Session 2aAA

Architectural Acoustics and Musical Acoustics: Performance Spaces for Modern Music

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

Bruce Olson, Cochair
Olson Sound Design LLC, 8717 Humbolt Ave. N., Brooklyn Park, MN 55444

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. Historical evolution of sound diffusion in spaces for modern music. Peter D'Antonio, Jeffrey Madison (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), and Trevor J. Cox (Acoust. Eng., Univ. of Salford, Salford, United Kingdom)

This presentation will review the historical evolution of the use of sound diffusors, from their initial use in recording control rooms to include almost all spaces for the performance, recording, and audition of music. Shortly after the invention of the reflection phase grating diffusor by Manfred Schroeder in 1973, the quadratic residue diffusor was installed in Michael Fowler Hall, New Zealand, Underground Sound recording studios in Largo, MD, and in the Oak Ridge Boy's Acorn Studio in Henderson, TN. The initial acceptance of these new surfaces led to installation in hundreds of recording control rooms and live rooms, home theaters, stages, auditoria, and worship spaces. The design of sound diffusors was expanded to use optimization algorithms created by Cox and D'Antonio to include decorative shapes to complement the original phase grating surfaces, thus opening their use in all architectural acoustic spaces. Following a brief review of the design theory options, many applications over three decades, in a wide range of venues will be presented. A speculation on what the future holds, based on evolving diffusive designs and advanced manufacturing methods, will also be presented.

8:25

2aAA2. Taming reverberation in an outdoor amphitheatre—The Ford. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The historic Ford Amphitheatre in Hollywood, CA, recently completed an overall renovation and expansion. The centerpiece is the 1,200-seat outdoor amphitheater, which features a dramatic arroyo backdrop. The side walls, rear wall, towers, and house flooring were built of concrete in 1931 after a brush fire destroyed the original 1920 wood structure. Several sound-absorptive treatments had been applied to the walls over the years previous to our involvement, but excessive reverberation and some anomalous reflections remained. The new design features an expanded "sound wall" that helps to mitigate highway noise while providing optimal lighting and control positions, and offering an opportunity to improve the acoustical treatment scheme. The result is a unique installation of decorative perforated-metal panels that tame the reverberation, especially in the low frequencies. Assorted design challenges, apparent arroyo hillside contributions, and the resultant reverberation will be discussed.

8:45

2aAA3. Active Acoustics at the Appel Room, Jazz at Lincoln Center. Tom Wetmore (Columbia Univ., New York City, NY) and Steve Ellison (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com)

The Appel Room (formerly Allen Room) is a 483-seat venue used primarily for jazz performances in the Jazz at Lincoln Center complex in New York City. In 2013 the Constellation active acoustic system by Meyer Sound was incorporated into the room's architecture. The system allows a wide range of acoustic conditions to be set for any performance. In the first four years of operation, a variety of solo and ensemble jazz artists have performed in this venue. This paper will describe the room, the system, and the results over the first four years of operation with a variety of musicians.

9:05

2aAA4. Amplified rehearsal and performance spaces on a community college campus. Sam Ortallono (Visual Performing Arts, Lee College, 711 W. Texas Ave., Baytown, TX 77522, sortallono@lee.edu)

Amplified rehearsal and performance spaces on a community college campus. Lee college in Baytown Texas has several rehearsal and performance spaces available to presenters and performers. Many of these spaces have access to sound reinforcement which interacts with the structure. In this presentation, we compare and contrast the amplified systems and spaces, including variable acoustic treatment. Each space has a dedicated purpose that determines the equipment.

9:25

2aAA5. It's All About That Bass, Design of a Showroom for Music, Comedy, and Theater. Bruce C. Olson (Olson Sound Design LLC, 8717 Humboldt Ave. North, Brooklyn Park, MN 55444, Bruce.Olson@afmg.eu) and Ana M. Jaramillo (AFMG Services North America LLC, Brooklyn Park, MN)

An 1800 seat casino showroom was designed to provide "state-of-the-art" acoustics for superstar country and pop performers, headlining comedians, and big production Broadway musicals. Challenges included a very wide stage, a curved back wall and a client requirement for hidden loudspeakers. This paper will walk you through how these challenges were met while maintaining the high-class look and feel of the room. We will also show the evolution of the sound system design from the initial system through successive upgrades.

9:45–10:00 Break

10:00

2aAA6. Design of halls for the enjoyment of American music. Richard Talaske (TALASKE | Sound Thinking, 1033 South Boulevard, Oak Park, IL 60302, rick@talaske.com)

Designing for contemporary music performance is more involved than achieving the accepted mid-frequency reverberation time within the performance space. While accomplishing the correct degree of reverberance is essential, this alone is not a sufficient condition for creating an acoustic environment suitable for the enjoyment of jazz, blues, folk, rock, computer, world, or other non-symphonic or non-choral genres of music. Control of low-pitched sound, achieving moderate running liveliness, and creating a microphone-friendly acoustic environment are also essential design goals. As usual, the acoustic goals should be accomplished while developing a welcoming room design which offers intimacy between performers and patrons and a communal audience setting. These acoustical and societal goals are discussed in relation to completed projects such as the Ellis Marsalis Center for Music in NOLA, The Sidney Harman Center in DC, and the Old Town School of Folk Music in Chicago.

10:20

2aAA7. The Great Outdoors— The unique challenges imposed by the outdoor environment for both sound reinforcement, as well as acoustical enhancement, of acoustical and amplified music. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com) and Jonathan Laney (d&b audiotechnik, Ashville, NC)

Outdoor performance venues epitomize the definition of a multi-purpose performing arts facility. However, they differ in many ways from their indoor counterparts. They can be larger than many arenas, but lack the architecture that both provides the aural cues of a large volume, and in addition, provides insulation from intrusive noise. The programming is often more diverse. One day the venue could host a three or four act amplified concert—the following day a symphony orchestra, or a week long music festival. Weather is also an enormous unknown factor. We will discuss the changing nature of both the programming, as well as listeners expectations, at these venues. We will also discuss the technologies employed at several outdoor venues that enable them to meet the expectations of the musicians as well as the audience.

10:40

2aAA8. A 240-Seat Recital Space for Amplified Performance. Damian Doria (Stages Consultants LLC, 75 Feather Ln., Guilford, CT 06437-4907, damianjdoria@gmail.com) and Dave Kotch (Criterion Acoust., Jersey City, NJ)

A new 240-seat recital space completed in 2017 for a Canadian University with a Bachelor of Music program teaching jazz, rock, pop, metal, blues, soul, county, electronica, and hip hop. The architectural design brief required a room that could accommodate the diverse programming of student recitals and other music sessions in a modern an aesthetically satisfying interior design. The concept for the room was quickly adopted by the Owner and Design Team, but included several room acoustics challenges. This paper will discuss the acoustical accommodation of activities in the recital space, along with integration of the house sound system, and a chronicle of the design, value-engineering, construction, and final commissioning process.

11:00

2aAA9. Ninety never sounded so good: Guiding an historical auditorium into the 21st century. Brandon Cudequest and David A. Conant (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, bcudequest@gmail.com)

The Sacramento War Memorial Auditorium opened in 1927 with a screening of the silent film, "Old Ironsides." Over its ninety-year career, the 3500-seat auditorium has hosted everything from Shriners conventions to Shinedown concerts, Ringling Brothers to Rolling Stones, and many more. In its current state, the room is excessively reverberant for amplified music. Heating, ventilation, and air conditioning systems were upgraded during the 1990s, albeit without apparent acoustical input. Street traffic noise is clearly transmitted via acoustically weak windows and stage house doors. McKay Conant Hoover is currently engaged in an acoustical, audiovisual, and

theatrical renovation of this historic venue. Importantly, the Auditorium must temporarily accommodate Sacramento's touring Broadway shows, Philharmonic, choral, and ballet performances while their main venue, The Community Center Theater, undergoes a several year-long renovation. This paper will discuss acoustical challenges faced, current findings, and how factors such as ill-conceived pit lifts, historical finishes, limited storage, and lead paint influenced our recommendations.

11:20

2aAA10. Worship space acoustics and architecture for contemporary services with modern music. Robert C. Coffeen (None, R. C. Coffeen Consultant in Acoust., Lawrence, KS 66047, bob@rccoffeen.net)

In relatively recent time, some religious worship facilities are presenting a contemporary worship service with music that can be considered modern and different from the more traditional music produced by an organ, piano, groups that employ instruments as used by a symphony orchestra, and by a choir. Modern music for a contemporary service is typically produced by a band using electric keyboards, electric and acoustic guitars, drums, and singing by a solo vocalist or by singing groups with all of the music electro-acoustically amplified. Thus, halls, auditoriums, and sanctuaries for contemporary worship services must generally be less reverberant than a facility for traditional music, and the sound reinforcement system must properly handle relatively high level music and vocals. It also seems that architecture for worship facilities where contemporary services will be held has changed from more traditional architecture to more informal designs with exposed ceiling structure, movable chairs, etc. This also seems to produce HVAC systems with higher noise due to exposed supply air ducts, short return air paths, and roof-top air handlers. Examples of and acoustical data for worship spaces with contemporary services will be presented.

TUESDAY MORNING, 5 DECEMBER 2017 — SALON E, 8:35 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics, Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography: In Memory of George Ioup: Acoustics in the Gulf of Mexico I

Natalia Sidorovskaia, Cochair
Physics, UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210

David K. Mellinger, Cochair
Coop. Inst. for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365

Chair's Introduction—8:35

Invited Papers

8:40

2aAB1. George Ioup's contribution to the Gulf of Mexico acoustic research: paving the path into the future. Natalia Sidorovskaia (Phys., UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210, nas@louisiana.edu) and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

By the end of the 1990s, researchers and regulators recognized the need for understanding how anthropogenic activities impact cetacean's populations in the Gulf of Mexico. In 2000, George Ioup was one of the founders of the Littoral Acoustic Demonstration Center (LADC), a consortium of Gulf Coast scientists, with the long-term goal of studying the anthropogenic soundscapes of the Gulf of Mexico and their impact on marine mammals. In 2001, LADC was the first team to collect long-term acoustic data rich in sperm whale phonations from bottom-moored autonomous buoys, technology developed by NAVOCEANO and adapted for LADC needs. The first step in establishing a baseline database was taken. At the same time, George sparked the interest of bioacousticians with ideas on how to employ the differences in sperm whale phonations to identify individuals similar to how humans recognize voices. LADC endeavors continued through designing acoustic surveys to characterize the soundscapes of seismic exploration arrays and being the first team to record beaked whales in the Gulf of Mexico in 2007. Recent advances in understanding marine mammal habitats, studying the oil spill impact on these animals, and introducing new acoustic technologies would not be possible without the seminal work of George Ioup.

