel technique using a small handheld device that can measure superficial lung tissue elastic properties. We aimed to evaluate the use of LUSWE to measure quantitative changes in lung elasticity caused by acute changes in EVLW. We performed LUSWE on consecutive days in 14 patients hospitalized for acute congestive heart failure with evidence of pulmonary edema (clinical EVLW). From day#1 to day#2, the patients had an average diuresis of net negative 2.1 l associated with an average decrease in 13 B-

Lines by lung ultrasound, signifying a reduction in EVLW. LUSWE analysis demonstrated a significant reduction ($p < 0.05$) in surface wave velocity in all interrogated intercostal spaces from day#1 to day#2. In summary, LUSWE performed at the bedside was able to demonstrate improvement in lung compliance (decreased elasticity) correlating with a reduction in EVLW in hospitalized patients being treated for congestive heart failure.

2:25

1pBA4. Dedicated signal processing for lung ultrasound imaging: Can we see what we hear? 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 and monitoring of the lung condition is nowadays receiving growing attention from both the clinical and technical world. The advantages of ultrasound are in fact numerous when compared to other imaging modalities such as CT: availability at patient site, low cost, real time, and safety. However, despite the vast amount of medical evidence showing how ultrasound can be used to gather diagnostic information, dedicated technical developments are still lacking. This leaves the clinitians with the only option of using standard ultrasound scanners and probes and consequently implies that decisions are often based on imaging artifacts. Standard ultrasound imaging is in fact based on the assumption that among the structures present in the field of view, only little variations in the speed-of sound are present. This is clearly not the case with the lung, due to the presence of air. In this talk, recently developed imaging modalities and signal processing techniques dedicated to the analysis of the lung response to ultrasound will be introduced and discussed. *In vitro* and clinical data will be presented which show how the study of the ultrasound spectral feature could lead to a quantitative ultrasound method dedicated to the lung.

Contributed Paper

2:45

1pBA5. Deep learning for automated detection of B-lines in lung ultrasonography. Ruud J. van Sloun (Eindhoven Univ. of Technol., Eindhoven, The 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 pleuraline 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.

Invited Papers

3:00

1pBA6. Diagnosis and monitoring of pulmonary fibrosis using ultrasound multiple scattering, an *in vivo* rodent study. Kaustav Mohanty (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell, Mir H. Ali, Thomas Egan (Dept. of Cardiothoracic Surgery, Univ. of North Carolina, Chapel Hill, Chapel Hill, NC), and Marie M. Muller (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

Idiopathic pulmonary fibrosis (IPF) affects 200,000 patients in the U.S. IPF is responsible for changes in the micro-architecture of the parenchyma, such as thickening of the alveolar walls, which reduces compliance and elasticity. In this study, it is proposed to verify the hypothesis that changes in the micro-architecture of the lung parenchyma can be characterized by exploiting multiple scattering of the ultrasound waves by the lung parenchyma. Ultrasound propagation in a highly scattering regime follows a diffusion process, which can be characterized using the Diffusion Constant. We hypothesize that in a fibrotic lung, the thickening of the alveolar wall reduces the amount of air (compared to a healthy lung), thereby minimizing the scattering events. Pulmonary fibrosis is created in Sprague-Dawley rats by instilling bleomycin into the airway. The rats are studied in groups of $n = 6$ (3 male and 3 female) 2, 3, and 4 weeks after bleomycin administration. This allowed us to provide a range of severity of pulmonary fibrosis for assessment. Using a 128-element linear array transducer operating at 7.8 MHz, *in vivo* experimental data are obtained from Sprague-Dawley rats, and the Diffusion Constant is calculated. Right after the ultrasound measurement, the rats are euthanized, and computed tomography scans are performed to validate the degree of fibrosis created. Significant differences ($p < 0.05$) in the D values between control and fibrotic rats showcase the potential of this parameter for diagnosis and monitoring of IPF.

3:20–3:35 Break

3:35

1pBA7. Dynamic optical coherence elastography: Emerging tool for noninvasive quantification of mechanical properties of ocular tissues. Kirill Larin (Univ. of Houston, 4800 Calhoun Rd., 3605 Cullen Blvd., Rm. 2028, Houston, TX 77204, klarin@uh.edu)

Optical coherence elastography (OCE) is an emerging method for noninvasive quantification of tissue viscoelastic properties. The underlying technology is based on Optical Coherence Tomography (OCT) imaging and analysis of external (or internal) force-induced mechanical waves propagating through the tissue. In this presentation, I will overview recent progress made in my lab on quantification of mechanical properties of ocular tissues (such as cornea and the lens of the eye) using OCE and their alternations during diseases progression. In particular, I will demonstrate that it is possible to quantify mechanical properties of the cornea and lens as a function of Intra-Ocular Pressure (IOP) and during controlled tissue modifications and treatments, such as corneal cross-linking. These results indicate that OCE is a powerful new technology that can be utilized for the nondestructive biomechanical characterization of ocular tissues in normal and pathological states and could not only assist in basic biomechanical studies but also lead to a new class of optical sensors for diagnosis of diseases. [Work supported, in part, by the U.S. National Institutes of Health (NIH) under Grant Nos. 2R01EY022362 and 1R01HL120140 and U.S. Department of Defense (DOD) Congressionally Directed Medical Research Programs (CDMRP) under Grant No. PR150338.]

1p MON. PM

3:55

1pBA8. Comparison of ocular biomechanical properties in normal and glaucomatous eyes using ultrasound surface wave elastography. Arthur J. Sit, Arash Kazemi (Ophthalmology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, Sit.Arthur@mayo.edu), Boran Zhou, and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Biomechanical properties of the eye are important in understanding glaucoma. However, specific tissues that may be affected are unclear. In this study, we compared glaucomatous and normal eyes for differences in corneal tissue elasticity (as indicated by wave speed) and global ocular rigidity. Both eyes of 10 glaucoma patients and 10 normal controls, matched for age and intraocular pressure (IOP), were included. The ocular rigidity coefficient was calculated from supine IOP measured with and without a 10 g weight added to the tonometer. The wave speed in the cornea was measured by ultrasound surface wave elastography. With this technique, a spherical-tipped probe (4 mm diameter) was placed on the closed eyelid and vibrated at 100 Hz for 0.1 s. The wave speed was calculated using the phase gradient method. Measurements for normal and glaucoma eyes were compared using generalized estimating equation models. The corneal wave speed was similar between normal and glaucomatous eyes ($P=0.4$). However, ocular rigidity was significantly lower in glaucomatous eyes ($P < 0.001$) compared with normal eyes. There was no difference in age or IOP ($P > 0.3$). Lower ocular rigidity in glaucomatous eyes suggests that a more compliant ocular shell may predispose to glaucoma. However, the lack of difference in the corneal wave speed suggests that corneal tissue is not significantly affected, and changes likely involve the sclera.

4:15

1pBA9. Evaluation of posterior sclera viscoelasticity in eyes with papilledema by using ultrasound vibro-elastography. John J. Chen (Ophthalmology and Neurology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, chen.john@mayo.edu), Boran Zhou (Radiology, Mayo Clinic, Rochester, MN), Arash Kazemi, Arthur J. Sit (Ophthalmology, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Papilledema is optic nerve swelling caused by increased intracranial pressure, which has the potential to cause significant vision loss. Papilledema is typically bilateral and symmetric but can sometimes be asymmetric and even unilateral. The cause for this asymmetry is unknown. The purpose of this study was to utilize ultrasound vibro-elastography (UVE) to assess for biomechanical differences in eyes with papilledema. Nine patients with papilledema and 9 age-matched controls were enrolled. An external harmonic vibration was used to generate wave propagation through the eyelid with three excitation frequencies of 100, 150, and 200 Hz. A 6.4 MHz ultrasound probe was used to noninvasively measure the wave propagation in the posterior sclera to provide shear wave speeds and viscoelasticity (fit with Voigt model). The magnitudes of the shear wave speed and viscoelasticity of the idiopathic intracranial hypertension patients' posterior sclera were significantly higher than those of healthy subjects. Moreover, for patients with unilateral papilledema, the magnitudes of wave speed and viscoelasticity of the posterior sclera were statistically higher in eyes with papilledema than in the contralateral eyes without papilledema. UVE provides a noninvasive technique to measure the viscoelastic properties of the posterior sclera, which is stiffer in eyes with papilledema.

4:35

1pBA10. A comparison of hyperelastic constitutive models applicable to Shear Wave Elastography (SWE) data in tissue-mimicking materials. David Rosen and Jingfeng Jiang (Biomedical Eng. Dept., Michigan Technol. Univ., 1400 Townsend Dr., M&M 309, Houghton, MI 49931, jjiang1@mtu.edu)

Shear wave elastography (SWE) techniques have received substantial attention in recent years. Strong experimental data in SWE suggest that shear wave speed changes significantly due to the known acoustoelastic effect (AE). This presents both challenges and opportunities toward the *in vivo* characterization of biological soft tissues. In this work, under the framework of continuum mechanics, we model a tissue-mimicking material as a homogeneous, isotropic, incompressible, hyperelastic material. Our primary objective is to quantitatively and qualitatively compare experimentally measured acoustoelastic data with model-predicted outcomes using multiple strain energy functions. Our analysis indicated that the classic neo-Hookean and Mooney-Rivlin models are inadequate for modeling the AE in tissue-mimicking materials. However, a subclass of strain energy functions containing both high-order/exponential term(s) and second-order invariant dependence showed good agreement with experimental data. Based on data investigated, we also found that discrepancies may exist between parameters inversely estimated from uniaxial compression, and SWE data though Mooney plots were consistent between the uniaxial compression and AE results. Overall, our findings may improve our understanding of the clinical SWE results.

Contributed Paper

4:55

1pBA11. An ultrasound surface wave elastography technique for non-invasive measurement of scar tissue. Boran Zhou (Radiology, Mayo Clinic, 321 3rd Ave. SW, Rochester, MN 55902, Zhou.Boran@mayo.edu), Saranya P. Wyles, Alexander Meves (Dermatology, Mayo Clinic, Rochester, MN), Steven Moran (Plastic Surgery, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Hypertrophic scars and keloids are characterized by excessive fibrosis and can be functionally problematic. Indeed, hypertrophic scarring is characterized by wide, raised scars that remain within the original borders of injury and have a rapid growth phase. There is a need for quantitative scar measurement modalities to effectively evaluate and monitor treatments. We aim to assess the role of the non-invasive scar-measuring device, ultrasound surface wave elastography (USWE), in accurately evaluating scar metrics. Two sites of breast tissue were tested control and scar portions. In USWE, a small, local, and 0.1-s harmonic vibration at three excitation frequencies (100, 150, and 200 Hz) was generated on these sites, and the resulting surface wave speed was measured via an ultrasound probe with a central frequency of 6.4 MHz. There was a statistically significant difference in the wave speed at three frequencies of the scar portion between prior and after treatment, suggesting that the scar portion was softener after treatment. USWE provides an objective assessment of the reaction of the scar to injury and treatment response.

MONDAY AFTERNOON, 13 MAY 2019 BREATHITT, 2:00 P.M. TO 4:35 P.M.

Session 1pMU

Musical Acoustics and Signal Processing in Acoustics: Transient Phenomena in Wind Instruments

Vasileios Chatziioannou, Chair

Department of Music Acoustics, University of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Building M, Vienna 1030, Austria

Chair's Introduction—2:00

Invited Papers

2:05

1pMU1. Multiphonic modeling using impulse pattern formulation. Simon Linke, Rolf Bader (Systematic Musicology, Univ. of Hamburg, Finkenau 35, Hamburg 22081, Germany, simon.linke@haw-hamburg.de), and Robert Mores (Media Technic, Hamburg Univ. of Appl. Sci., Hamburg, Germany)

Multiphonics, the presence of multiple pitches within the sound of wind-instruments can be produced in several ways. Either complex fingerings are used or the blowing pressure is very low or very high. Such multiphonics can be modeled by the Impulse Pattern Formulation (IPF) proposed previously [R. Bader, *Nonlinearities and Synchronization in Musical Acoustics and Music Psychology* (2013)]. This top-down method assumes musical instruments to work with impulses which are produced at a generator, travel through the instrument, are reflected at various positions, are exponentially damped, and finally trigger or at least interact with succeeding impulses produced by the generator. While modeling sounds produced at blowing-threshold, the IPF fully captures transitions between regular periodicity at nominal pitch, bifurcation scenarios, and noise, just like regular instruments do, when multiphonics appear in the transition regime. Using IPF, complex fingerings translate to multiple reflection points at open finger holes with different reflection strengths. Here, complex multiphonics can be modeled. The IPF can also synthesize multiphonic sounds when applying the typical impulse form of wind instruments.

2:25

1pMU2. Initial transients in free reed instruments: A survey of experimental results. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

For free reeds in steady oscillation, the fundamental transverse beam mode dominates, but transverse modes and some torsional modes are also typically present. The motion of the free-reed tongue in early stages of the initial transient has been studied experimentally with the aim of determining the presence of higher modes and their possible role in the attack transient. These transients have been studied for free reeds mounted on a wind chamber with several methods used to initiate the attack transients. The resulting displacement and velocity waveforms have been studied using laser vibrometry, variable impedance transducer proximity sensors, and high speed video with the tracking software. The most realistic procedure used a pallet valve mechanism simulating the attack transient in a key-

operated instrument. Short-term spectra derived from the waveforms have been analyzed, showing that both higher transverse modes and some torsional modes are observed in the initial transient, with the second transverse mode and the torsional mode especially prominent in the earlier stages of oscillation. Comparisons of reed tongues of different designs have been made to explore the role of these modes in the initial excitation.

2:45

1pMU3. Detecting articulations in clarinet playing. Jack D. Gabriel and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, jgabriel@rollins.edu)

A method is suggested to automatically distinguish slurred transitions from tongued transitions in clarinet playing based on the mouth pressure signal. Player data will be presented from musicians playing a clarinet with a sensor-equipped mouthpiece. Playing parameters such as blowing pressure and pressure in the mouthpiece were captured, and data recorded in controlled tests and musical passages will be presented. Possible future applications of the method will be discussed, such as automatic detection of the quality clarinet and clarinet accessories.

3:05

1pMU4. Reproducing tonguing strategies in single-reed woodwinds using an artificial blowing machine. Montserrat Pàmies-Vilà, Alex Hofmann, and Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, pamies-vila@mdw.ac.at)

Articulation on woodwind instruments is achieved inside the player's mouth, where the tongue interacts with the vibrating reed, while the player adjusts the blowing pressure, the lip force, and the vocal tract configuration. The performed articulation technique defines the characteristics of the attack and release transients and thus the transitions between tones. In this study, an artificial blowing machine with a built-in tonguing system is used to analyze different tonguing strategies in single-reed woodwinds. The tonguing system is controlled via an electronically monitored shaker, offering the possibility to reproduce tongue articulation, while assuring repeatability. To reproduce different playing techniques, parameters obtained from measurements with players are used to set up the pressure in the artificial mouth and the behaviour of the tonguing system. During the experiment, the artificial-mouth pressure, the mouthpiece pressure, the reed displacement, and the shaker acceleration are recorded. The recorded signals are then compared to real-playing clarinet and saxophone measurements. Different trajectories for the artificial tongue are tested as well as different tongue-reed-contact durations. The artificial blowing and tonguing set-up, along with the images obtained with a high-speed camera, provide an in-depth understanding of the processes taking place inside the player's mouth.

3:25–3:40 Break

3:40

1pMU5. Imaging of transient and steady-state flow in organ pipes. Thomas Moore and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

The transient and steady-state flow at the flue and open end of organ pipes has been studied using high-speed electronic speckle pattern interferometry. We show that the flow at the flue is similar to square and round organ pipes; however, at the open end, the flow depends on the pipe shape. There is a significant flow from a square pipe and no detectable flow from a round pipe. These results can be directly compared to the results of simulations recently developed by Thacker and Giordano. [Work supported by NSF Grant #PHY-160749.]

4:00

1pMU6. Transients in wind instruments modeled with the Navier-Stokes equations. Nicholas Giordano (Phys., College of Sci. and Mathematics, Auburn Univ., Auburn, AL 36849, njg0003@auburn.edu)

Transient components make an important contribution to the "color" of a musical tone. While such transients can be observed in experiments, realistic modeling can be very challenging. We describe a modeling study of initial transients in the tone produced by a recorder using Navier-Stokes-based simulations. We have studied how the harmonic content of a recorder tone during the attack portion of the tone depends on a variety of factors including the labium position, the presence of chamfers at the exit of the windway, and the initial dynamics of the blowing pressure. While our results are obtained for a realistic model of the recorder, they should also be applicable to similar instruments such as flue organ pipes. [Work supported by NSF Grant PHY1513273.]

Contributed Paper

4:20

1pMU7. Computational studies of flow in flue pipes. Jared W. Thacker and Nicholas Giordano (Phys., Auburn Univ., 210 East Thach Ave., Apt. 24E, Auburn, AL 36830, jwt0024@tigermail.auburn.edu)

Direct numerical solutions of the Navier-Stokes (NS) equations have been used to study the air flow in a recorder. When the recorder is driven with a DC flow into the windway, we observe the familiar oscillating flow out of the window adjacent to the labium. The associated flow at the open end is also studied. For a pipe with a square cross-section, we find the flow at the open end to be small but nonzero, consistent with recent studies using speckle pattern interferometry by Coyle and Moore. In addition, we have analyzed the NS results for the air density so as to compare directly with the interferometry experiments. We also describe simulations for circular pipes. [Work supported by NSF Grant PHY1513273.]

Session 1pNS

Noise, Psychological and Physiological Acoustics: Acoustic Vehicle Alerts: Effects on Soundscape, Quality of Life, and Traffic Safety

Jeanine Botta, Cochair
SUNY Downstate Medical Center School of Public Health, 720 East 31st Street, Apartment 7G, Brooklyn, NY 11210

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

Chair's Introduction—1:30

Invited Papers

1:35

1pNS1. Acoustic vehicle alerts and sleep disruption: A content analysis of online complaints and inquiries. Jeanine Botta (The Right to Quiet Society for Soundscape Awareness and Protection, 720 East 31st St., Apartment 7G, Brooklyn, NY 11210, jeanine.botta@downstate.edu)

Research on the effects of traffic noise on sleep focuses on sounds created by vehicles that are traveling, whether in continuous motion on a highway or road or temporarily slowed on a congested street. Most acoustic vehicle alerts are introduced without consideration for their potential to disrupt sleep or affect quality of life. Panic alarm, marketed as a safety feature and standard with most cars, was accepted by regulatory and consumer protection agencies without question or concern for its necessity or its potential to create new noise. Criticism about car alarms, remote lock signals that use horn sounds, panic alarm, and backup beeping in passenger cars is common in everyday discourse but rare within the public health sphere. This study will use content analysis of online complaints posted in public databases, discussion forums, and car owner forums to explore common experiences related to sleep disruption. Focus will be on technologies not used while driving and will not include pedestrian alerts or backup beeping used by commercial or industrial vehicles. The study will include posts by those whose sleep has been affected and by car owners inquiring about methods of turning off a sound source out of concern for a neighbor.

1:55

1pNS2. Soundscape, traffic safety, and requirements for public health. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

In October 2018, the World Health Organization has published the Noise Guideline for the European Region strongly focusing on health effects caused by noise from different sources as transportation (road traffic, railway, and aircraft) noise, wind turbine noise, and leisure noise. As outlined in the Introduction, they "provide robust public health advice underpinned by evidence, which is essential to drive policy action that will protect communities from the adverse effects of noise." Otherwise, the new technology in the development of electrical vehicles causes regulations calling for safety reasons for alert signals that may be counterproductive with regard to a harmonic and healthy soundscape. Regulations and needs will be discussed with respect to the public health recommendations on exposure to environmental noise and soundscapes.

2:15

1pNS3. The conflict of soundscape requirements with the introduction of alert signals. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

Quiet road vehicles have to be equipped with acoustic alert signals for type approval to account for their potentially reduced audibility. According to the FMVSS No. 141, the alert sound has to be recognizable as a motor vehicle in operation that allows blind and other pedestrians to detect nearby electric vehicles or hybrid vehicles operating at lower speeds. Unfortunately, the detailed minimum sound requirements for type approval vary from regulation to regulation, for example, between FMVSS No. 141 and UNECE No. 138. The manufacturers can consider brand-related alert signal design within certain ranges but must adapt their alert sounds to the respective regulations, because an international harmonization of the signal requirements is missing. In this context, the impact on traffic noise in general and on soundscape in particular seems to be rarely systematically discussed. It is expected that some side issues will emerge with the systematic introduction of alert signals. Moreover, the effectiveness of the different regulations regarding the significant reduction of pedestrian collision risk was not systematically analyzed so far. This paper will discuss potential shortcomings of the introduction of alert signals regulated by law, and the implications on urban soundscapes will be outlined.

2:35

1pNS4. How to design vehicle alert signals? Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

E-Mobility becomes more and more popular. At the same time, acoustical alert signals are requested by law. This is in conflict with less traffic noise and better environmental sound. Especially, the request that the alert signals should contain tones, and these tones should have a frequency shift in the dependency of speed can create disharmonic sounds, e.g., if more than one vehicle is present. This will be a negative impact on the soundscape, and the annoyance will increase. Different concepts of alert signals will be discussed with respect to detectability, localization, and sound quality.