9:00

2aAB2. A comparison of automated and manual techniques for acoustically identifying individual sperm whales with changing aspect. Christopher Tiemann (R2Sonic LLC, 5307 Industrial Oaks Blvd., Ste. 120, Austin, TX 78735, chris@r2sonic.com)

As shown by his extensive history of research in the Gulf of Mexico, Dr. George Ioup had a fascination with the marine mammals that lived there. In particular, he questioned whether individual whales of a given species could identify each other acoustically, and by extension, whether humans could do the same. George was an early proponent in the community of advancing automated methods for discriminating individual odontocetes by analyzing the clicks they make, a problem that grows considerably more difficult when multiple animals are clicking simultaneously. I had the good fortune of being able to explore this subject with George in an independent research project that may not have been widely reported but was important and enjoyable for us nonetheless. George realized quickly that conventional methods for automated grouping of clicks from the same click train were failing because there was no guarantee of click structure consistency as an animal changed aspect relative to a receiver. An automated method for grouping sperm whale clicks based on a cross correlation method originally applied to dolphin clicks provided the needed flexibility to follow an evolving click shape as shown by the close match between George's automated results and my manually generated groupings.

9:20

2aAB3. Modeling as a complementary tool to acoustic data for understanding the impact of environmental disasters on marine mammals. Azmy S. Ackleh, Ross Chiquet, Tingting Tang, Amy Veprauskas (Mathematics, Univ. of Louisiana at Lafayette, P.O. Box 41010, Lafayette, LA 70504, ackleh@louisiana.edu), Hal Cawell (Univ. of Amsterdam, Amsterdam, Netherlands), Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), and Baoling Ma (Millersville Univ., Millersville, PA)

This study is focused on how environmental disasters, such as the Deepwater Horizon oil rig explosion in 2010, affect the dynamics of marine mammal populations, particularly sperm whales and beaked whales, in the Northern Gulf of Mexico. We briefly describe how modeling techniques are used to estimate densities of marine mammals using passive acoustic data. We then develop a matrix model to examine the possible long-term effects of a disaster. We consider cases in which the effects of a disturbance result in reductions in either survival (lethal impacts) or fecundity (sublethal impacts). This model, combined with demographic stochasticity, allows us to study the long-term recovery process following an environmental disaster. In particular, recovery probabilities and recovery times of the population are computed, and formulas are derived to compute the sensitivity of the recovery time to changes in properties of the population or the environmental disturbance. We then extend the modeling methodology to consider how marine mammals may be affected by the response of their prey population to a disturbance. Our analysis highlights the difficulty of projecting impacts and recovery in the absence of detailed demographic data, and the value of population models in exploring scenarios and identifying important processes and general relationships. [This research was made possible in part by a grant from The Gulf of Mexico Research Initiative.]

9:40

2aAB4. Passive acoustic monitoring in the Northern Gulf of Mexico using ocean gliders. David K. Mellinger, Sharon L. Nieukirk, Sara L. Heimlich, Selene Fregosi (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Elizabeth T. Küsel, Martin Siderius (Northwest Electromagnetics and Acoust. Res. Lab., Portland State Univ., Portland, OR), and Natalia Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

Although George Ioup did not use ocean gliders for passive acoustic monitoring, he recognized their value as platforms for PAM and encouraged others to use them. They function well as PAM platforms because (1) they move slowly, minimizing flow noise; (2) they have no propeller or continuously running machinery, minimizing motor noise; (3) they collect acoustic data nearly continuously; (4) they traverse the upper water column every few hours, measuring temperature and salinity as needed for calculating sound speed profiles and enabling accurate modeling of long-range acoustic propagation; (5) they can cover hundreds to thousands of kilometers in distance during a deployment, enabling them to monitor a large area and/or repeatedly monitor a smaller area; and (6) some models can dive to 1000 m, the depth at which some deep-diving cetaceans—sperm and beaked whales, frequent targets of PAM operations—forage and vocalize. Two models of gliders equipped with passive acoustic recording systems were deployed in the Northern Gulf of Mexico in the summers of 2015 and 2017 to study cetacean occurrence and behavior. Here, we summarize the virtues and hazards of glider PAM, and describe acoustic detection and classification of cetaceans in these recordings. [Research supported by GoMRI.]

10:00

2aAB5. Echolocation for restoration: Odontocete monitoring in the Gulf of Mexico. Kaitlin E. Frasier, Rebecca Cohen, Jennifer S. Trickey, Sean M. Wiggins, Alba Solsona Berga (Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037, kfrasier@ucsd.edu), Melissa Soldevilla, Lance Garrison (Protected Resources Div., NOAA SEFSC, Miami, FL), Simone Baumann-Pickering, and John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA)

In the late 1990s, George E. Ioup began studying echolocation clicks as a means of understanding marine mammals in the Gulf of Mexico (GOM). He also led one of the few research programs focused on pelagic species in this chronically impacted region in the years preceding the Deepwater Horizon oil spill. Today, passive acoustic monitoring (PAM) is one of the primary tools used to study the nearly 20 pelagic odontocete species found in the GOM, including sperm whales, beaked whales, dolphins, and *Kogia* species. Since 2010, PAM devices have been deployed nearly continuously in the region, driven by an urgent need to understand the long-term effects of both acute and chronic anthropogenic impacts on GOM marine mammal populations. Recent advances fueled by robust, reliable PAM technologies include the development of multi-year timeseries documenting changes in species densities across continental shelf and slope habitats, differentiating GOM odontocete species based on echolocation click properties, and leveraging long-term datasets to understand the influence of habitat variability on offshore species distributions. These capabilities are proving PAM is indispensable as an observation method for damage assessment, decision support, and restoration activities for marine mammals, especially inaccessible pelagic populations.

10:20–10:40 Break

10:40

2aAB6. Airborne electromagnetic profiling in the Gulf of Mexico. Kenneth W. Holladay (Mathematics Dept., Univ. of New Orleans, New Orleans, LA 70148, khollada@uno.edu)

First, I will give an overview of the research on Airborne Electromagnetic Profiling (AEM) by the University of New Orleans (UNO) group that included George Ioup. The standard techniques of error correction, calibration, modeling, inverse methods, interpolation, and smoothing were used to convert raw profiler data into meaningful data and useful products. As a part of a NASA EPSCoR project, the group applied layer models to AEM data to measure shallow water bathymetry and water column stratification in the Gulf of Mexico and coastal near surface geomorphology in the Barataria basin of Louisiana. In the second part of the talk, I will sketch the history of the UNO Engineering and Applied Science doctoral program and George Ioup's large role in it.

11:00

2aAB7. Lognormal bubble size distributions. Jerald W. Caruthers (29 Pecan Dr., Long Beach, MS 39560, jcaruth123@gmail.com)

In the study of oceanic bubbles based on empirical acoustics, the distributions among the sizes are typically represented by power laws with negative slopes as convenient descriptors of data in log/log format. However, power laws do not address the fact that as bubble sizes approach zero their numbers must approach zero. Multiple power laws, including a horizontal segment preceded by a positive power-law segment and followed by the expected negative power-law segment, have been used to recognize and mitigate this problem. Although not universally adopted as yet, several acoustic researchers have suggested that at least some oceanic bubble distributions are more appropriately represented by lognormal distributions. In 1941, A. N. Kolmogorov used the 1922 work of L. R. Richardson concerning a stochastic downward cascade of random sizes of turbulent vortices that asymptotically result in lognormal distributions of vortices. This current paper uses such a cascade of vortices to begin a downward cascade of bubbles sizes causing cascading shear forces on large bubbles that were created by breaking waves. In combination with this decreasing effect of turbulent shear on these fragmenting bubbles, the downward cascade of bubbles sizes overlaps and continues with a strengthened partial-pressure effect on ever increasing surface tension caused by their diminishing sizes. Issues associated with this approach, such as summing lognormal generators and intermittency, will be discussed.

11:20

2aAB8. In memory of George E. Ioup: Founder of the University of New Orleans physics program at the Stennis Space Center. Stanley A. Chin-Bing and Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, chinbing@att.net)

In the late 1970s, two U.S. Navy organizations were formed and located in an area now known as the Stennis Space Center, Mississippi, located nearly 50 miles East of the University of New Orleans (UNO). These two organizations employed nearly 1500 scientists and technicians who needed advanced training in physics, specifically courses involving acoustics and signal processing. In 1982, Professor George E. Ioup took the initiative to have UNO develop and teach these courses on-site at the Stennis Space Center. In the following 33 years, nearly 20 different graduate level courses were developed and taught multiple times at the Stennis Space Center. It was possible to take all the necessary courses needed for the Masters and Ph.D. degrees on-site, while maintaining full-time employment. Under Professor Ioup's leadership, several dozen Navy scientists received advanced degrees from the University of New Orleans, and many more received specific training that enhanced their professional careers. This presentation will highlight the dedicated efforts and successes of George Ioup in creating the UNO physics program at the Stennis Space Center, MS.

11:40

2aAB9. Relationship between head size and biosonar transmit and receive beams in odontocetes. Whitlow Au (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu)

The unique shape of a dolphin head, the different specie specific shape and the internal head structure suggest a very complex propagation mechanism for the biosonar signals to travel from the phonic lips into the water. Despite these factors, a circular planar aperture of the appropriate diameter can produce a transmission beam that resembles the corresponding beam of a dolphin. In the similar manner, the reception process is also very complex and not completely understood. Once again, despite the complexity involving the reception of biosonar echoes, a simple circular planar aperture of the right diameter can have a receiving beam that approximate or resemble the receiving beam of odontocetes. For both transmission and reception of broadband biosonar signals, the size of the head compared to the wavelength of the signal will determine the degree of directionality of the beams in odontocetes in a similar manner as a planar transducer. Beam pattern data from *Tursiops truncates*, *Delphinapterus leucas*, *Pseudorca crassidens*, and *Phocoena phocoena* will be used to demonstrate the relationship between head size and directionality for the transmit signal. Receiving beam pattern data from *Tursiops truncates* and *Phocoena phocoena* will be used in a similar manner as for the transmit beam.