2:55

1pNS5. Should hybrid and electric vehicles have acoustic alerting systems? Rene Weinandy (Noise Abatement in Transport, German Environment Agency, Woerlitzer Platz 1, Dessau-Rosslau 06844, Germany, rene.weinandy@uba.de), Lars Schade, and Jan Gebhardt (Noise Abatement in Transport, German Environment Agency, Dessau-Rosslau, Saxony-Anhalt, Germany)

Noise is an oft-overlooked environmental issue within densely populated regions. Vehicles, railways, and airports operating within or near cities are all contributing to the growing noise pollution problem—causing negative health and economic impacts. Due to this, it is of primary importance to make our cities quieter. The German Environment Agency is working on noise and its effects on humans, especially with respect to policy in order to make traffic as quiet as possible by addressing all the relevant elements from roads and tracks to vehicles, operational procedures, and measures along the sound propagation path. Europe, as well as most of the world, faces a future full of environmentally friendly hybrid or pure electric road vehicles. Concerns were raised that these low-emission vehicles could pose a risk to blind and low vision pedestrians. To address this concern, the European Union has legislated that future hybrid and pure electric cars must be equipped with acoustic vehicle alerting systems (AVAS). The presentation provides a critical assessment of the effectiveness of AVAS and of their negative side effects. Furthermore, it explores alternative non-acoustic approaches addressing aspects such as environmental protection, road safety, feasibility, and usability.

3:15

1pNS6. Upgrading backup alarms to reduce encroachment on soundscapes in Denali National Park. Davyd Betchkal (Natural Sounds and Night Skies Div., National Park Service, MP 237 Parks Hwy., P.O. Box 9, Denali Park, AK 99755, davyd_betchkal@nps.gov) and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

Denali maintains a sizeable fleet of maintenance vehicles that are equipped with backup alarms. Across many years, park acoustical monitoring demonstrated that tonal backup alarms were audible at one to two miles distance from the hazard zone immediately behind the truck. Following a successful demonstration of backup alarms with broadband signals, 77 vehicles at Denali were upgraded. The aggregate cost of the devices was just under $11,000, and the Denali auto shop provided 25 h of staff time for the upgrade. CadnaA modeling was used to compare the park area in which the old and new alarms were audible, assuming that both broadcast the same level (107 dB, A-weighted). The broadband alarm was predicted to be audible in 2.9 km^2, compared with 6.9 km^2 for the tonal alarm. The adaptive level feature of the new alarms yielded additional benefits in quiet locales. After the upgrade, backup alarm noise became noticeably less along the Denali Road corridor, and these improvements motivated the park to incorporate language regarding backup alarms into construction contracts.

Session 1pPAa

Physical Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, Noise, Psychological and Physiological Acoustics, and Speech Communication: Battlefield Acoustics II

W. C. Kirkpatrick Alberts, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20723

Gregory W. Lyons, Cochair
Construction Engineering Research Laboratory, U.S. Army Engineer Research and Development Center, 2902 Newmark Dr., Champaign, IL 61822

Chair's Introduction—1:00

Invited Papers

1:05

1pPAa1. Hearing on the battlefield: The challenge of protecting hearing while enabling the mission. Eric R. Thompson and Brian Simpson (711th Human Performance Wing, Air Force Res. Lab., 2610 Seventh St., B441, Wright-Patterson AFB, OH 45433, eric.thompson.28@us.af.mil)

The auditory sense plays a critical role in human performance on the battlefield. Effective verbal communication is key to the success of all military operations, the ability to detect and localize sounds in the immediate environment is critical for supporting situation awareness, and the ability to identify sound sources is required for labeling items and events as hostile or friendly. However, operational environments tend to be extremely noisy, and most warfighters are regularly exposed to hazardous noise levels, necessitating the use of hearing protection to reduce the chance of experiencing temporary or permanent hearing loss. Unfortunately, the use of hearing protection can have a deleterious effect on communication, sound source identification, and overall situation awareness. For this reason, operators often choose to perform all, or part, of their mission without the appropriate hearing protection in place. In order to protect warfighters' hearing, appropriate hearing protection needs to be selected that will provide enough protection from the expected hazardous sounds, while also enabling the hearing-critical tasks that are required for the mission. This presents a great challenge for those who make the decisions on what hearing protection to prescribe.

1:25

1pPAa2. Improving speech intelligibility performance for aircraft maintainers by selecting an appropriate combination of hearing protection devices. Hilary Gallagher (Battlespace Acoust. Branch, Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, hilary.gallagher.1@us.af.mil) and Billy Swayne (Systems Eng. Solutions, Ball Aerosp. and Technologies Corp., Wright-Patterson AFB, OH)

Aircraft maintainers work in hazardous noise environments throughout their career. Hazardous noise levels on the flight line vary, affecting the level of hearing protection required to reduce their risk of hearing loss and other hearing related disabilities. Current practice is to provide double hearing protection: a circumaural headset/earmuff with foam earplugs. This combination may provide an adequate amount of protection from aircraft noise, but the proper use of foam earplugs may degrade speech intelligibility (SI) performance. Due to this degradation, many maintainers fit the foam earplugs incorrectly in order to improve SI performance, therefore lowering the amount of protection provided. The objective of this study was to assess the noise attenuation and SI performance of a headset and filtered earplugs worn alone and in combination to provide a more appropriate recommendation of hearing protection devices. The filtered earplug can provide suitable levels of protection when worn alone or in combination with a headset (depending on the noise environment) and can provide adequate "hear through" capability for acceptable SI performance when communicating face-to-face (single hearing protection) or through the aircraft communication system (double hearing protection).

1:45

1pPAa3. Acoustic validation of military aircraft noise models. Alan T. Wall and Frank S. Mobley (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com)

The National Environmental Policy Act (NEPA) of 1970 established the requirement to assess noise impacts from aircraft operations on the community surrounding military bases. A process for measuring and modeling aircraft noise sources and their propagation to ground locations was developed in response. The NOISEMAP suite of software programs produces noise footprints of yearly average impacts due to all flights that occur to and from an airbase. Through NOISEMAP's decades of evolution, it retains the same fundamental measurement/modeling architecture and remains the approved model for environmental impact studies for aircraft within the U.S.

Department of Defense (DoD). However, new advanced source and propagation models exist, which result in lower uncertainties and the calculation of additional useful noise impact metrics. No clear path exists to allow the use of these models for DoD environmental impact studies. This work presents a scientific validation experiment for aircraft overflight source/propagation models, focusing on suitability for use in a NEPA environmental impact study. The experiment relies on high-fidelity data collected during controlled overflight acoustic measurements of real aircraft and provides a quantitative benchmark for the evaluation of future models.

2:05

1pPAa4. Acoustical modeling considerations in noise assessment of live fire training ranges. Michael J. White and Michelle E. Swearingen (US Army ERDC/CERL, P.O. Box 9005, Champaign, IL 61826, michael.j.white@usace.army.mil)

In the US, long-term noise assessments near military firing ranges are required to estimate the weighted average of simulated sounds from thousands of noise-producing activities. Each of these activities are modeled with a set of parameters and heuristics. Parameters include source type, including quantification of firing and munition explosive charges; location information; high or low angle of fire designation; and basic environmental conditions including time of day, time of year, generalized surface type, and some meteorological information. Heuristics model the source strength and directivity, divergence of waves, ground reflection, atmospheric refraction, absorption of sound, and shielding by noise barriers and terrain. The heuristics represent limitations to the fidelity of the modeling result, and they produce estimates with significant uncertainties. In view of the uncertainties, validation of results is particularly difficult and must be performed on single activity calculations rather than long-term averages. To evaluate the uncertainty, we compare single event calculations to controlled measurements, thereby minimizing the parameter uncertainty. These ideas are discussed, and an assessment of the model uncertainty is presented.

Contributed Papers

2:25

1pPAa5. Nonlinear tuning curve demonstration: Comparing a buried mine simulant with a clamped elastic plate—Cylindrical soil column oscillator. Ava B. Twitty (Phys. Dept., U.S. Naval Acad., 5568 Carvel St., Churchton, MD 20733, avat1226@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

Experiments using soil-plate-oscillators (SPO) involve a cylindrical column of granular media (masonry sand) supported by a clamped circular elastic acrylic plate (12.7 cm diameter, 3.2 mm thick). The plate is clamped between two 20.3 cm O.D., 12.7 cm I.D., 6.4 cm thick flat toroidal brass "rings." Two 15 cm diameter subwoofers (10 cm above the soil) are driven by an amplified swept sinusoidal chirp which drives the 2.5-cm soil column. A spectrum analyzer measures the laser Doppler vibrometer particle velocity versus frequency near the center of the column. The resonant frequency decreases with the increasing amplitude—representing softening in the nonlinear system. The back-bone curve (locus of the resonant frequency versus corresponding peak velocity coordinates) has a distinct arching shape where the slope of the velocity increases with the decreasing frequency. Here, the resonant frequency goes from 247 to 203 Hz. A lumped element bilinear hysteresis model describes the shape of the tuning curves and backbone curve. Next, a drum-like simulant (made by replacing the upper toroidal ring by a 0.64-cm thick ring) is buried 2.5-cm deep in an open square concrete tank (57 cm). Nonlinear tuning curve experiments "on" and "off" the mine are compared with the SPO results.

2:40

1pPAa6. Inter-array coherence-based identification of static acoustic clutter sources. W. C. Kirkpatrick Alberts (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20723, william.c.alberts4.civ@mail.mil)

In multi-array acoustic detection and tracking applications, there often exist valid acoustic sources that are unwanted because of their position or nature, e.g., generators and air-handling equipment. During a tracking scenario where desired targets can be moving and/or piecewise static, the presence of these clutter sources can negatively impact the localization of a desired target by interrupting a track or generally increasing the noise floor local to an array. In directions away from the clutter source, beamforming methods onboard an array, e.g., minimum variance distortionless response, can be used to suppress the signature of the unwanted source. However, when a desired target passes through the azimuth of a clutter source, beamformers may fail to separate the desired source from the clutter. Here, inter-array coherence between beamformed solutions will be used to identify a static source by its properties in order to remove the clutter source as a potential target in a tracking application. Acoustic data on clutter sources measured by multiple microphone arrays separated by tens of meters will be discussed.

Session 1pPAb

Physical Acoustics, Archives and History, and Education in Acoustics: On His 100th Birthday, Isadore Rudnick Speaks for Himself

Julian D. Maynard, Chair
Physics, Penn State University, 104 Davey Lab, Box 231, University Park, PA 16802

Chair's Introduction—3:15

Invited Papers

3:20

1pPAb1. Isadore Rudnick's spectacular acoustics demonstrations. Robert M. Keolian (Sonic Joule LLC, 732 Holmes St., State College, PA 16803, keolian@psu.edu), Steven R. Baker (Phys., Naval Postgrad. School, Marina, CA), and Arthur Huffman (Phys., Univ. of California, Los Angeles, CA)

While soft-spoken in conversations and lectures, Isadore Rudnick could shout when teaching acoustics with demonstrations. At a special, auditorium stage, video-taped plenary session of the Fall 1980 meeting of the Acoustical Society of America, Izzy presented 90 min of spectacular acoustic demonstrations. The demonstrations ranged from simple ones common to acoustics classes to ones that filled the stage and involved elaborate equipment. This talk will be followed by a viewing of Izzy's plenary session video. It is hard to imagine that any acoustician would not find something of interest in, and indeed learn something from, this video. For those who wish to study Izzy's demonstrations further or present the material in a classroom, the video is available for ordering as advertised in the Journal of the Acoustical Society of America.

5:00

1pPAb2. The unusual properties of liquid helium. Steven L. Garrett (151 SycamoreDr., State College, PA 16801, sxg185@psu.edu)

Each year since 1925, one UCLA faculty member has been selected to present the Faculty Research Lecture. In 1976, Isadore Rudnick was so honored. As an experimentalist, Izzy chose to make his presentation a lecture demonstration, like those of Michael Faraday's famous Christmas presentations at the Royal Society in London. This required that an entire low-temperature laboratory be recreated on the stage of Schoenberg Hall, including a transparent Dewar vessel. The success of those live demonstrations, and the fact that two of Izzy's sons were film makers, prompted him to make a 17-min film version of those demonstrations. That film won the award for best Technical and Scientific Film at the 21st Annual San Francisco International Film Festival, in 1977. Also, at this same time, the Cultural Revolution in China had ended, and Izzy was one of the first U.S. physicists invited to lecture there. As a gift to his Chinese hosts, he had this film translated into Mandarin. Due to the scarcity of the material available for those early Chinese television broadcasts, the film was played frequently, and Izzy became a TV celebrity throughout China. The film, shown in this talk, is included in ASA's *Collected Works of Isadore Rudnick*.

Session 1pPP

Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics: Applications of Signal Detection Theory in Perception and Physiology

Jennifer Lentz, Cochair
Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405

Christopher Conroy, Cochair
Boston University, 635 Commonwealth Ave., Boston, MA 02215

Contributed Papers

1:05

1pPP1. Lessons learned from signal-detection-theory-oriented studies of the processing of auditory patterns. Charles S. Watson (Speech and Hearing Sci., Indiana Univ., CDT, Inc., 3100 John Hinkle Pl, Bloomington, IN 47408, watson@indiana.edu)

One extension of Fechner's (1860) methods by which the influence of sensory stimuli on human observers could be quantified in physical units, sometimes termed a new psychophysics was Stevens (1938) theory and methodology of perceived magnitude estimation that attempted to bridge the gap between sensation and perception. A more far-reaching second extension began in 1954, when Peterson, Birdsall, and Fox applied statistical decision theory to the recovery of physical signals embedded in background noise and Tanner, Green, and Swets applied those principles to develop a new psychophysical methodology. What became known as "signal detection theory" provided methods of determining the capacity of systems to detect, discriminate, or recognize physical stimuli, a second new psychophysics. The two alternatives to Fechner represent complimentary approaches to a single set of phenomena, one of which can be characterized as the study of *sensory capabilities* and the other of *response proclivities*. The two require entierly different methods of measurement and address fundamentally different questions. Both are of practical and theoretical importance. Levels of stimulus uncertainty and the type and duration of experience with stimuli can have profound influences on measurements of both types. Examples of these effects for complex sounds, both speech and nonspeech, will be presented.

1:20

1pPP2. Signal detection theory and psychoacoustics. Christopher Conroy (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cwconroy@bu.edu) and Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

This talk will trace the early history of signal detection theory (SDT) with a particular emphasis on its applications in psychoacoustics. SDT, as developed at the University of Michigan (U-M) in the mid-1950s, revolutionized psychophysics by introducing a core assumption: all sensory judgements are limited by "noise" of one sort or another and, as such, are reflective of underlying decision processes. To test and flesh-out this new theory, early empirical work in vision quickly turned to audition and a remarkably fruitful 12-year period followed, from roughly 1954 to 1966. This talk will focus on that 12-year period, bracketed, on the one end, by the establishment of an auditory research laboratory at U-M by Wilson P. "Spike" Tanner and, on the other, by the publication of Green and Swets' (1966) seminal text. The principle documents and publications of that period—particularly those that appeared in this society's *Journal*—will be reviewed, with a discussion of their intellectual and historical context. Emphasis will be given to the investigations of the Electronic Defense Group at U-M and those of other legendary psychoacousticians. In particular, the work of Tanner, the key instigator in the transition from "statistical decision theory" to SDT in psychoacoustics, will be discussed.

Invited Papers

1:35

1pPP3. A detection-theoretic approach to cocktail-party listening. Robert Lutfi, Brianna Rodriguez (Dept. Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL 33620, rlutfi@usf.edu), and Jungmee Lee (Commun. Sci. and Disord., Univ. of South Florida, Madison, WI)

Cocktail-party listening (CPL) is a key-term referring to difficult listening situations wherein one must h" and follow the speech of individual talkers in a crowd. Such situations are commonplace in everyday listening but can be a challenge to study. Important factors affecting performance, such as voice similarity among talkers and uncertainty associated with the dynamic variation of speech, can be difficult to quantify. Spectral and spatial acoustic cues distinguishing talkers are of critical importance but are expressed in different physical units making their relative role difficult to evaluate beyond the particulars of a study. There are also often huge individual differences in listener performance that can complicate the interpretation of results. This paper reviews recent applications of detection theory designed to address these challenges. The distinguishing feature of the approach is that key factors are evaluated in terms of their contribution to the information divergence of talkers, D_{KL}; a single statistic that dictates optimal performance for each task [Lutfi *et al.*, *J. Acoust. Soc. Am.* **134**, 2160–2170 (2013)]. Both published and unpublished studies are reviewed demonstrating the application of the approach to each of the major challenges described above. [Work supported by NIDCD R01-DC001262].

1:55

1pPP4. On the theoretical benefit of combining multiple observations: Dependence on the form of the probability distribution of the observations. Huanping Dai (Speech Lang. and Hearing Sci., Univ. of Arizona, P.O. Box 21071, 1131 E. 22nd St., Tucson, AZ 85721-0071, hdai@email.arizona.edu) and Emily Buss (Dept. of Otolarynology/Head and Neck Surgery, Univ. of North Carolina, Chapel Hill, Chapel HIll, NC)

Auditory research has benefited immensely from the application of the Signal-Detection Theory (SDT; Green and Swets, 1966). The theory has provided a rigorous framework within which the design of experimental studies and the interpretation of experimental results can be carried out in a principled way. A fundamental assumption often made, either explicitly or implicitly, in the application of SDT is that the observations obey Gaussian distributions. Many important results were derived on an equal-variance Gaussian assumption for the "noise" and the "signal"events. One example is the well-known relation between the detectability attainable based on a single observation (d'_1) and the detectability that can be achieved when multiple ($m > 1$) independent and equally informative observations are combined: $d'_m = \sqrt{m}\, d'_1$. As researchers often need to deal with situations where the form of the probability distributions of the observations either is unknown or is known to be non-Gaussian, it is important to understand how the form of the probability distribution can influence the outcome. This presentation will examine different ways by which the interpretation of listeners' performance with multiple observations would depend on the form of the probability distributions.

2:15

1pPP5. Physiological parallels with perception. John C. Middlebrooks (Otolaryngol., Univ. of California at Irvine, Rm. 116 Medical Sci. E, Irvine, CA 92697-5310, j.midd@uci.edu)

Signal detection theory (SDT) lends itself to comparison among perceptual and physiological results in humans and other animals. In physiological recordings, trial-by-trial distributions of single-unit spike counts or of event-related-potentials typically yield non-normal distributions, calling for non-parametric tests. An empirical receiver operating characteristic (ROC) curve can compare distributions of physiological measures between pairs of stimulus conditions. The area under the ROC curve is a non-parametric measure of sensitivity, yielding the proportion correct and, thence, the sensitivity index, *d'*. The *d'* cumulated over a succession of the increasing values of a stimulus parameter provides a measure of the growth of response over that parameter range. I will present examples from several ongoing projects including: (1) studies of activation of the central auditory pathway by cochlear implants and by a penetrating auditory nerve electrode, with single-unit recordings from the inferior colliculus; (2) comparisons of spatial stream segregation among 2-alternative forced choice in humans, hold-response measures in cats, and cortical single-unit recording in cats; and (3) studies of frequency sensitivity using the auditory change complex and masked onset responses. In general, these techniques derived from SDT yield a satisfying agreement among perceptual measures and their physiological underpinnings.

2:35

1pPP6. Accounting for a wide variety of binaural detection data by combining cross-correlation and signal-detection theory approaches. Leslie R. Bernstein and Constantine Trahiotis (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, Farmington, CT 06032, lbernstein@uchc.edu)

We will discuss several experimental contexts in which a Theory of Signal Detection (TSD) approach to modeling has yielded successful, straightforward, and intuitively appealing accounts of data concerning binaural auditory processing. The primary focus will be on recently published empirical data and quantitative modeling from our laboratory. Those reports demonstrate that data obtained in binaural detection experiments conducted across the last five decades can be accounted for by combining a signal-detection-based decision variable with a cross-correlation-based model of binaural processing that incorporates stages of peripheral auditory processing. Notably, some of the data obtained in those experiments had, therefore, been "problematic" in that they had remained either theoretically unaccounted for or, at best, had been only accounted for via *ad hoc* approaches. Key to developing the unified account of those experimental results was the calculation and inclusion of the variability of the interaural correlations of the outputs of the model for both masker-alone and signal-plus-masker conditions. We will also discuss recent developments showing how our TSD-inspired approach has proven useful in discovering and explaining why some listeners with slightly elevated, but clinically negligible, audiometric thresholds exhibit reliable and meaningful deficits in binaural performance. [Work supported by Office of Naval Research (N00014-15-1-2140 and N00014-18-1-2473).]

2:55–3:10 Break

3:10

1pPP7. Coping with the black swan in psychophysics. Yi Shen (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S. Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

Aberrant responses that are associated with lapses in attention and microsleeps during a behavioral experiment violate basic assumptions underlying signal detection theory. These responses (i.e., the black swan events) occur infrequently but could lead to severe biases in the estimated perceptual sensitivity (d') and misinterpretations of the data. When adaptive procedures are used to estimate behavioral thresholds, performance lapses occurring early in an adaptive track may cause convergence failures. Moreover, microsleeps may become more frequent as the task difficulty decreases, leading to potential instabilities in common adaptive procedures. Through a series of Monte-Carlo simulations, the potential effects of performance lapses on typical psychophysical tasks (e.g., Yes/No, 2-alternative forced choice) were demonstrated. Several possible experimental and computational techniques to cope with performance lapses were investigated. These included (1) introducing additional parameters representing performance lapses into the psychometric model, (2) allowing a "Not Sure" response category, and (3) screening and removing the likely aberrant responses. These simulation studies provide useful guidelines to design adaptive psychophysical procedures for subject populations from which performance lapses are expected (e.g., young children and behaving animals).