Session 2aBA

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

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

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

Chair's Introduction—8:25

Invited Papers

8:30

2aBA1. Parity-time synthetic phononic media. Johan Christensen (UC3M Madrid, UC3M, Madrid 28935, Spain, johan.christensen@uc3m.es)

Classical systems containing cleverly devised combinations of loss and gain elements constitute extremely rich building units that can mimic non-Hermitian properties, which conventionally are attainable in quantum mechanics only. Parity-time (PT) symmetric media, also referred to as synthetic media, have been devised in many optical systems with the ground breaking potential to create non-reciprocal structures and one-way cloaks of invisibility. Here, we demonstrate a feasible approach for the case of elasticity where the most important ingredients within synthetic materials, loss and gain, are achieved through electrically biased piezoelectric semiconductors [1]. We study first how wave attenuation and amplification can be tuned, and when combined, can give rise to a mechanical PT synthetic media with unidirectional suppressed reflectance, a feature directly applicable to evading sonar detection [2]. [1] J. Christensen, M. Willatzen, V. R. Velasco, and M.-H. Lu, *Phys. Rev. Lett.* **116**, 207601 (2016). [2] S. A. Cummer, J. Christensen, and A. Alu, Nature Rev. Mater. **1**, Article number: 16001 (2016).

8:50

2aBA2. Beyond the single-scattering assumption for analysis of diffuse ultrasonic scattering experiments. Joseph A. Turner and Nathanial Matz (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

Characterization of diffuse ultrasonic scattering is challenging because accurate models require assumptions about the degree of scattering. Most previous research has focused on the single-scattering regime in which rays are assumed to scatter only once before detection. In this presentation, scattering effects beyond single scattering are examined with a focus on polycrystalline materials. The contribution of the second scattering within the response is quantified with respect to measurement parameters and sample properties. The results show that single-scattering models are appropriate for weakly scattering materials, such as aluminum, for a wide range of experiments and grain sizes. However, stronger scattering materials are predicted to have significant components beyond single-scattering for certain measurement parameters even in the Rayleigh regime. Experimental results for a steel alloy are used to verify model predictions. The experimental work shows the domain for which the doubly-scattered response becomes significant as well as the limitation at which the double scattering model is no longer applicable. The results can be used to predict the applicable frequency range for each level of scattering for a given experimental configuration. Finally, the impact of the higher-order scattering is discussed with respect to the detection of defects within a heterogeneous medium.

9:10

2aBA3. Acoustic propagation in a fractal network of scatterers. Vincent Gibiat, Etienne Bertaud (Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr), MArie-Fraise PONGE (I2M, Bordeaux Univ., Bordeaux, France), and Xavier Jacob (Toulouse Univ., Toulouse, France)

Fractal networks, built on Cantor set, Fibonacci series or Sierpinsky sets are characteristics of a structural organization between periodic and random. On the other hand, they have been proved to be a good descriptor of irregular systems as the well known description of Brittany coasts. Such systems can be seen as irregular boundaries, as in the case of irregular coasts, while the wave equations are still usable inside theses boundaries. Along the boundaries, wave localization is possible. Another case is that where boundaries are regular while the organization of the material where waves are propagating carries the fractality. Wave propagation on such systems can be considered through the aspect of multiple scattering, and on 1D systems as well on 2D systems built on Cantor or sierpinski set, it is possible to show

the existence of localized modes. On the contrary to the previous case, localization arises not along the boundaries but inside the propagating medium. Less common organization of scatterers, for instance, those built on Fibonacci series can also be compared to periodic organizations as well as random ones. Various examples will be shown on experimental material as on theoretical or numerical examples.

9:30

2aBA4. Relationship between ultrasound scattering and acoustic impedance maps in sparse and dense random media. Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org), Kazuki Tamura (Graduate School of Eng., Chiba Univ., Chiba, Japan), Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York City, NY), Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), and Emilie Franceschini (Laboratoire de Mécanique et d'Acoustique, Aix Marseille Université, CNRS, Centrale Marseille, Marseille, France)

Quantitative modeling of ultrasound scattering from soft tissues has been used extensively to characterize soft tissues. In this approach, tissues are typically modeled as a random medium containing scatterers of specific shapes, acoustic properties, and spatial arrangements. Under plane-wave insonification and assuming weak scattering (i.e., Born approximation), the backscattered coefficient (BSC) of such a random medium is fully described by the power spectrum of its three-dimensional (3D) impedance map (ZM). A two-dimensional (2D) ZM can be obtained by scanning acoustic microscopy (SAM) applied to thin tissue sections using very high-frequency ultrasound (>100 MHz). Under isotropic assumptions, 2DZMs can predict the BSC accurately; nevertheless, in the case of dense media, where the locations of the scatterers can be correlated, some of the theoretical assumptions fail, which requires introduction of the structure-factor model (SFM). Using experimental and simulated data, this presentation will review computation of the BSC from ultrasound measurements, the working principles of SAM, the use of 3DZMs and 2DZMs to predict the BSC, SFM estimation from 2DZMs and 3DZMs, and the use of SFM for BSC computation and tissue characterization.

9:50

2aBA5. Simulation of guided waves in layered fluid/viscoelastic/poroelastic media using semi-analytical finite element method. Vu-Hieu Nguyen (Univ. of Paris-Est, 61 Ave. du général de Gaulle, Creteil 94010, France, vu-hieu.nguyen@u-pec.fr)

Understanding of the ultrasound transmission in functionally graded structures materials is of great interest in many engineering applications such as geophysics, biomedical diagnostics, aircraft, and automobile. This paper will present a computational method and its implementation procedure to study the wave propagation problem in multilayer structures made from a combination of fluid, anisotropic viscoelastic, and poroelastic materials. The poroelastic material is described by using the Biot theory. The developed approach is based on the Semi-analytical Finite Element Method (SAFE), which only requires the discretization of the cross-section of the structure. For the finite element solver, high-order spectral element has been used, showing a significant improvement of the computational efficiency compared to the use of conventional high-order elements. Numeral validation in both time and frequency domains show that the proposed approach is efficient to investigate the transient response as well as the dispersion of layered media. Some results in the context of quantitative ultrasound axial transmission techniques for assessing properties of cortical long bones will also be presented.

10:10–10:25 Break

10:25

2aBA6. Nonlinear coda wave interferometry: Characterizing damage in complex solids. Vincent Tournat (LAUM, CNRS, Université du Maine, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr)

In this talk, we report results on the nonlinear interactions of ultrasonic coda waves, in reverberating or multiple scattering mesoscopic solid media. Using the method of coda wave interferometry (CWI), we analyze the effect of mixing a coda wave with an additional lower frequency pump wave. The extracted CWI parameters, known to be highly sensitive to small geometric or elastic modifications of the tested medium, are shown to be pump-amplitude dependent and to capture finely the results of the nonlinear interactions. Although nonlinear self-action effects with coda waves have been reported in unconsolidated granular media, they are difficult to implement in cracked solids or concrete. Instead, the reported nonlinear CWI class of methods (NCWI) shows robustness, a high sensitivity, and has been applied successfully to various complex media and structures. We show through several examples on « model » media (cracked glass plates) and on concrete structures, that NCWI can be useful for the nondestructive evaluation of complex solids that are strongly scattering at usual probing frequencies. Preliminary results and prospects in nonlinear elastic properties imaging and quantitative evaluation with NCWI are discussed.

10:45

2aBA7. Propagation of ultrasonic waves in complex media: Investigation of coda wave technique and traditional polar and C-scan imaging. Nico F. Declercq, Lynda Chehami (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France, nico.declercq@me.gatech.edu), Pascal Pomarède (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France), Othmane Ez-Zahraouy, Esam Ahmed Mohammed (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, Metz, France), and Fodil Meraghni (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France)

A discussion of nondestructive techniques is presented for the investigation of complex media, with a focus on composite samples. Traditionally, one applies ultrasonic C-scans, or polar scans, which are easy to implement and to interpret. However, in many realistic cases, it is important to use more sophisticated approaches as C-scans often do not reveal any useful information. Typically, the early part of received signals is used to extract information, whereas the later part is considered either as noise or as a useless coda wave as in musical acoustics. Nevertheless, it appears that the coda part carries useful information about the medium, and therefore, it is important to explore techniques to extract that information. In addition, it turns out that the coda is very sensitive to material properties and damage as those sound waves interact longer with the material than early arrival waves. First, earlier results will be shown which compare experimental polar scans with numerical simulations, then, for the same samples, coda wave results will be presented to show the effect of damage on the composite samples. The main damage indicator is the change in relative wave velocity which is caused by the damage.

Contributed Paper

11:05

2aBA8. Ultrasound tomography of materials with high sound speed contrast. Timothe Falardeau and Pierre Belanger (Mech. Eng., Ecole de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C1K3, Canada, timothe.falardeau.1@ens.etsmtl.ca)

Non-destructive evaluation of materials using ultrasounds is frequently used in industry as a way to characterize material properties and to locate defects. Imaging methods based on back propagation of echo signals are limited to reconstruction of low spatial frequency. Ultrasound diffraction tomography is a transmission-based imaging method which gives the possibility of characterizing anisotropic materials. Recent progress in ultrasound tomography using a circular array of transducers enabled velocity mapping of materials with high sound speed contrast relative to the background. In this study, an image of acoustic properties of a titanium rod submerged in water was obtained using bent ray time-of-flight tomography. The experiments were performed on a circular array ultrasonic test bench at 2 MHz frequency on 322 transduction points. A 2D finite element model of wave propagation through water and titanium was developed in order to validate experimental results. Finite element reconstruction matched experimental results within a 10% error for geometry and velocity.

Invited Papers

11:20

2aBA9. Recent advances in resonant ultrasound spectroscopy (RUS) for the measurement of the stiffness tensor of anisotropic and attenuative materials. Quentin Grimal (Biomedical Imaging Lab., Sorbonne Universités - Université Pierre et Marie Curie, 15 rue de l'école de médecine, Paris 75006, France, quentin.grimal@upmc.fr)

The elasticity tensor of a small sample of anisotropic material can be advantageously determined with Resonant Ultrasound Spectroscopy (RUS). In RUS, resonant frequencies of a sample are measured, and computed frequencies of a numerical model of the sample are fitted, yielding the stiffness tensor. RUS was developed in the 1990s, but until recently, it was in practice limited to measure materials with a high quality factor. We have recently adapted the method to measure attenuative materials such as plastics and hard biological tissues (bone and tooth tissues) with a typical quality factor of about 25. Our strategy combines Bayesian methods to retrieve overlapped resonant peaks in the RUS spectra and to solve the inverse problem using a limited number of resonant frequencies. The method allows a quasi-automated processing of RUS spectra where it is not necessary to know a priori the pairing between measured and computed frequencies. In the last years, we have extensively used RUS to document the anisotropic elastic properties of human bone and we explored application in additive manufacturing.