3:30

1pPP8. Extending signal detection theory approaches up the central pathway: Auditory midbrain models. Laurel H. Carney (Biomedical Eng. & Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu) and Miriam Furst (Elec. Eng., Tel Aviv Univ., Tel Aviv, Israel)

The application of signal detection theory (SDT) to peripheral encoding of stimulus parameters provided a quantitative basis for understanding auditory-nerve (AN) responses. Siebert's development of this strategy, based on relatively simple analytical models, laid the groundwork for Heinz's extension that took advantage of computational AN models. These studies applied the Cramer-Rao lower bound (CRLB) to estimate just-noticeable differences of stimulus parameters, assuming that underlying neural responses could be described as non-homogenous Poisson processes (NHHP) and that the brain acts as an optimal processor. Krips and Furst carried this approach into the central nervous system, first showing that coincidence-detector neurons that receive excitatory and/or inhibitory NHHP inputs produce outputs that are also NHHP. These models were tested using the CRLB for sensitivity to interaural differences in time and level. Here, we carry the approach a step further along the auditory pathway to the auditory midbrain. Model neurons that combine excitatory-excitatory and etxcitatory-inhibitory mechanisms explain rate tuning for the amplitude-modulation frequency, a key response property of inferior colliculus (IC) neurons. These models also exhibit direction selectivity for fast frequency sweeps observed in IC responses to Schroeder-phase harmonic complexes. These models facilitate SDT predictions of psychophysical tasks involving amplitude and frequency fluctuations.

3:50

1pPP9. Data-driven methods in a model-driven world: Making sense of auditory classification images for continuous speech. Jonathan Venezia (Auditory Res., VA Loma Linda Healthcare System, 11201 Benton St., Loma Linda, CA 92357, jonathan.venezia@va.gov)

Some of my recent work has focused on the development of Auditory Bubbles, a method for generating classification images that show the acoustic features underlying different percepts of continuous speech. Auditory Bubbles work by applying randomly shaped spectrotemporal filters to the modulation power spectrum of sentence-level speech stimuli. A reverse correlation is then used to relate the filter patterns to a behavioral outcome (e.g., keyword recognition). This technique is data driven—i.e., a computational approach (regression) is taken to estimate the relation between stochastically varying acoustic-speech features and listener responses (the classification image). For a complex stimulus such as continuous speech, this relation may be determined by multiple underlying factors—e.g., task demands, listener characteristics, response strategies, and perceptual acuity. As such, it can be difficult to interpret classification images, which (blindly) reflect the combined influence of these underlying factors. In this talk, I present the classification images from multiple speech experiments comparing (a) different stimuli (single- versus multi-talker), (b) different tasks (intelligibility versus emotion perception), and (c) different listener groups (persons with versus without hearing loss, musicians versus non-musicians) and discuss preliminary efforts to constrain the interpretation of classification images by appealing to the tenets of signal detection theory.

4:10

1pPP10. Multidimensional signal detection theory: Theoretical issues and empirical applications. Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu)

Signal detection theory (SDT) provides a powerful set of tools for modeling choice data in a wide variety of domains. Fundamental to SDT is the distinction between noisy evidence and deterministic response selection. Multidimensional signal detection theory (MDSDT, aka general recognition theory) extends the basic concepts of SDT so that they are applicable to more complex tasks and data. Building on the SDT foundation of noisy evidence and deterministic response selection, MDSDT adds multiple ways in which dimensions may (or may not) interact. The additional complexity of multiple dimensions both broadens the range of applications of these tools and presents unique challenges. I will discuss the origins of MDSDT, a number of empirical applications of MDSDT in speech and hearing sciences, and some recent theoretical developments, focusing on model mimicry and identifiability and multilevel extensions of the basic model.

Contributed Papers

4:30

1pPP11. Regarding sources of irreducible variation in transformed up-down staircase experiments. Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 100 Exploration Way, Hampton, VA 23666, andrew.christian@nasa.gov)

A modicum of variation in the result is always expected when a psychoacoustic experiment is run multiple times on a single subject. Historically, this variation was attributed to human shortcomings: lapses in attention, inexperience, and the effects of memory, learning, and fatigue. It is demonstrated that the use of any psychophysical method that requires a binary response (i.e., a correct or incorrect response) will generate irreducible variation in the result—even for a perfect subject. At NASA, a recent battery of low-frequency simultaneous masking experiments was undertaken (see NASA-TM-2018-220120). This effort was preceded by a lengthy test design phase—also documented therein—during which it was determined that the test should consist of a large number of subjects, each to be tested minimally. For this, transformed up-down staircases were simulated in order to produce a prediction of the irreducible variance as a function of the staircases' parameters (e.g., length in the number of trials). The results from repeated test conditions taken during the experiment indicate that at least half of the variation observed within subjects may be coming from this irreducible source, though the propensity to produce inconsistent results may be a feature of some subjects more than others.

4:45

1pPP12. The invasion of psychoacoustics by inferential statistics. William Yost (ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu) and M. Torben Pastore (ASU, Troy, NY)

The basic results that form the psychoacoustic foundation of our current knowledge of auditory perception, collected over nearly 50 years, were generated without any use of null-hypothesis significance testing (inferential statistics). The common use of inferential statistics in psychoacoustic articles published in JASA did not occur until the 1980s. Yet, today it is almost impossible to publish a psychoacoustic article in JASA, or anywhere

else, without using inferential statistics. In addition, recent controversies concerning replication in the social and behavioral sciences are beginning to encroach on the field of psychoacoustics. In this presentation, a case will be made that several elements of the experimental design inherent to many, if not most, psychoacoustic experiments often make the use of inferential statistics unnecessary. These aspects (many derived from the framework of the Theory of Signal Detection) include careful experimental control of stimuli and response acquisition, ratio-scale measurements, experienced subjects, large number of trials per subject, within-subject (repeated measures) experimental design, and strong theoretical context (e.g., an ideal observer). Some scenarios of experimental design and data presentation will be considered, suggesting that inferential statistical analysis might not be needed in some psychoacoustic research. [Work supported by NIDCD and Facebook Reality Labs.]

MONDAY AFTERNOON, 13 MAY 2019 STOPHER, 1:30 P.M. TO 3:55 P.M.

Session 1pSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Architectural Acoustics: Smart Materials for Acoustics and Vibrations II

Kathryn H. Matlack, Cochair
Department of Mechanical Science and Engineering, University of Illinois at Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801

Bogdan Ioan Popa, Cochair
Mechanical Engineering, University of Michigan, University of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Invited Papers

1:30

1pSA1. Nonreciprocal metamaterial with programmable nonlinear virtual resistors. Amr M. Baz (Mech. Eng., Univ. of Maryland, 2137 Eng. Bldg., College Park, MD 20855, baz@umd.edu)

Emphasis is placed on the development of a class of active acoustic diodes and metamaterials in an attempt to control the flow and distribution of acoustic energy in acoustic cavities and systems. The proposed active nonreciprocal acoustic metamaterial (*ANAM*) cell consists of only one-dimensional acoustic cavity provided with active flexible boundaries. These boundaries are made from piezoelectric bimorphs with the inner layers which interact directly with cavity acting as sensors for monitoring the pressures of the propagating acoustic waves. The outer layers of the bimorphs provide the necessary control actions to an array of programmable nonlinear shunted resistors. These resistors are programmable and are designed in such a manner to introduce simultaneous nonlinear damping and cubic hardening stiffness effects. A lumped-parameter model of the *ANAM* cell is developed to control the nonreciprocal characteristics of the cell by the selection of the proper balance between the nonlinear damping and stiffness effects. Lyapunov stability criterion is used to generate the structure of such a balanced control strategy. The Harmonic Balance Method is used to predict the limit cycle behavior of the *ANAM*, the backbone characteristics and the stable limits of the control gains. Numerical examples are presented to demonstrate the effectiveness of the proposed *ANAM* in tuning and programming the directivity, flow, and distribution of acoustic energy propagating though the metamaterial.

1:50

1pSA2. Magnetorheological elastomer panels with reconfigurable magnetic mass for acoustic wave absorption. Carson Willey, Vincent W. Chen, Ken Scalzi (UES, Inc./Air Force Res. Lab., 2179 12th St., Bldg. 654/Rm. 304, Wright-Patterson AFB, OH 45433, carson.willey.ctr@us.af.mil), Philip Buskohl (Air Force Res. Lab., WPAFB, OH), and Abigail T. Juhl (Air Force Res. Lab., Wright-Patterson AFB, OH)

Lightweight metamaterials based elastomeric panels are used for sound absorption of low-frequency noise. It is beneficial to tune the resonant frequency of the metamaterial, by varying the stiffness of the panel, to enable use over a broader frequency range. The stiffness of magnetorehological elastomeric (MRE) panels, composed of iron particles embedded in a PDMS matrix, can be tuned by exposure to a magnetic field. This work represents an experimental realization and theoretical understanding of a magnetic mass decorated MRE metamaterial, which allows for tunable stiffness and easy reconfiguration of the masses applied to the panel. The circular panels are clamped at the rim, outfitted with neodymium magnets, and installed in an acoustic impedance tube, where they act as an acoustic barrier between the transmit and receive sides of the tube. There are three competing mechanisms that affect the sound transmission across the panel: (1) the mass increase associated with the magnetic masses, (2) the stiffening of the panel due to the magnetic field of each mass, and (3) the increased reflection from the placed masses. The experimental results demonstrate significant sound absorption by a thin metamaterial panel over a large frequency range through careful placement of magnetic masses.

2:10

1pSA3. Non-reciprocal sound propagation using acoustic resonators via space-time modulation. Junfei Li, Chen Shen (ECE, Duke Univ., 101 Sci.Dr, Rm. 3417, FCIEMAS, Durham, NC 27708, junfei.li@duke.edu), Xiaohui Zhu (ECE, Duke Univ., Harbin, Heilongjiang, China), and Steven Cummer (ME, Harbin Inst. of Technol., Durham, NC)

Breaking reciprocity is of fundamental importance in communication and signal processing and is less explored in the field of acoustics. Here, we show that non-reciprocal acoustic transmission can be realized in cascaded resonators that are periodically modulated. In the continuum limit, we also study the sound propagation in a continuously modulated effective medium. Functionalities such as mode conversion, parametric amplification, and phase conjugation are demonstrated. Finite-difference time-domain (FDTD) simulations are carried out to verify the results. An experimental platform is constructed which is composed of a waveguide system with side-loaded resonators. The back walls of the resonators are driven dynamically using speakers so that time modulation can be achieved.

2:25–2:40 Break

2:40

1pSA4. Design and fabrication of a linear broadband non-reciprocal acoustic waveguide using feedforward control. Nate Geib, Aritra Sasmal, Karl Grosh, Bogdan Ioan Popa, and Yuxin Zhai (Mech. Eng., Univ. of Michigan, 1587 Beal Ave. Apt 13, Ann Arbor, MI 48105, geib@umich.edu)

Acoustic metamaterials that exhibit non-reciprocal transmission have received substantial attention due to their wide range of applications such as noise control, diagnostic imaging, communications, and acoustic cloaking. Passive metamaterials achieve non-reciprocity through system resonances combined with nonlinearities or manipulation of phononic bandgap and are usually effective over a narrow band of frequencies. In this study, we describe the development of a new class of acoustic waveguides that exhibit linear non-reciprocal sound transmission over a wide range of frequencies. We use an open loop feedforward control mechanism where the measured local pressure is passed through an electronic controller and transmitted downstream to actuate an acoustic source. Analysis of the idealized one-dimensional model with a continuous distribution of probes and sources shows spatial asymmetry with attenuation in one direction and amplification in the other for frequencies ranging from one-half to five times the ratio of the active length of the segment to the acoustic wavelength. We show that such behavior can be replicated in a physical system with a finite array of acoustic probes and sources and develop the framework to compute the stability of this class of active waveguides. The experimental results for a simple feedforward model system are presented.