11:40

2aBA10. Optical-resolution photoacoustic imaging with speckle illumination. Emmanuel Bossy (LIPhy, Université Grenoble Alpes/CNRS, LIPhy, 140 rue de la Physique, Saint-Martin d'Hères 38400, France, emmanuel.bossy@univ-grenoble-alpes.fr)

Conventional approaches for optical-resolution photoacoustic microscopy generally involves raster scanning a focused spot over the sample. Here, we show that a full-field illumination approach with multiple speckle illumination can also provide diffraction-limited optical-resolution photoacoustic images. Two different proof-of-concepts are demonstrated with micro-structured test samples. The first approach follows the principle of ghost imaging [1], and is based here on solving a linear inverse problem under sparsity assumptions: the object is reconstructed through a pseudo-inverse computation of a reference matrix made of speckle patterns measured during a calibration step. The second approach is a speckle scanning microscopy technique, which adapts the technique proposed in fluorescence microscopy by Bertolotti *et al.* [2]: in our work, spatially unresolved photoacoustic measurements are performed for various translations of *unknown* speckle patterns. Because speckle patterns naturally appear in many various situations, including propagation through biological tissue or multi-mode fibers, speckle-illumination-based photoacoustic microscopy provides a powerful framework for the development of novel reconstruction approaches, well-suited to compressed sensing approaches. [1] Katz *et al.*, "Compressive ghost imaging," *Appl. Phys. Lett.* **95**(13), 2009. [2] Bertolotti *et al.*, "Non-invasive imaging through opaque scattering layers," *Nature* **491**(7423), 2012.

Session 2aEA

Engineering Acoustics: Thermophone Transduction

Thomas R. Howarth, Cochair
NAVSEA Division Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841

Andrew R. Barnard, Cochair
Mechanical Engineering - Engineering Mechanics, Michigan Technological University, 815 R.L. Smith MEEM Building, 1400 Townsend Drive, Houghton, MI 49931

Chair's Introduction—9:00

Invited Papers

9:05

2aEA1. Platinum films to carbon nano structures: The history of the thermophone transducer. Andrew R. Barnard and Mahsa Asgarisabet (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu)

The thermophone is a device which creates sound using rapid heat oscillations on its surface. This results in a thin and lightweight loudspeaker with no moving parts. Braun was the first to address this phenomenon in the late 1800s. Arnold and Crandall developed the first theory and correlated experimental results in 1917. Unfortunately, the materials necessary to make an efficient thermophone did not exist in their time. In 2008, Xiao rediscovered the thermophone effect using carbon nanotube thin films, a material much better suited to efficient thermophones. Since then many researchers have been working on developing thermophone technology using carbon nanostructures including nanotubes, nanofibers, and graphene. This talk will cover the history of the thermophone from its early days through today and give a broad overview of the application areas for this technology, which span from underwater transducers to consumer electronics and automotive applications.

9:25

2aEA2. Thermoacoustic sound projector: Beyond the fundamental efficiency of carbon nanotubes. Ali E. Aliev (Alan G. MacDiarmid Nanotech Inst., Univ. of Texas at Dallas, P.O. Box 830688, BE 26, Richardson, TX 75083, Ali.Aliev@utdallas.edu)

Advances in thermophone transduction from the perspective of novel nanostructured materials, device design, and signal processing will be presented. The comparison of studied 2D and 3D networks of nanostructured materials with aerogel structures will be given. The energy conversion efficiency of encapsulated thermoacoustic devices excited by short pulses with varying duty cycle, shape of pulse, and carrier signal frequency will be analyzed for a variety of fabricated devices. I will provide an extensive experimental study of pulse excitation in different thermodynamic regimes for freestanding carbon nanotube sheets with varying thermal inertias (single-wall, multi-wall with varying diameters, and number of superimposed sheets) in vacuum and in air. The experimental observations are rationalized within a basic theoretical framework. The acoustical and geometrical parameters providing further increase in efficiency and transduction performance for open and closed resonant systems will be discussed. [Research supported by ONR, Grant No.: N00014-17-1-2521.]

9:45

2aEA3. The fabrication and characterization of nanocarbon foams for their utilization in thermoacoustic device. Mei Zhang, Paul Wolmarans (High-Performance Mater. Inst., Florida State Univ., IME, FAMU-FSU College of Eng., 2525 Pottsdamer St., Tallahassee, FL 32310, mzhang3@fsu.edu), and Ali E. Aliev (A. G. MacDiarmid NanoTech Inst., Univ. of Texas at Dallas, Richardson, TX)

Thermo-acoustic (TA) sound generation (thermophone) is a non-resonant technique where electrical energy is converted to sound waves through Joule heating of a resistor (TA heat source) without any mechanical vibration, thus allowing for a wideband operation. It is clear that the material and the structure of the resistor play an important role on the performance of the thermophone. Recently, thermophone transducers fabricated from nanoscale materials hold the promise of a new transducer technology. Transducers made from these nanomaterials operate over a broad frequency range and can be designed to be lighter and thinner than competing technologies. Here, we report the TA heat source using nanocarbon foams. Nanocarbon foams are carbon nanotubes (CNTs) based all carbon porous materials. They have a hierarchically porous structure and the pore size and porosity can be tuned easily during the fabrication process. The foams consist of highly porous conductive CNT networks. They are freestanding, flexible, and mechanically robust in various environments. It is demonstrated that the foams can be used as elastically compressible, flexible TA heat source. The detailed fabrication process, the morphology of the foams, their thermal and electrical properties, and their performance as thermophone transducers will be presented.

10:05

2aEA4. Thermophones for sonar applications. Nathanael K. Mayo, Christian Schumacher, Donald Cox, Jeffrey E. Boisvert, John B. Blottman, and Thomas R. Howarth (Naval Undersea Warfare Center: Newport, Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841-1708, nathanael.mayo@navy.mil)

Thermophones are electrically driven thermoacoustic sound projectors which have historically been used as a primary source of sound. Thermophones have a sound spectra that are largely determined by their housing and support and efficiencies that are determined by the device's ability to maintain a dynamic temperature gradient across a gaseous layer surrounding the thin heating element. A number of factors make thermophones an attractive technology for underwater use including the relative ease and low cost of production, the large thermal reservoir provided by the surrounding aquatic environment, and the ability to tune the spectra over a broad range of frequencies. We present calibrated acoustic underwater tests performed on thermophones which demonstrate the potential for a new class of sonar transducers. Small, 6.35 cm diameter, inert gas filled laminate pouch thermophones were fabricated which provide a low frequency resonance. Additionally, a liquid filled thermophone demonstrates a smooth response over a wide frequency band.

10:25–10:40 Break

10:40

2aEA5. Cylindrical heat conduction models for enclosed thermophones. Benjamin Dzikowicz, James Tressler, and Jeffery W. Baldwin (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil)

The efficiency of enclosed thermophones is governed by the time-dependent heat transport between the heating element and the enclosure gas. Efficiency gains have been made by reducing the heat capacitance of the heating elements so that a maximum amount of energy is available for transport. However, efficiency is still limited when there is poor conduction into and out of the enclosure gas. An analytical model for the conduction in an enclosed thermophone with a heating element composed of a regularly spaced linear array of conducting fibers will be presented. The model considers interaction between the individual fibers and treats the enclosure wall as a heat sink. The conduction model will be incorporated into a device model which can calculate steady-state acoustic output. The results are compared against experimental data. In addition, the device model can be used to explore the optimal fiber spacing for enclosed thermophones with regularly spaced linear array heating elements. [Work funded by the Naval Research Laboratory.]

Contributed Papers

11:00

2aEA6. Carbon nanotube coaxial thermophone for automotive exhaust noise cancellation. Suraj M. Prabhu, Andrew R. Barnard (Mech. Eng., Michigan Technolog. Univ., 1400 Townsend Dr., 815 R.L. Smith Bldg., Houghton, MI 49931, smprabhu@mtu.edu), and Mahsa Asgarisabet (Mech. Eng., Michigan Technolog. Univ., Plymouth, MI)

Exhaust noise from automobiles is a major concern and needs to be controlled. Current noise control systems, mainly mufflers, have significant size, weight, and performance limitations at low frequencies. Passive control systems are unable to cancel noise efficiently over the entire frequency spectrum. Active noise control system helps to overcome the above limitation. This paper deals with the design and testing of a coaxial carbon nanotube speaker for active cancellation of automotive exhaust noise. Carbon nanotubes are virtually massless and work on the thermo-acoustic principle. Their ability to sustain heat makes them suitable for use in elevated temperature environments such as automotive exhausts. Analytical calculations are performed using transfer matrix method to evaluate the effect of coaxial thermo-phone and side branch speakers on the sound pressure level in the exhaust tailpipe. Design of the coaxial CNT thermophone and testing of it by mounting on an automotive exhaust tailpipe are discussed. Low frequencies (20 Hz–1000 Hz) are considered to evaluate the performance of the thermophone. For active cancellation of exhaust noise, Filtered-X Least Mean Square (FXLMS) algorithm is developed and implemented using NI LabVIEW.

11:15

2aEA7. An experimentally validated electrical-thermal-acoustic model of non-planar carbon nanotube thermophones. Mahsa Asgarisabet, Andrew R. Barnard, and Suraj M. Prabhu (Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, masgaris@mtu.edu)

Carbon nanotube (CNT) thin-film thermophones are a solid state, transparent, magnet-free, stretchable, and lightweight transducers that work via the thermoacoustic effect. The rapid change in the temperature of the CNT film with low heat capacity produces a temperature wave accompanied by an acoustic wave in a frequency range of 1 Hz to 100 KHz. The existing lumped parameter models for the planar CNT thermophones are not appropriate for the complex geometries of the CNT film. Using COMSOL multiphysics, an electrical-thermal-acoustic model of the non-planar CNT thermophone has been developed. By applying an alternating electrical current to the CNT film, the temperature variation was obtained and used to simulate the pressure distribution. For different input power levels, the temperature distribution on the CNT film was compared to the experimental data from thermal camera. The experimental sound pressure level at different locations in front of the CNT thermophone were recorded and used to validate the non-planar model.

11:30

2aEA8. Application of carbon nanotube thermophones in consumer electronics. Stephania M. Vaglica (Mech. Eng., Michigan Technolog. Univ., 52134 wembly ct, Shelby twp, MI 48315, smvaglic@mtu.edu) and Andrew Barnard (Mech. Eng., Michigan Technolog. Univ., Houghton, MI)

A thermoacoustic transducer is a device that converts electrical energy into heat energy which then generates sound energy. Carbon nanotubes (CNT) are a thermoacoustic transducer that have the ability to transform this energy. Carbon nanotubes are hexagonal lattice tubes of carbon that have diameters one ten thousandth of the thickness of human hair. These graphite allotropes of carbon have a range of different applications and properties, particularly thermoacoustic conductivity. Because of their unique thermoacoustic properties, a lightweight loudspeaker can be designed by creating an aligned film of carbon nanotubes. CNTs form nanoscale tubes that are positioned in a "forest" like configuration. Unlike a typical loudspeaker, there is no magnetic coil that generates sound. Instead, sound is generated by the nanotubes fast cooling and heating property. This cooling and heating effect will expand and contract the air around the speaker which will result in the propagation of a sound wave. This new type of speaker creates a brand new market for speciality consumer electronics.