2:55

1pSA5. Non-reciprocity in mechanically modulated elastic metamaterials with geometric asymmetry. Benjamin M. Goldsberry, Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu)

Elastic metamaterials with time- and space-dependent effective material properties have received great attention as a means to generate non-reciprocal wave propagation in acoustic and elastic media. These materials have promise for applications such as acoustic communication devices with increased data throughput and improved vibration isolation components. One means to modulate the material properties of a heterogeneous medium is via nonlinear mechanical deformation on time scales that are slow compared to propagating waves. This approach has been explored by the authors using the finite element method (FEM) to demonstrate non-reciprocal elastic wave propagation in negative stiffness honeycombs with a time- and space-varying external pre-strain [Goldsberry *et al.*, *JASA* **144**, 1763 (2018)]. One benefit of FEM is that it can be used to study the degree of non-reciprocity when varying sub-wavelength geometry or geometric modulations that are difficult or impossible to represent using analytical models. The present work employs FEM to investigate non-reciprocity in elastic lattices consisting of unit cells with varied geometric asymmetry and more general forms of mechanical modulation. [Work supported by National Science Foundation EFRI under Award No. 1641078.]

3:10

1pSA6. Design and experimental validation on acoustic valley Hall edge states in reconfigurable phononic elastic waveguides. Ting-Wei Liu and Fabio Semperlotti (Purdue Univ., 177 S Russell St., West Lafayette, IN 47907-2099, liutw@purdue.edu)

We report on the design and experimental validation of a tunable topological elastic lattice capable of supporting guided waves that are robust against back-scattering from disorder and defects. The topological lattice consists in a patterned aluminum thin plate having through-holes arranged in a hexagonal periodic configuration. The occurrence of the topological edge state is attributed to the acoustic analogue of the electron quantum valley Hall effect. By connecting two lattices having broken space inversion symmetry due to the application of opposite tunable strain fields, a topological transition emerges at the domain wall (i.e., the interface between them). Such a domain wall supports the formation and propagation of quasi-unidirectional edge states. The experimental validation of the topological waveguide is conducted on an aluminum plate that is already fabricated in a deformed (i.e., broken symmetry) configuration. The results confirm the existence of quasi-unidirectional edge states traveling along the domain wall and robust against sharp corners back-scattering. The experimental results also confirm the presence of mechanical energy flux vortices within the unit cell that are used to investigate the origin of the edge states robustness against disorder.

3:25

1pSA7. Architected hollow sphere foams for simultaneously tunable noise and vibration control. Yanyu Chen (Univ. of Louisville, 332 Eastern Pkwy, Louisville, KY 40292, yanyu.chen@louisville.edu)

Architected metamaterials, which are rationally designed multiscale material systems, exhibit novel functionalities and unique properties that cannot be readily achieved in natural bulk solids. In addition to unusual mechanical and physical properties, such as high specific stiffness and strength, negative Poisson's ratio, negative coefficient of thermal expansion, and negative compressibility, architected metamaterials have been designed and optimized for novel elastodynamic wave phenomena. One example of such architected metamaterials is phononic metamaterial, which consists of periodically topological structures and materials dispersions. These rationally designed structures enable the manipulation of propagating acoustic or elastic waves but not simultaneously. Here, we report a new type of architected hollow sphere foam that can attenuate sound and elastic wave synchronically. Our numerical simulation results suggest that the acoustic wave attenuation is controlled by the drilled area and thickness of the spheres, while the elastic wave propagation can be manipulated by changing the connectivity among the hollow spheres. We printed the architected hollow sphere foams and experimentally demonstrated the existence of omnidirectional acoustic and elastic wave bandgaps. Our findings reported here offer new opportunities to design lightweight architected metamaterials to simultaneously control undesired noise and vibration over a wide range of frequency.

3:40

1pSA8. Design of broadband active sound isolators based on acoustic bianisotropic metamaterials. Yuxin Zhai, Bogdan Ioan Popa, and Hyung-Suk Kwon (Mech. Eng., Univ. of Michigan, 3632 G.G. Brown Bldg., 2350 Hayward St., Ann Arbor, MI 48105, yxzhai@umich.edu)

The acoustic wave propagation through conventional materials is controlled by two parameters: the mass density and the bulk modulus. It has been shown that two additional parameters called Willis coupling terms are needed in order to describe the dynamics of certain complex media called bianisotropic (Willis) materials. The few bianisotropic (Willis) materials reported so far have experimentally demonstrated the ability of these complex media to control the propagation of sound beyond what is possible with conventional media. For instance, it has been shown that bianisotropic materials could control reflection and transmission coefficients independently and enable high-efficiency sound beam steering devices. In this presentation, we will describe a previously unexplored property of acoustic bianisotropy. Namely, we will present a design of ultra-compact active bianisotropic metamaterials that can serve as excellent sound isolators. In addition, we will discuss the bandwidth and stability of our design. We will show that unlike conventional active sound control devices, active bianisotropic materials are stable even in dynamically changing environments without employing any instability mitigation algorithms.

MONDAY AFTERNOON, 13 MAY 2019 COMBS CHANDLER, 1:00 P.M. TO 5:15 P.M.

Session 1pSC

Speech Communication and Signal Processing in Acoustics: Exploring the Interface Between Linguistic Processing and Talker Recognition

Rachel M. Theodore, Cochair
University of Connecticut, 850 Bolton Road, Unit #1085, Storrs, CT 06269

Tyler K. Perrachione, Cochair
Department of Speech, Language, and Hearing Sciences, Boston University, 635 Commonwealth Ave., Boston, MA 02215

Chair's Introduction—1:00

Invited Papers

1:05

1pSC1. Listener sensitivity to structured phonetic variation. Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

Memory for spoken language is not a veridical representation of experience. Instead, memory reflects integration across our interlocutors' messages, resulting in robust memory for meaning with relatively poor memory for the specific form of the message. This is striking considering that in the process of mapping from speech to meaning, listeners show exquisite sensitivity to the acoustic-phonetic structure of speech. In this talk, I will review selected findings from work in our laboratory that examines listeners' ability to dynamically adapt to structured phonetic variation, focusing on variation associated with talkers' idiolects. These studies examine mechanisms that allow listeners to exploit structured variation for speech perception, voice recognition, and memory of spoken language. Collectively, the results (1) identify principles that govern how listeners modify the mapping to speech sounds to reflect cumulative experience with talkers' phonetic input, (2) show that sensitivity to structured phonetic input facilitates identification of a talker's voice in addition to the linguistic message, and (3) demonstrate that talker identity provides a critical structure for the integration of spoken language in memory. These findings help explicate a theoretical framework that accounts for tension in a linguistic architecture that uses both abstract and instance-specific representational knowledge to guide spoken language processing.

1:25

1pSC2. Recognizing speech in the context of talker variability. Shannon L. Heald and Howard C. Nusbaum (Psych., Univ. of Chicago, 5848 S. University Ave., B406, Chicago, IL 60637, smbowdre@uchicago.edu)

The relationship of the acoustic patterns of speech to phonetic categories varies across talkers. A change in talker therefore requires listeners use this relationship which depends on determining the acoustic-phonetic mapping for a new talker. The work from our group has demonstrated that tuning the perceptual system to the acoustic-phonetic system of a new talker is associated with a redirection of attention and a momentary increased load on working memory. These empirical findings support the hypothesis that speech perception is best thought of as an active cognitive process, where the ambiguity of the acoustic signal determines the mobilization of cognitive resources needed to support accurate perception. In this talk, I will (1) discuss the implication of these results for both cognitive and neural models of speech perception and (2) make the argument that extant theories of speech perception must address recognition in the context of acoustic-linguistic pattern variability, such as talker variability, as it is central to the computational problem of understanding speech perception.

1:45

1pSC3. Building talker knowledge from talker variability. Xin Xie (Univ. of Rochester, 850 Bolton Rd., Storrs, CT 06269, xxie13@ur.rochester.edu)

Listeners reliably extract invariant linguistic category information from speech despite massive cross-talker variability. Existing work shows that talker variability is in part handled by the learning and maintaining of category-specific acoustic distributions towards the statistics in the spoken input from a talker. *When* do listeners maintain information about talkers' speech after we meet them, and *what kind* of information should be maintained in principle? In this talk, I present a computational framework that addresses these questions by linking the statistics of the speech input (*production*) to predictions about *perception*. The results provide constraints on the type of inference underlying talker-related adaptivity during speech perception and have direct implications for current research on talker recognition.

2:05

1pSC4. Specificity and generalization in perceptual adaptation to systematic variation in speech. Lynne C. Nygaard and Christina Tzeng (Dept. of Psych., Emory Univ., Atlanta, GA 30322, lnygaar@emory.edu)

A signature problem in our understanding of spoken language processing is listeners' perceptual constancy in the face of considerable variability in the instantiation of linguistic form. Abundant evidence now suggests that listeners are sensitive to the fine-grained structure of linguistic units that signal differences among talkers and track and adapt to this structure during the perception of speech. This talk will present data from a perceptual learning paradigm addressing both the constraints and flexibility of speech perceptual mechanisms. The aim was to address the degree to which exposure to variation in and expectations about talker- and group-specific attributes influence the degree of specificity and generalization in the perceptual adaptation process. Listeners were exposed to talker-dependent variation in a linguistic form, and expectations about talker characteristics were manipulated. The results suggest that listeners dynamically adapt to and encode systematic changes in the linguistic category structure as a function of expectation and appear to flexibly integrate talker- and group-dependent sources of variation into linguistic representation and processing.

2:25–2:45 Break

2:45

1pSC5. Mutual predictability among speech sounds for talker adaptation and recognition. Eleanor Chodroff (Lang. and Linguistic Sci., Univ. of York, 2016 Sheridan Rd., Evanston, IL 60208, eleanor.chodroff@gmail.com) and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD)

Processes of talker recognition and adaptation rely on a high degree of inter-talker phonetic variability and systematicity, respectively. While superficially in opposition, talker recognition in part depends on adaptation to the talker at hand. In this talk, we present evidence that talker variability is simultaneously extensive and structured within natural classes of speech sounds. In American English, talker mean peak frequencies for [s] span over 3000 Hz, but the variation in [s] is not independent of that in [z]: strong correlations of the talker mean peak frequency, among other phonetic dimensions, are observed between sibilant fricatives. Covariation among speech sounds indicates mutual predictability, such that evidence from one speech sound could be used to refine estimates or make predictions about a second. Listeners indeed demonstrate perceptual knowledge of covariation in generalized adaptation to novel talkers. After exposure to a talker with a relatively high- or low-peak frequency [z], listeners adjusted their [s]-[ʃ] boundary in accordance with the empirical covariation. As talker recognition entails estimation of a talker's phonetic parameters, prior perceptual knowledge of covariation could be used to refine estimation of multiple speech sounds from minimal exposure, thus accelerating processes of talker adaptation and recognition.

3:05

1pSC6. Who's talking and what are they saying: Phonetic cue distributions link speech and talker recognition. Dave F. Kleinschmidt (Psych., Rutgers NB, 152 Freylinguysen Ave., Piscataway, NJ 08854, dave.f.kleinschmidt@gmail.com)

On the one hand, talker variability is one of the fundamental challenges for speech recognition: each talker has their own mapping from linguistic units to sounds, which means that an effective listener must use a different recognition function for each talker. On the other hand, talker variability means that speech is a source of rich information about who the talker is. This dual nature of talker variability means that speech and talker recognition are inextricably linked: knowing something about who is talking makes it easier to understand what they are saying, and knowing something about how someone talks unlocks the rich social meaning of speech. I argue that the concept of a talker's generative model, or the probabilistic distributions of sounds associated with each phonetic/linguistic category, is a useful general purpose conceptual tool for understanding the link between talker variability, speech recognition, and social identity. With such phonetic cue distributions, we can use information theoretic tools to quantify both the extent and structure of talker variability across different phonetic systems and establish in-principle consequences of talker variability for both speech recognition and socio-indexical inferences from speech.

3:25

1pSC7. Asymmetries in phonetic memory across voices and contexts. Meghan Sumner (Dept. of Linguist, Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305-2150, sumner@stanford.edu)

Listeners store highly detailed phonetic representations in memory. These representations are tied to talkers, whereby changing the talker negatively influences performance. This talk focuses on two related issues: (1) memory is not perfectly reliable and (2) the encoding of a single utterance across various voices and contexts is likely to vary. In this talk, I present the result from three experiments that highlight asymmetries that arise when we look the memory for spoken words in the continuous recognition paradigm when talker is a

between-subject variable, when talker is a within subject variable, and when talker is a within-subject variable in a divided attention condition. I show that memory for spoken words depends on talker voice, presenting talker voices together increases the differences found across talkers. These differences are larger again, under divided attention conditions. The phonetic memory for a single word depends on the talker, the social information processed about the talker by the listener, and stereotypes cued by the voices. This work focuses on the memory of words spoken by different talkers and is important in a broader context: Reports of linguistic experiences depend on linguistic memory, and here, we show that memory is both predictably unreliable and easily malleable.

3:45–4:05 Break

Contributed Papers

4:05

1pSC8. Toddlers' sensitivity to talker age in novel word learning: Children are better teachers. Yuanyuan Wang (Otolaryngology-Head and Neck Surgery, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43202, Yuanyuan.Wang@osumc.edu) and Amanda Seidl (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

From early on, young children are sensitive to talker-specific attributes present in the speech signal. For example, infants attend selectively and learn better from their mother's voice than another female voice. Since age influences vocal quality, we asked whether toddlers show similar selective attention and learning from talkers of specific ages. We recorded visual and auditory stimuli from 3 adult males and females and 3 boys and girls. In Experiment 1, 24 toddlers viewed two side-by-side video clips of two talkers reciting a nursery rhyme and heard a soundtrack that matched with either the age or gender of one of the two talkers. The results revealed that children were able to match both vocal age and gender attributes to visual attributes. In Experiment 2, 24 toddlers were tested in a novel word learning task with talkers of different ages and genders. The results revealed that toddlers learned novel words from child talkers but failed to learn from adult talkers. These results suggest that young learners are sensitive to talker age information in speech and are biased towards learning words from younger talkers. We discuss the origins of such a bias.

4:20

1pSC9. Simulated cochlear-implant processing results in major loss of acoustic information regarding differences in talkers' voice qualities. Meisam K. Arjmandi, Hamzeh Ghasemzadeh, and Laura Dilley (Communicative Sci. and Disord., Michigan State Univ., Rm. 211A, 1026 Red Cedar Rd. Oyer Speech & Hearing, East Lansing, MI 48824-1220, khalilar@msu.edu)

The ability to recognize different voice qualities is essential for good talker identification; yet, little is known about how well voice quality cues of talkers are transmitted through the degraded speech signals delivered by cochlear implants (CIs). This study examined how CI speech filtering affects acoustic distinctiveness of individuals with and without voice disorder. Sustained /a/ vowels uttered by speakers with normal or disordered voice were processed using 4, 8, 12, 16, 22, and 32 channel noise-vocoders. The effect of CI processing on the distinctiveness of talkers with normal and disordered voices was measured using the Mahalanobis distance measure on Mel-frequency cepstral coefficients derived from samples across these groups. The analysis confirmed that CI vocoding dramatically degrades acoustic cues in frequency sub-bands that signal abnormal voicing behavior. Remarkable spectral degradation was observed in low- (<2 kHz), mid- (~4-12 kHz), and high-frequency (>12 kHz) bands for simulated conditions compared with unprocessed signals. These findings indicate that CI users likely have almost no ability to distinguish talkers differing in voice qualities. These results highlight challenges that CI users face for recognizing talkers differing in voice quality, due to these users' lack of access to fine-grained spectro-temporal details in voices.

Invited Papers

4:35

1pSC10. How task effects and reading ability relate to talker processing. Sandy Abu El Adas (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., NY 10012, sandyabu@nyu.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Studies show that listeners are better at processing talker information in their native language compared to an unfamiliar language. Several studies have explored two possible sources of this effect by manipulating lexical-semantic information on the one hand and phonological familiarity on the other. To probe these two types of information, researchers have manipulated the stimuli (e.g., phonologically similar or different languages, nonwords) and the listeners (e.g., listeners with reading impairments because reading ability is linked to phonological processing). These prior studies have found that individuals with poorer phonological processing also have poorer talker processing. However, it is unclear whether these individuals show poor performance in talker processing due to task demands related to creating talker representations in long-term memory. To test the question of task effects, we compare performance of individuals with a range of reading abilities on two tasks of talker processing: a discrimination task and an identification task. In addition, to control for both the lexical and phonological properties of the stimuli, stimuli were either words or nonwords and had either high or low phonotactic probability. Our preliminary results suggest that the effect of reading on talker processing depends on the type of task.

4:55

1pSC11. Talker variability, auditory attention, and speech processing efficiency. Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu)

A classic finding in speech perception is that speech is processed more efficiently from a single, continuous talker than from mixed talkers. Given enormous variation in the acoustic realization of speech, it is thought that talker adaptation is necessary to ascertain the mappings between talkers' speech acoustics and listeners' abstract phonological representations. However, a suite of empirical studies

from our laboratory may suggest a predominately attention-based explanation for effects attributed to talker adaptation: using behavioral experiments, neuroimaging, and noninvasive brain stimulation, we have probed how speech processing efficiency under talker variability is affected by temporal, phonological, contextual, and expectational factors. We find that processing benefits from talker continuity are automatic and feedforward, depending on temporal continuity in the source of speech but not the amount of talker-specific phonetic detail or listeners' expectations about source continuity. Correspondingly, processing costs from talker discontinuity occur even when phonetic contrasts are unambiguous across talkers but are insensitive to the magnitude of phonetic variability, amount of preceding exposure to a talker, or top-down expectation about discontinuity. We consider how domain-general models of attention and auditory streaming may parsimoniously account for these differences in the speech processing efficiency between single and mixed talkers.

MONDAY AFTERNOON, 13 MAY 2019 MCCREARY, 1:30 P.M. TO 4:45 P.M.

Session 1pUW

Underwater Acoustics: Target and Radiation by Structures

Steven G. Kargl, Chair
University of Washington, Applied Physics Laboratory, 1013 NE 40th St., Seattle, WA 98105

Contributed Papers

1:30

1pUW1. Long-range detection of a spherical target located near the gas-saturated bottom. Natalie S. Grigorieva (St. Petersburg State Electrotech. Univ., Apt. 7, Krasnoputilovskaya St. 21, St. Petersburg 198152, Russian Federation, nsgrig@natalie.spb.su)

The study is devoted to modeling of the backscattering from a spherical target located near the gas-saturated bottom. Two cases are compared: when the sound speed in sediment is larger and smaller than the sound speed in water (rigid and soft bottom). The bottom is assumed to be a homogeneous attenuating fluid half-space. The transmitter/receiver is located in a homogeneous water half-space. Modeling is performed in the frequency band of 70–90 kHz for distances between the target and transmitter/receiver from 500 m up to 1 km. The spherical scatterer of a radius a is assumed to be acoustically rigid; $0.3 < a < 0.5$ m. For calculating the echo signal in the frequency domain, we have followed the Hackman and Sammelmann's general approach. The arising scattering coefficients of the sphere were evaluated with the use of the steepest descent method. The use of the obtained asymptotic formulas for the scattering coefficients allowed to decrease essentially a number of summands in the formula for the form function of the backscattered acoustic field. [Work supported by Russian Ministry of Educ. and Sci. under Grant 02.G25.31.0149.]

1:45

1pUW2. Canonical cylindrical target acoustic backscattering investigations. Joshua S. Davis (X11, Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, joshua.davis1@navy.mil), Rodolfo Arrieta (X12, Naval Surface Warfare Ctr. Panama City Div., Panama City, FL), and Jermaine L. Kennedy (X11, Naval Surface Warfare Ctr. Panama City Div., Panama City Beach, FL)

In littoral environments where visibility can be low and objects of interest are often either partially or fully buried with respect to the water-sediment interface, the task of identifying unexploded ordinances (UXOs) is a challenging one. Downward looking sonar (DLS) systems have been shown to be capable of penetrating the seafloor of various littoral environments and therefore could be an effective tool for searching for UXOs. However, although DLS's can penetrate the seafloor, classification and identification (C&I) of targets of interest versus clutter are still a technical obstacle of interest. To address this, researchers at the Naval Surface Warfare Center Panama City Division (NSWC PCD) have been working on algorithms that can identify an object based on its acoustic backscatter response. The concept is similar to identifying the material composition of an object with optical spectroscopy but in this case using the acoustic response rather than the optical spectra. Initial investigations, analysis, and algorithm development have focused on the analysis of acoustic backscattering measurements conducted on various targets in the free-field using linear frequency modulated (LFM) interrogation pulses. Experimental measurements are compared with model results derived from a paper by Lecroq [1]. Data/model results as well as a process for potentially aiding automatic target recognition development will be discussed in this presentation. [1] F. Lecroq, L. Fernand, D. Décultot, and G. Maze, *J. Acoust. Soc. Am.* **91**, 1388 (1992).