11:45

2aEA9. Thermal and vibration induced rotor blade deicing using carbon nanotube thermophones. Steven Senczyszyn and Andrew R. Barnard (Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, sasenczy@mtu.edu)

The accumulation of ice on helicopter blades imposes limitations on the design and operation of rotorcrafts due to the large power requirement of thermal heating mats. Carbon nanotube (CNT) loudspeakers present a lightweight and efficient solution to this issue, utilizing both thermal and vibratory methods to remove ice and prevent it from forming. To evaluate the deicing capabilities of CNT, an enclosure was created to encapsulate the loudspeaker between two thin layers of aluminum to simulate the rotor blade surface. The speaker was then operated at a range of amplitudes and frequencies in simulated cold weather conditions. Preliminary tests show promising results, with surface temperatures exceeding 120°C in under four minutes when operated in a −20°C environment and surface excitation resultant from the ultrasonic vibration. Although further testing is needed, CNT loudspeakers show great promise in becoming a lightweight and efficient solution to the issue of ice accumulation on the surface of rotor blades.

TUESDAY MORNING, 5 DECEMBER 2017 STUDIOS FOYER, 9:00 A.M. TO 12:00 NOON

Session 2aED

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

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

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

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

2aED1. Music a scientific art: A call for review of Department of Music, University of Nigeria Nsukka's curriculum and course outline. Stephen G. Onwubiko (Music, Univ. of Nigeria, University of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Every academic discipline ranging from the Social Sciences, Engineering, Physical Sciences, Arts, and Humanities is typically categorized as either a Science or Arts discipline. While Music is artistic and creative, there is harm in designating music as solely an Arts discipline. It is literally true that music is art, yet the viral damage of categorizing music as solely an Art discipline cannot be overemphasized. Just like a "wolf in sheep skin," music is science dressed up like art. Music is an offspring of pure science (physics) expressed as art. Music is made up sound, frequency, pitch, resonance, vibration, echo, and Doppler effect; these topics are pure science with mathematical derived formulas and not just art. For example, the action of a typical upright piano obeys Newton's third law. This paper condemns the outright definition and categorization of music as strictly an art and recommends the re-evaluation of music education at the university. The music curriculum should be rejuvenated and embedded with elective courses from the department of Physics and Astronomy to enable scholars know more about the science of their instruments, while acquiring the mastery of the art of music.

2aED2. Vowel mergers in the American South. Shawn C. Foster, Joseph A. Stanley, and Margaret E. Renwick (Linguist, Univ. of Georgia, 1907 S. Milledge Ave., Apt. D-4, Athens, GA 30605, shawnchristianfoster@gmail.com)

This study investigates the occurrence of a series of vowel mergers in the English of the Southern United States. Investigated mergers include the pin-pen merger, which may be considered characteristic of Southern English, as well as the cot-caught, pull-pool, fill-feel, and cord-card mergers, which are less strongly identified with the region. First and second formant values of over 300,000 vowel measurements from 41 speakers were automatically extracted from.wav files contained in the Digital Archive of Southern Speech (DASS). Vowel distributions were statistically analyzed via Pillai scores, Euclidean distance, and mixed-effects modeling to determine to what degree, if any, the speakers participate in these mergers. Previous studies on vowel merger in the United States have typically classified speakers into discrete "merged" or "unmerged" categories, often relying on speaker or listener judgments to do so. The richness of the acoustic data and high number of tokens available for this study allow speakers to instead be classified along a spectrum, where they are described by "degree of merger" rather than binarily. The results in turn provide new perspective on how Southerners have participated in or resisted dialectal changes that have shaped American English from the late 18th through the mid-20th century.

2aED3. Student experiment for measure of target strength. Zhengliang Cao, Jianfeng Tong, and Wei Shen (Hadal Sci. and Technol. Res. Ctr., College of Marine Sci., Shanghai Ocean Univ., #999 Hucheng Huan Rd., Shanghai 201306, China, zlcao@shou.edu.cn)

The target strength is a measurement of the reflection coefficient of a sonar target. Its value and feature is very important to detection and identification in active sonar, especially for fish by fishery acoustic students. A steel cylindrical shell is used as the target, whose outer diameter 194 mm, inner diameter 184 mm, length 500 mm, and both end closed covers with thickness 4 mm. A broadband transducer is selected to directionally project sound wave and a hydrophone receives the reflection wave from the target. The transducer and receiver are located at 5 m and 4 m from the target with the same depth 5 m in the water. When single frequency pulses are transmitted with signal interval frequency 50 Hz, the target strength is calculated by amplitude ratio of target echo and incidence wave at different frequencies. The result is compared with theoretical value for an infinite elastic cylindrical shell and finite length elastic cylindrical shell. This experiment will give a strategic guidance to understand what target strength is, how to understand it, and why we need it. [Work supported by National Natural Science Foundation of China (Grant No. 41374147).]

2aED4. Evaluating the validity and efficiency of tablet audiometry in school-age children. Jenna Supinski (Health Sci. and Human Performance, Ithaca College, 953 Danby Rd., Ithaca, New York 14850, jsupinski@ithaca.edu), Olivia M. Pereira (Ctr. for Pediatric Auditory and Speech Sci., Nemours/ Alfred I. duPont Hospital for Children, Wilmington, DE), Mackenzie Hammond (Dept of Commun. Sci. & Disord., West Chester Univ., West Chester, PA), Samantha Scharf (Dept of Linguist and Cognit. Sci., Univ. of Delaware, Wilmington, DE), Thierry Morlet, and Kyoko Nagao (Ctr. for Pediatric Auditory and Speech Sci., Nemours/ Alfred I. duPont Hospital for Children, Wilmington, DE)

Tablet-based automated audiometry offers a portable alternative to traditional audiometry. The limited research available (e.g., Rourke *et al*., 2016) supports the clinical use of automated audiometry for pediatric hearing screenings, but its accuracy in children and efficiency outside of clinical environments is undetermined. This study's objective is to evaluate the validity and efficiency of automated audiometry in school-age children. This initial phase aimed to establish its validity in a clinical setting. Hearing thresholds for 0.5, 1, 2, 4, 6, and 8 kHz were collected in children ages 6–12 years old using standard audiometry and an iPad app automated audiometry in a sound-proof booth. Tympanometry, acoustic reflex thresholds, and distortion product otoacoustic emissions were administered to examine peripheral hearing function. Preliminary results of 16 children show no significant difference between the two test durations. Automated audiometry thresholds were within 5 dB of the standard audiometry thresholds for each tested frequency, except at 6 kHz where they were within 7dB. There was no test preference among the participants. Our preliminary results support the use of automated audiometry in children. Current testing is evaluating its validity and reliability under less ideal testing environments. [Work supported by NIH COBRE Grant P30GM114736 and the Nemours Foundation.]

2aED5. Assessing temporal compensation of speech due to delayed auditory feedback. Samantha N. Davis (Univ. of Washington, 5521 Summer Blvd., Galena, OH 43021, sndavis94@gmail.com) and Francois-Xavier Brajot (Ohio Univ., Athens, OH)

This study examined the behavioral compensation of speech influenced by delayed auditory feedback. It was hypothesized that delayed auditory feedback would yield similar compensation patterns as other auditory perturbations to speech do, in a hypothesis untested by previous experiments. The results of past studies which experimented with pitch, loudness, and formant frequencies look similar to each other, due to somatosensory feedback's role in adaptation. Compensatory responses, when measured against non-altered productions, seem to approach an asymptote, yielding incomplete compensations. Compensatory responses, when measured against the size of the perturbation, resemble a decreasing exponential function, as relative magnitudes get drastically smaller with increasing perturbation size. The data supported the hypothesis as predicted, meaning the compensation to delayed auditory feedback appears to follow the same principles as those observed in other altered auditory feedback paradigms. These results suggest that this compensation technique is consistent for all altered auditory feedback methods, and that it follows a generalized rule for sensorimotor feedback control in speech production.

2aED6. Implementing electronic speckle pattern interferometry for a variety of activities in a general education acoustics class. Nicholas J. Razo and Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

A familiar acoustics laboratory experiment is finding Chladni patterns using flat plates of various geometries. Another method used in musical acoustics to find mode shapes of vibrating objects is electronic speckle-pattern interferometry (ESPI). We introduced students in a general education acoustics class to ESPI in addition to the traditional Chladni plate laboratory activity. A variety of ways to implement ESPI in the class were developed including: comparing traditional Chladni patterns to ESPI, examining traditional musical instruments with ESPI, and an advanced investigation of a well-known acoustics demonstration. A classic acoustics demonstration is rubbing the rim of a wine glass to create a musical tone. A coffee mug, unlike a wine glass, has an asymmetry due to the mug handle. ESPI can be used to study the effects of filling the mug with water in terms of the mug's natural frequency, amplitude, and mode shapes. Results from ESPI showed that the mode shapes do not change with the addition of water. Filling the coffee mug with water lowered the natural frequencies of the lowest modes in a manner similar to that of a wine glass.