2:00

1pUW3. Broad bandwidth acoustic backscattering from small fish : Measurements and cylindrical model. Henry M. Manik (Dept. of Marine Sci. and Technol., Bogor Agricultural Univ. (IPB) Indonesia, Marine Ctr. 3 FL Departemen ITK - FPIK IPB Kampus IPB Dramaga Bogor, Bogor 16680, Indonesia, henrymanik@ipb.ac.id)

Acoustic backscattering measurements were conducted of individual ray-finned fishes (*Gnathopogon caerulescens*) at frequencies from 30 to 150 kHz in an acoustic tank laboratory. Acoustic backscatter measurements for dead ray-finned fishes were made versus incidence angle from -30 deg to 30 deg relative to lateral aspect directions. Backscatter spectra from whole fish vary with the fish length and shift to lower frequencies at higher pitching angles. The backscatter spectra from pitching angle differ in both amplitude and positions. A cylindrical model of fish backscatter was developed and compared with the acoustic measurements.

2:15

1pUW4. Scattering of focused beams by spheres: Understanding the high-frequency angular structure. Timothy D. Daniel and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu)

Previous work on scattering of Bessel beams by spheres [P. L. Marston, *J. Acoust. Soc. Am.* **122**, 247–252 (2007)] and recent work expanding linear focused beams in terms of a Bessel beam superposition [T. D. Daniel *et al*., *J. Acoust. Soc. Am.* **144**, 3076–3083 (2018)] provide a framework for exploring the scattering of focused beams by spheres. In analogy with the work done on the scattering of Gaussian beams by spheres [P. L. Marston, *J. Acoust. Soc. Am.* **129**, 1773–1782 (2011)], the angular scattering pattern of a sphere in a focused beam is calculated from a partial wave series with appropriate weighting factors (also known as beam shape coefficients). The high-frequency pattern for rigid spheres is found to have distinctive shadow boundaries in both the forward and backward scattering hemispheres. The sphere is centered on the beam axis. A physical optics model of the scattering reproduces the location and general shape of the shadow boundaries in both scattering hemispheres, confirming the calculated beam shape coefficients for a spherical cap focused source. [Work supported by ONR.]

2:30

1pUW5. Target strength measurements of spherical and wobbly bubbles. Alexandra M. Padilla and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Forest Park Apt. 281, Durham, NH 03824, apadilla@ccom.unh.edu)

Methane gas bubbles released from the seafloor transport gas upwards through the water column and in some cases even to the atmosphere. Researchers exploit the large acoustic impedance contrast between the gas within the bubble and the surrounding water to acoustically estimate the flux of methane in the ocean. Flux estimation employs the use of analytical acoustic scattering models to convert acoustic backscatter measurements of gas bubbles to size estimates. However, these models assume that bubbles are both spherical and that their radius is much smaller than the acoustic wavelength ($ka \ll 1$). Typically, bubbles in the ocean range from 1 to 5 mm in radius are non-spherical in shape and are observed for *ka* values that are greater than 0.1. A controlled shallow water tank (<6 m) experiment was conducted to assess the uncertainty associated with using analytical models that assume bubbles are spherical and that *ka* values are much smaller than one. Small, spherical and large, wobbly gas bubbles (radii ranging from 0.7 to 4.5 mm) were ensonified over a broad range of frequencies (10–250 kHz) in order to observe *ka* values from 0.1 to 5.

2:45

1pUW6. A focused lens for measuring object response to modulated radiation pressure. Timothy D. Daniel (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ivars P. Kirsteins (NUWC, Newport, RI), Ahmad T. Abawi (HLS, Inc., La Jolla, CA), and Philip L. Marston (Phys., WSU, Pullman, WA)

An ultrasonic lens was designed and tested for use in modulated radiation pressure experiments for the purpose of exciting and identifying the low frequency modes of objects in water. The lens was designed to produce a focused beam, and care was taken to minimize the spherical aberration introduced by the lens. This was done by carefully specifying the shape of the lens surface that was then machined using a computer-controlled lathe. The lens was placed in contact with an electrically excited piezoceramic plate. The quality of the focused sound field was tested and found to display properties predicted by simple models of the focal plane wave-field. The lens was then used to drive low-frequency modes of elastic objects in water using modulated ultrasonic radiation pressure in a way previously summarized involving a commercially made focused transducer [T. D. Daniel *et al*., *J. Acoust. Soc. Am.* **140**, 3123 (2016)]. [Work supported by ONR.]

3:00–3:15 Break

3:15

1pUW7. Verification and validation of a coupled elasto-acoustic-damage system. Jonathan Pitt (AOE, Virginia Tech, 900 N. Glebe Rd., Arlington, VA 22203, jspitt@vt.edu)

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

3:30

1pUW8. Sound scattering from targets in shallow water by finite element method combined with normal-mode method. Jinyu Li, Dejiang Shang, Jinpeng Liu, and Chao Zhang (Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin, Hei Longjiang 150001, China, jenny.mamie@163.com)

A hybrid method to calculate the sound scattering field of the target in waveguide is presented, combining the theory of the normal mode with the finite element method (FEM). Although FEM used only can provide accurate results, it is deeply restricted by the property of the computers and does not have a satisfying computing speed, for its calculation is closely related to the size and complexity of the target. The hybrid method presented in this paper can calculate the sound scattering field of more arbitrary and more complicated target over farther ranges in oceanic waveguide at low frequencies. First, separate the sound field into two parts, domain A, containing the target in the nearer range, and domain B meaning the rest domain of the field. In domain A, the sound field was calculated via FEM. Then, apply the normal mode theory to forecast the sound scattering field over farther ranges, using the pressure data on the intersection area between domains A and B obtained by FEM. The method has been used to calculate the sound scattering field of a rigid sphere in waveguide and based on the results, it is demonstrated that this hybrid method is accurate and efficient.

3:45

1pUW9. The acoustical characteristics of a metal ball entering into water. Dajing Shang and Qi Li (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Nangang District Nantong St. No. 145., Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn)

The sound of objects into the water is very important to the detection and identification of underwater targets. The Metal ball into the water will produce an unique sound signal, which can be used for monitoring the target and evaluating the performance of the sonar. The main influence terms of the radiation noise of the metal ball into the water are analyzed in this paper, and the noise of the metal ball entering into water consists of initial impact sound and bubble sound, and the characteristics of the initial impact sound and the bubble sound are summarized. A set of metal ball entering the water equipment was set up to make the metal ball enter the water at different depths and angles, and the test was finished in the pool of the Harbin Engineering University Underwater Department. A special acoustic signal during the quiet area is discovered, which is related to the size of metal ball and the speed of the metal ball entering water. The sound of the metal entering water was measured, and the relationship between the sound characteristics of the metal ball entering water and the metal ball size, the velocity, and the angle of metal entering water was obtained by processing the measuring data.

4:00

1pUW10. The classification of spherical shells with varying thickness-to-radius ratios based on the auditory perceptive features. Xiukun Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China) and Yushuang Wu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 150001, China, wuyushuang1009@hotmail.com)

The detection and classification of underwater objects are an important part of the coast and harbor security. For silent targets, active sonar is always employed, and then, the information contained in the scattering echo is extracted and applied. Previous studies are aimed at targets with varied shapes, and the classification of objects with similar shapes has yet to be studied comprehensively. To address this issue, the classification of spherical shells with varying thickness-to-radius ratios (TRRs) is studied. The numerical results indicate that spherical shells can be roughly divided into three classes (extremely thin, thin, and thick) according to their scattering components. In this work, the scattering features, especially auditory perceptive features for spherical shells with different TRRs, are extracted and analyzed. Then, the features are imputed into a support vector machine classifier. The simulation results of the recognition rates under different features and signal-to-noise ratios will be present and analyzed. This study provides a novel concept for the classification of silent underwater targets.

4:15

1pUW11. The research of passive detection parameter measurement based on vector sensor. Juan Hui, Xianzhong Bu (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), Anbang Zhao (Underwater Acoust., Harbin Eng. Univ., Harbin, HLJ, China), Dayu Wang, and Jin Li (The 54th Res. Inst. of China Electronics Technol. Group Corp., Shijiazhuang, China)

As one of the indispensable technologies for underwater target detecting, underwater acoustic passive detection technology has always been a hot spot in marine system research. A vector sensor can detect underwater targets by combination with the pressure and vibration velocity information; vector sensor array can obtain higher gain than pressure array with the same number hydrophones. Vector sensor can reduce the length of the underwater sensor array and can detect underwater targets in the case of low signal to noise ratio. This is the goal that many researchers strive to solve for many years; therefore, vector sensor passive detection technology has far-reaching scientific research value. Through the verification of the simulation experiment, the azimuth estimation technique and the adaptive frequency estimation technique are feasible. By estimating the azimuth sequence and frequency sequence of the target radiation signal, the relevant information of the target can be obtained, so it is of great significance in the research of underwater acoustic detection technology.

4:30

1pUW12. Acoustic radiation from elastic structures in shallow water by finite element method combined with normal modes. Buchao An, Dejiang Shang, Chao Zhang, and Yan Xiao (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, an_buchao@foxmail.com)

Acoustic radiation problem of elastic structures in shallow water has not been solved effectively. Finite Element Method (FEM) is not suitable for higher frequency and large range problems. In principle, the Combined Wave Superposition Method can deal with such problems. However, it is hard to optimize a large number of virtual sources. For low and middle frequency bands, we propose a new method combining FEM and normal modes. FEM is used to calculate the near range acoustic field of the structural source, then the eigen function expansion is performed on the field, and the coefficients of modes can be obtained by using the orthogonality of the eigen function. Therefore, we can calculate the acoustic field at any range. This method avoids the process of solving inverse matrix of a large complex matrix, which is an essential step in wave superposition method, and is very efficient to calculate the far field. To show the efficiency and accuracy of the method, the simulation results of various structural sources and waveguide models are compared with that of using FEM directly.

Additional registration fee required to attend this lecture.

MONDAY AFTERNOON, 13 MAY 2019 CLEMENTS, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Computational Methods for Describing Acoustic Propagation in Forests

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Computational methods for describing acoustic propagation in forests. Michelle E. Swearingen (U.S. Army ERDC/CERL, Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil)

Acoustic propagation through stands of trees, both large and small, can be markedly different from propagation through an open environment. Ground properties, meteorology, and significant scattering effects all play a role. An understanding of these effects, and computational methods used to describe them, can be utilized in applications ranging from noise mitigation to wildlife communication. This tutorial explores the myriad of computational methods that have been developed to describe acoustic propagation in forested environments. Beginning with the early simple empirical models describing attenuation, the presentation then narrows the field of view to contributions of individual components within a forest. Models for describing scattering by trunks and foliage, both individually and as ensembles, are presented. Next, the integration of these individual components into the fully coupled system that includes ground properties and meteorology, within computational methods such as the parabolic equation (PE) and finite-difference, time-domain (FDTD) methods are shown. Finally, examples are presented showing how computational methods for forest acoustics can be used for evaluating noise mitigation strategies and wildlife studies.

MONDAY EVENING, 13 MAY 2019 WILSON ROOM, 5:00 P.M. TO 6:15 P.M.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

J. T. Nelson, Chair ASC S2
Wilson Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

R. J. Peppin, Vice Chair ASC S2
5012 Macon Road, Rockville, MD 20852

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.