2aED7. Sound speed profile calculations for the Northern Gulf of Mexico. Bradley J. Sciacca (Dept. of Phys., Univ. of New Orleans, 43 Lake Lynn Dr., Harvey, LA 70058, bsciacca@uno.edu) and Juliette W. Ioup (Dept of Phys., Univ. of New Orleans, New Orleans, LA)

Many sound speed calculations for the Gulf of Mexico use an average sound speed over the entire Gulf. In a variety of applications, such as localization or tracking of marine mammals, a more precise sound speed profile is helpful to make more accurate calculations. Using a reasonable approximation to Del Grosso's equation, the sound speed profile may be computed as a function of temperature, pressure, and salinity. The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015 using Environmental Acoustic Recording Systems (EARS), returning to sites previously surveyed by LADC. Oceanographic data were also collected at those sites and can be used for more accurate sound speed profiles at the specific locations and specific times of the acoustic data recordings, increasing the accuracy of subsequent calculations such as real-time tracking and other applications. Sound speed profiles for sites in the northern Gulf of Mexico will be presented. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

2aED8. Noise levels during recreational flight of a Cessna 172 Skyhawk. Drew Jones (Audiol. & Speech Pathol., Univ. of Tennessee, 1144 Wincreek Dr., Collierville, TN 38017, djone120@vols.utk.edu), James D. Lewis, and Mark Hedrick (Audiol. & Speech Pathol., Univ. of Tennessee, Knoxville, TN)

Recreational and private pilots are regularly exposed to potentially hazardous noise levels during flight training sessions and recreational flying. For the first part of this study, light airplane noise was recorded during recurrent 1.5-hour flight training sessions. Measurements were made at the level of the pilot's left and right shoulders (using dosimeters). Data demonstrated that changing inflight operations and power adjustments were associated with specific acoustic patterns. Noise levels (Leq) ranged from 76 dBA to 96 dBA during flight. The second part of this study examined the effect of the noise on subjective and objective measures of the pilot's auditory function. Measurements were made before and immediately after flight training sessions. Pure-tone audiometric thresholds, middle-ear absorbance and click-evoked otoacoustic emissions did not show significant changes from pre- to post-flight. Although inflight noise levels are potentially hazardous, the use of an aviation headset acted to mitigate the risk of hazardous exposure. Future research directions include measurements in diverse aircraft and across longer flight durations.

2aED9. Regional dialect perception by British and American listeners. Emma C. Bonifield (Speech and Hearing Sci., Indiana Univ., 1615 Springhill Rd., Warsaw, IN 46580, embonifi@indiana.edu), Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), Bronwen Evans, Isla Tyrrell, and Fernanda Aguilera (Univ. College London, London, United Kingdom)

Speakers' regions of origin, across and within countries, can substantially impact their native language pronunciation patterns. Previous work suggests that listeners are sensitive to these dialect differences within their home country (Clopper & Pisoni, 2004) and have at least broad representations of dialects outside of their home country (Bush, 1967). The current experiment expands on these findings by investigating dialect perception within and across multiple countries for listeners from two countries: United States and United Kingdom. Fifty-one talkers from the Speech Accent Archive (Weinberger, 2013) representing 6 U.S. regional dialects, 5 U.K. dialects, and 6 dialects from other English-speaking countries produced the same two sentences. Participants ranked these speech samples on their proximity to "Standard American English" or "Standard Southern British English" in two ladder tasks. American and British listeners' ladder rankings for the American baseline were very highly correlated, with a clear split between North American and non-North American dialects. For the British baseline, American listeners again had a split between the North American and non-North American dialects, while the British listeners showed more gradation across the dialects. The results suggest that listener home dialect and dialect familiarity, possibly through media exposure, shape perceptual representation of regional dialects.

2aED10. Full-spectrum comparison of denoising algorithms for real-time magnetic resonance imaging acoustics. William Klock and Marissa Barlaz (Linguist, Univ. of Illinois at Urbana-Champaign, 707 S Mathews Ave., 4080 Foreign Lang. Bldg. MC-168, Urbana, IL 61801, wklock2@illinois.edu)

Using real-time MRI acoustic data, we employ two methods of signal denoising (DLWP and CS-SNG) to conduct a preliminary comparison between noisy, denoised, and noiseless data. The acoustic data collected in the MRI serve as the noisy, "baseline" group. Data collected from the same speakers in a sound-attenuated environment served as the noiseless, "ground truth." We calculate acoustic power across the frequency spectrum in 32, 64, 128, and 256 bin experiments and perform k-means clustering on the first three principal components to compare the output of the denoising algorithms to the ground-truth and noisy data. Results show a quantitative difference between the denoising methods, through their different affinities for clusters associated with reference group labels. The groupings indicate that the CS-SNG data are better suited for establishing a map between the visual data from the MRI and the acoustic output because of its association with the noiseless data and its distinction from the noisy group. Since both denoising algorithms form independent clusters, there are potential differentiating features that could drive future improvement of these methods.

2aED11. Measurements of the Doppler effect due to a rotating sound source. Elizabeth McQueney, Maryam Landi, and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, mcqueney.e@gmail.com)

The Doppler effect is a common principle in all types of waves when sources and receivers have a relative motion. In acoustics, there are scenarios where sound sources rotate and produce the Doppler effect. These scenarios include sound produced by a speaker with a rotating horn and by the rotational motion of helicopter blades and airplane propellers. In this study, we experimentally measure the frequency shift due to a rotating sound source. A buzzer (2.8 KHz) is set to rotate at a fixed radius (75 cm) with the rotational velocity (60 rpm) controlled by a stepper motor. The radiated sound signals are recorded by a microphone at different locations. Spectrograms of the recorded signals display a shift from the buzzer's original frequency. The shift oscillates with the period of the source rotation. We characterize the dependence of the shift on the position of the microphone. A formula for the shift is derived and is used to compare with the measurements. Our measurements and analyses gain insight into the Doppler effect due to a rotating source.

2aED12. Localizing crow vocalizations in social aggregations. Derek Flett, Virdie Guy, Shima Abadi (Eng. and Mathematics, Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, flettd23@uw.edu), and Douglas Wacker (Biological Sci., Univ. of Washington, Bothell, WA)

The North Creek Wetlands Restoration on the University of Washington Bothell campus is home to a large nocturnal American crow (*Corvus brachyrhynchos*) roost. Each day from Autumn to Spring, crows form pre- and post-roost aggregations, which consist of tens to hundreds of crows. Crows on these aggregations are often highly vocal, but the functions of their vocalizations are not well understood. Identifying any context-dependent patterns in these vocalizations is critical to fully understand communication in this highly social and intelligent species. Previous studies have shown the presence of human observers near large groups of crows may disrupt natural vocal and non-vocal behavior. In this study, the potential confound of these observer effects are eliminated by recording crow vocalizations using a remotely activated, time-synched microphone array. Simulations are undertaken to study the performance of the Time Difference of Arrival (TDOA) method to localize individual callers. A parametric study is used to analyze the effects of number of receivers, signal frequency and duration, and crow location on the performance of TDOA. In addition to the simulation, different types of recorded crow vocalizations are used to design robust playback experiments to fine-tune our localization technique for use in actual crow aggregations in future.

2aED13. Comparisons of measured sound power levels across octave bands gathered from different methods and labs. Samuel H. Underwood and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, samuelunderwood@unomaha.edu)

This poster presents results to date from an interlaboratory study, aimed at quantifying the bias and reproducibility of three different sound power measurement methods used in the heating, ventilation, and air-conditioning industry: free field method, diffuse field method, and sound intensity method. The sound power of a certain loudspeaker sound source has been measured in at least 15 testing facilities using the methods generally applied in those facilities. Two test signals are used in the measurements: a broadband signal whose slope decreases −5 dB per octave band, and the same signal with four discrete frequency tones at 58, 120, 300, and 600 Hz. Comparisons of the measured sound power level data, gathered to date, are presented across one-third octave bands. The project continues to collect data from other test facilities so that rigorous findings on the repeatability, reproducibility, laboratory bias, and measurement method biases may be determined, according to ISO 5725. [Work supported by the Air-conditioning, Heating and Refrigeration Institute.]

2aED14. Modeling nonlinear acoustic landmine detection using a soil plate oscillator apparatus. Emily V. Santos and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com)

In modeling nonlinear acoustic landmine detection, there will be a cylindrical drum-like buried target. This target, which has a clamped thin elastic plate, rigid sidewalls, and bottom plate, is buried in an open concrete "soil tank" containing dry sifted masonry sand. Subwoofers located 40 cm above the surface are driven with a swept tone from 50 to 450 Hz to generate airborne sound that couples into the sand exciting vibration in the buried target. A small geophone is used to measure the "ground" vibration profile across the buried target. Nonlinear tuning curves of vibration particle velocity vs. frequency are used to compare "over—off the target" results. The backbone curves (peak velocity vs. corresponding resonant frequency) are

different in these cases—such that resonances due to soil alone or soil over a compliant buried target can be distinguished. Next, a soil plate oscillator SPO (consisting of two circular flanges sandwiching and clamping a thin circular elastic plate that supports a cylindrical level column of sand above the plate) is driven by airborne sound. Nonlinear tuning curve vibration at points across the sand are compared with the results in the "idealized" landmine detection experiment to develop a lumped element model of the system.

2aED15. Computer simulation of synthetic aperture sonar or radar for classroom demonstration: Part II. Kathryn P. Kirkwood and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m193342@usna.edu)

In our presentation, a Mathematica® simulation of Synthetic Aperture Acoustic Radar will demonstrate how point-like targets placed on a smooth surface can be imaged from a collection of acoustic echoes. The transmitter and receiver are collocated and modeled as a single element that stops along a linear track at collection points and hops to the next location (stop and hop approximation). The transmitter on the hypothetical apparatus will transmit a linear frequency modulated LFM chirp pulse that reflect off the targets and are recorded as echoes by the receiver at discrete locations along the track. A matched filter correlation process will perform a pulse compression of the LFM chirp. A series of still frames from the animated simulation will be displayed. These frames show the echoes arriving at different times from the targets in conjunction with a receiver wave recorder screen. The time correlation backprojection algorithm is motivated when pulse compressed arrivals are displayed (at various track locations) including the echo arrival of a hypothetical point target adjusted in the x,y plane on or near one of the actual point targets. [See Yegulap, "Fast backprojection algorithm for SAR," in Proc. 1999 IEEE Radar Conf., Waltham, MA, April 20–22, 1999, 60–65].

2aED16. Investigation of sound level variations in occupied K-12 classrooms. Jared Paine (Civil Eng., Case Western Reserve Univ., 2095 Martin Luther King Jr. Dr., Cleveland, OH 44016, jap198@case.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

Sound level data have been logged in 220 K-12 classrooms, as part of an investigation underway at the University of Nebraska—Lincoln on environmental conditions inside occupied classrooms. In each classroom, equivalent sound levels were logged every 10 s with an integration time of 10 s, during 36 hour periods that spanned two occupied school days, at three different times throughout an academic year. Previous presentations on this dataset have focused on average occupied and unoccupied sound levels for each classroom; this poster investigates the variation of the sound levels in the classrooms throughout the occupied days and how those differ across the grade levels measured (specifically, 3rd, 5th, 8th, and 11th grades). Among the quantifiers analyzed to understand sound level variation are standard deviation, L10-L90, and occurrence rates. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

TUESDAY MORNING, 5 DECEMBER 2017 — STUDIO 4, 8:15 A.M. TO 11:15 A.M.

Session 2aMU

Musical Acoustics: Measurement Methods and Instrumentation for Musical Acoustics

Wilfried Kausel, Cochair
Inst. of Music Acoustics, Univ. of Music and Performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria

Thomas Moore, Cochair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—8:15

Invited Papers

8:20

2aMU1. Detailed experimental modal analysis of a trumpet: An application of laser Doppler vibrometry. Martin Cockrill (ASDC Dept. of Eng., The Univ. of Leicester, UK, ASDC Unit 5, The Control Ctr., MIRA Technol. Park, Nuneaton CV10 0TU, United Kingdom, mc490@le.ac.uk) and Maximilian Chowanietz (ASDC Dept. of Eng., The Univ. of Leicester, UK, Leicester, United Kingdom)

Much work has been conducted on the physics of musical instruments and the trumpet has been included in numerous studies. However, the majority of the work is directed at understanding the air column physics since that is the dominant mechanism of a brass instrument. While this is the largest contribution to understanding the sound, it fails to explain the minute differences between instruments from both the player's point of view and the audience. We have looked to add to the global body of work by conducting a full experimental modal analysis of an entire trumpet. By utilizing a 3D scanning laser Doppler vibrometer, we have been able to measure the dynamic response both in and out-of-plane over the entire trumpet. We have been able to do this at around 500 locations providing detailed geometric resolution and resolving higher order modes. We have then solved the resulting dynamic model to provide natural frequencies and mode shapes. It is hoped that the results of this work can be combined with the virtual modelling of other researchers to refine the state of the art in terms of understanding the properties and mechanisms inherent in trumpets and the quality of the sound they produce.

8:40

2aMU2. Experimental techniques for the visualization of the acoustic fields radiating from brass musical instruments. Amaya López-Carromero, D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, 1608, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh, Scotland/Midlothian EH9 3FD, United Kingdom, s1374028@sms.ed.ac.uk), and Jonathan A. Kemp (Music Ctr., Univ. of St Andrews, St., Andrews, Fife, United Kingdom)

Radiation from brass instrument bells under artificial excitation can be visualized using a number of experimental techniques. In this work, we present two different approaches: Schlieren optical imaging using a high speed camera and acoustic 2D scanning using a linear microphone array. The potentials and limitations of both methods, as well as their methodological requirements, are discussed. Examples of measurements on several instruments are presented. Schlieren optical imaging is applicable for the detection of large amplitude wavefronts formed by nonlinear wave steepening and is this is particularly sensitive to the high frequencies that form within these waves. The 2D scanning linear microphone array on the other hand can, in principle, measure all frequencies and the use of the exponential sine sweep method allows any nonlinear behaviour (in the loudspeaker and microphone or in the air) to be removed to get an accurate measurement of the linear radiation field for any frequency of choice within the measurement bandwidth.

9:00

2aMU3. Flow visualizations using electronic speckle pattern interferometry. Whitney L. Coyle and Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, wcoyle@rollins.edu)

Imaging air flow in and around a musical instrument is a difficult task, but it can be important for understanding the physics associated with sound production as well as verifying the accuracy of computer simulations. Particle image velocimetry (PIV) has been successfully used to image air flow, but it requires expensive equipment, extensive technical expertise, and sophisticated software. Furthermore, PIV is usually restricted to observing only a small area of the flow. As an alternative to PIV, we demonstrate a method based on electronic speckle pattern interferometry that allows large-area real-time imaging of air flow using minimal optical equipment.

9:20

2aMU4. Searching for early reflected waves after strike of Caribbean steelpan using time-resolved electronic speckle pattern interferometry. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu), Thomas Moore, and Daniel Zietlow (Dept. of Phys., Rollins College, Winter Park, FL)

Electronic speckle-pattern interferometry has become a popular tool for studying the operating deflection shapes and normal modes of musical instruments. Normally, ESPI is used with cameras having frame rates below 50 Hz, and the resulting interferograms are the result of a time average over many oscillations. Using ESPI with a high speed camera operating at frame rates of several thousand frames per second allows for time-resolved examinations of transient motion, and this technique has been used to study the motion immediately following the strike on a Caribbean steelpan. Caribbean steelpans are a complex system of coupled oscillators and it has been suggested that the inflections from the concave shape of the steelpan bowl to the convex shape of the note areas serves as a boundary where mechanical waves are partially reflected. High speed ESPI movies of strikes on a low tenor steelpan were acquired in an effort to search for evidence of these reflected waves.

Contributed Papers

9:40

2aMU5. Humidity influences on natural Timpani heads. Wolfgang Nagl and Alexander Mayer (Dept. of Music Acoust. (Wiener Klangstil), mdw – Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, wolf.nagl@gmx.at)

Varying air moisture is an issue timpanists using natural skins have to deal with. Moisture, which in concert halls is often created by the audience or by the musicians on the stage, causes natural skin to loose its tension and therefore flattens the pitch of the instrument. The present paper deals with the effects of moisture on the pitch of timpani with natural skin presenting an experiment for measuring the relationship between air moisture and skin tension resp. pitch variation. An experimental setup has been created where the natural skin head was mounted on a copper shell containing an ultrasonic humidifier. The humidification process was computer controlled in order to establish exact humidity levels. The resulting change of the kettledrum skin parameters was measured using an impulse hammer and force transducers at all mounting points of the tensioning hoop. Stretching parameters have been captured by a special video-system. Measurements have been made in a relative humidity range of up to 91%. The results may also guide musicians and help them to cope with humidity variations during their concerts.

9:55

2aMU6. Sound fields forever: Mapping 3D sound fields using position-aware smartphone technology. Scott H. Hawley, Sebastian Alegre, and Brynn Yonker (Chemistry & Physics, Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu)

Google Project Tango is a suite of built-in sensors and libraries allowing certain mobile devices to track their motion and orientation in three dimensions, without the need for any external hardware. Our new Android app, “Sound Fields Forever,“ combines position information with sound intensity data for multiple frequency bands obtained from a comoving external microphone plugged into the phone’s analog audio jack, and filtered digitally. These data are sent wirelessly via a WebSockets connection to a visualization server running in a web browser. This system is intended for multiple roles: (1) education, providing visual representations of introductory acoustics laboratory phenomena such as interference patterns and room modes; (2) live sound reinforcement, providing front-of-house engineers real-time maps of a loudspeaker system’s effects throughout the venue; (3) architectural acoustics, aiding builders and designers in evaluating acoustical properties of halls. The relatively low cost of our approach (∼$400 for the smartphone) compared to more sophisticated 3D acoustical mapping systems could make it an accessible option for such applications. We present visualizations of various sound field measurements and intend to demonstrate how the bandpass filters make it suitable for use even in the presence of nontrivial ambient noise.

10:10–10:30 Break

10:30

2aMU7. Study of the impact of material on clarinet-like instrument: Correlation between impedance measurement and musician tests. Louise Hovasse, Jonathan Cottier, Jerome Selmer (Henri Selmer Paris, PARIS, France), and Vincent Gibiat (ICA Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr)

The acoustical effect of the material of woodwind instruments seems to be the favorite discussion subject between acoustician and instrument maker. Indeed in scientific literature, different hypothesis are demonstrated by musician tests or scientifically experimentation. This study, based on the acoustical input impedance measured on tubes and clarinets, as well as musician tests, tends to highlight material could have part of influence on the acoustic of woodwind instrument. Moreover, correlations between measurements and musician tests are found. Indeed, three clarinets have been tested by musicians, one as reference in African Blackwood, one half synthetic-half African Blackwood with the same density as the reference instrument and one in other species of wood with a lower density than the reference instrument. The correlation between measurement and musicians tests is that the instrument felt more brilliant is the one with the highest quality factors on the impedance peaks and reflection coefficients, whereas the instrument felt duller have the smallest quality factors and reflection coefficients. Since the quality factor and the reflection coefficient are linked to the visco-thermal losses, themselves depending on the surface condition, the acoustic of the clarinet-like instruments appears to be more impacted by the surface condition than the material itself.

10:45

2aMU8. Investigating vocal tract effects during note transitions on the saxophone. Montserrat Pàmies-Vilà (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Inst. 22, Vienna AT1030, Austria, pamies-vila@mdw.ac.at), Gary Scavone (McGill Univ., Montreal, QC, Canada), Alex Hofmann, and Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Vienna, Austria)

The resonances of the vocal tract are used by single-reed woodwind players for tuning purposes, timbre modification, and other musical effects. Using pressure or impedance measurements, the influence of the vocal tract on saxophone playing has been previously studied for steady tones. This study considers note transitions in which the players might perform vocal tract modifications (e.g., wide intervals) and combines pressure and reed bending measurements to analyze the effect of the vocal tract as well as its dependence on the articulation technique. The measurement setup consists on recording the acoustic pressure in the players' mouth and in the mouthpiece of the instrument. A strain gauge attached to the reed surface captures both the reed oscillations and the tongue-reed interaction due to articulation. Complementary details about the playing technique are obtained by recording the upper teeth force on the mouthpiece and the open-closed configuration of two selected keys. Preliminary results show that saxophonists are able to drive the vibrations of the reed by modifying the vocal tract configuration, while adapting the blowing pressure and tongue-reed interaction to the articulation technique.

11:00

2aMU9. Preliminary measurements of vibrations and resonances of a *Kundum* xylophone. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The *kundum* xylophone is unique among bar instruments because it has bull horn resonators. The *kundum* is made and played primarily by the Birom ethnic group found in Vom in the Plateau state of Nigeria, which is in the central and northern part of the Nigeria. It complements both the Western and African musical genres but is played mostly in traditional folk music. The unique tone quality of the *kundum* is far more superior and richer than other xylophone types found in traditional Nigerian societies. The handmade *kundum* consists of a row of wooden slabs that are struck with a hammer. The vibrations of the wooden slabs are amplified by closed-tip bull horns placed beneath the slabs. The bull horns are sized to produce a high-pitched note; the *kundum* uses overtones of the open-closed bull horns. This paper examines preliminary measurements of the vibrations of the wooden slabs, as well as natural resonances of closed-pipe bull horns. Comparisons are made with the resonances of ideal cylindrical and conical open-closed tubes to see if the effect of the curved tip of the bull horns warrants further study.

2a TUE. AM

Session 2aPA

Physical Acoustics: Sound Used as an Investigative Tool for Industrial Solutions

Gabriela Petculescu, Chair
Physics, University of Louisiana, Lafayette, 240 Hebrard Blvd., Lafayette, LA 70504

Chair's Introduction—9:00

Invited Papers

9:05

2aPA1. Grain size distribution measurement of Ti-6Al-4V plate using laser-ultrasonics. Feng Dong (National Eng. Res. Ctr. of Flat Rolling Equipment, Univ. of Sci. and Technol. Beijing, 2145 Sheridan Rd., Tech B224, Evanston, IL 60208-3109, dongf515@gmail.com), Quan Yang (National Eng. Res. Ctr. of Flat Rolling Equipment, Univ. of Sci. and Technol. Beijing, Beijing, China), and Sridhar Krishnaswamy (Ctr. for Smart Structures and Mater., Northwestern Univ., Evanston, IL)

As the most widely used titanium alloy, Ti-6Al-4V, has wide range of applications in aerospace industry and medical prostheses. Grain size is a significant parameter that should be controlled during material processing for the reason that it can affect the strength and toughness directly. Laser-ultrasonics is an appropriate method for in-situ measurement of the grain size and distribution during heat treatment of Ti-6Al-4V as it is a nondestructive and non-contact technique. In this study, a laser-ultrasonic system was established combining an 8 ns width pulsed laser for ultrasound generation and a two-wave mixing (TWM) interferometer for detection. Several Ti-6Al-4V samples were heat treated variously to get different grain sizes but with the same phase ratios. Attenuation rates of ultrasonic waves in the different specimens were precisely calculated using exponential fitting for various frequencies. The true values of grain size and distribution were observed via scanning electron microscopy and electron backscatter diffraction (SEM-EBSD). The frequency dependent attenuation of ultrasonic waves was obtained for various grain sizes of Ti-6Al-4V. A statistical correlation between the grain size distribution and the frequency-dependent ultrasonic attenuation was established.

9:25

2aPA2. Ultrasonic characterization of the complex Young's modulus of polymers produced with micro-stereolithography. Clinton B. Morris (Mech. Eng., The Univ. of Texas at Austin, 204 Dean Keeton, Austin, TX 78705, clint_morris@utexas.edu), John M. Cormack, Michael R. Haberman, Mark F. Hamilton, and Carolyn C. Seepersad (Appl. Res. Labs. and The Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Micro-stereolithography is capable of producing polymer parts on the scale of millimeters that contain micron-scale features. Accurate prediction of the dynamic performance of components produced using micro-sterolithography requires knowledge of the as-fabricated material properties, but very little published material property data exist for these materials. This is complicated by the fact that the properties vary as a function of build parameters (i.e., laser exposure time). Designers therefore have limited useful material property information for part design. Frequency-dependent material parameters are often determined by measuring the wave speed and attenuation of an ultrasonic pulse as it propagates through the material. This work employs laser Doppler vibrometry to detect extensional waves in a solid polymer rod of circular cross-section that is excited by a longitudinal ultrasonic contact transducer. Transverse motion associated with extensional waves propagating along the rod axis is measured at multiple surface points. The time series gathered from all locations are used to produce a frequency-wavenumber spectrum. These data, coupled with a forward model that accounts for both viscoelastic and geometric waveguide effects, are then used to invert for the lossy material properties from 0.4 to 1.2 MHz.

9:45

2aPA3. Study of temperature and pressure dependent elastic properties of porous ceramics. Ashoka Karunarathne, Josh R. Gladden, and Gautam Priyadarshan (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of MS, 145 Hill Dr., University, MS 38677, atholang@go.olemiss.edu)

Resonant Ultrasound Spectroscopy (RUS) is a widely used experimental approach to investigate the elastic properties of solid materials at different temperature and pressure. In RUS, experimentally obtained vibrational resonance spectrum of a sample is used to calculate the full elastic tensor of the material. We report here a series of RUS measurements of novel porous ceramics material designed by LG which has applications in high temperature fuel cells. The elasticity and the acoustic damping of a porous material are influenced by its porosity and its saturation fluid. The time dependent elasticity obtained from RUS measurements exhibits a two-step process of stiffening with two distinct time constants. The stiffening-softening behavior during the heating-cooling cycles and the reduction of acoustic damping in the low-pressure conditions were observed depending on the synthesis history of the material.

10:05

2aPA4. Acoustic monitoring of aluminum-alloy sensitization. Shankar Kharal (Physics, Univ. of Louisiana, Lafayette, 240 Hebrard Blvd., Lafayette, LA 70504), Nicholas Jones (Naval Surface Warfare Ctr. - Carderock Div., Bethesda, MD), and Gabriela Petculescu (Physics, Univ. of Louisiana, Lafayette, Lafayette, LA, gp@louisiana.edu)

Segregation of atomic species in metastable solid-solution alloys results in a gradual change of the alloy's properties, sometimes rendering the materials ineffective for the particular task they were designed for. An example of such a case is the sensitization of strong and lightweight Mg-rich aluminum alloys used in marine applications. The ASTM standard is an acid-corrosion test (G67), a destructive and lengthy procedure that requires large specimens analyzed off-site. Ultrasound is a well-known tool for nondestructive material characterization and it can offer a solution for on-site testing. To this end, the sensitivity of ultrasonic parameters to the degree of material degradation through sensitization has been identified. Velocity and attenuation for shear and longitudinal waves were measured as a function of sensitization for 5083 and 5456 aluminum alloys with two different methods: Resonant Ultrasound Spectroscopy (RUS) and Pulse Echo (PE). The longitudinal and shear velocities change by 0.5% and 1.5%, respectively, while the attenuation coefficient of longitudinal waves changes by of 20% (all represent saturation values). The JMAK equation for phase kinetics is used to understand the observed evolution of the acoustic parameters vs. sensitization.

10:25–10:40 Break

10:40

2aPA5. Electroacoustic modeling and characterization of an ultrasonic projector. Guy Feuillard, Pascal Tran Huu Hue, Naoufal Saadaoui, Van Thien Nguyen (Insa Ctr. Val de Loire, Blois, France), Marc Lethiecq (Université de Tours, Bat E, 20 Ave., Monge, Parc de Grandmont, Tours 37000, France, marc.lethiecq@univ-tours.fr), and Jean François Saillant (Areva, Chalon/Saone, France)

Ultrasonic projectors are new type ultrasonic transducers that take advantage of the ultrasonic wave emitted on the rear face of the transducer. It consist in an acoustic symmetrical design where both waves generated on the faces of the piezoelectric ceramic are collected in phase with a 45° angle wedge resulting in an increase of the sensitivity. In this case, the absence of backing endows that the conventional design rules of a transducer cannot be directly applied. In this work, the electroacoustic modeling and characterization of such projector is reported. Using the KLM well known model, the electroacoustic response of a projector in based on a piezoelectric ceramic with one matching layer is calculated. Simulations show that the optimal acoustic impedance of the matching layer should be lower than for conventional transducers leading to a bandwidth of approximately 50%. The characterizations of such transducer based on PZ27 (Meggit/Ferroperm) and a lead free material PIC700 (PI Ceramic) have been carried out. Bandwidth and sensitivity are reported. They are found close to the simulation results and demonstrate that these new types of transducers has to be designed according new trends compared to conventional transducers.

11:00

2aPA6. An acoustic approach to assess natural gas quality in real time. Andi Petculescu (Univ. of Louisiana at Lafayette, 240 Hebrard Blvd., Lafayette, LA 70503, andi@louisiana.edu)

The natural gas industry needs fast and robust techniques to monitor the quality of the natural gas, if possible during the flow measurement process. A potential technique relies on extrapolating the dynamic specific heat of the gaseous mixture via measurements at a few frequencies (Petculescu and Lueptow, *Phys. Rev. Lett.* **94**, 238301 (2005); Sensors and Actuators B: Chemical **169**, 121–127 (2012)). At the core of this approach lies a first-principles model for sound absorption and dispersion in polyatomic gases. We will show and discuss the applicability and limitations of this potential technique to predicting the content of nitrogen, oxygen, carbon dioxide, and water in natural gas.

11:20

2aPA7. Investigating the effects of formulation and processing conditions on the mechanical properties of wheat flour noodle dough. Reine-Marie Guillermic, Sébastien O. Kerhervé (Dept. of Phys. and Astronomy, Univ. of MB, Allen Bldg., Winnipeg, MB R3T 2N2, Canada, reine-marie.guillermic@umanitoba.ca), Huiqin Wang (Dept. of Food Sci., Univ. of MB, Winnipeg, MB, Canada), Anatoliy Strybulevych (Dept. of Phys. and Astronomy, Univ. of MB, Winnipeg, MB, Canada), Dave W. Hatcher (Canadian Grain Commission, Grain Res. Lab., Winnipeg, MB, Canada), Martin G. Scanlon (Dept. of Food Sci., Univ. of MB, Winnipeg, MB, Canada), and John H. Page (Dept. of Phys. and Astronomy, Univ. of MB, Winnipeg, MB, Canada)

Characterization of the mechanical properties of soft food materials is crucial in the food industry, both for process design and for quality enhancement purposes. The goal of this project is to develop a non-contact on-line quality control technique for use in processing of sheeted products such as Asian noodles. In the case of the Asian noodle industry, composition and work input during the sheeting process are important parameters that influence the mechanical properties of the dough, and as a consequence, final product quality. An accurate and fast determination of noodle properties and how they are influenced by formulation and processing parameters will certainly improve quality control during production. To address this need, we are conducting a full study of noodle dough with ultrasonic techniques. A non-contact ultrasonic technique in transmission is used to assess the mechanical properties of noodle dough sheets and has been evaluated in pilot-plant trials, showing its feasibility for real-time quality control. Additionally, for a better understanding of the material properties, including how both bubbles and the dough matrix are affected by sheeting, contact longitudinal experiments over a wide range of frequencies (0.5 to 12 MHz), along with ultrasonic shear experiments, have been performed.

11:40

2aPA8. Ultrasonic investigation of fatigue, tensile damage, and impact damage in woven glass-fiber fabric reinforced composites with comparison to X-ray tomography. Nico F. Declercq (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France, nico.declercq@me.gatech.edu), Pascal Pomarède (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France), Othmane Ez-Zahraouy (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, Metz, France), Nada Miqoi (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France), Lynda Chehami (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, Metz, France), Fodil Meraghni, and Laurent Peltier (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France)

This work was done on woven glass-fiber fabric reinforced composite samples. Those materials exhibit a complex anisotropic evolution of defects induced by several damage mechanisms. In order to non-destructively evaluate the damage accumulation within this material, a methodology based on the measurement of the complete stiffness tensor is considered. After validation of the detectability of increasing damage state with this method, a new damage indicator is proposed to thoroughly quantify it. Samples were damaged by tensile tests (quasi-static and fatigue) at increasing stress levels along and out-of fibers axis. Afterwards, drop-weight impact is performed to consider several damage situations. Finally, an X-ray tomography is conducted to identify the damage mechanisms as well as the evolution of the void volume fraction. It is shown that this evolution has the same tendency with the ultrasonic damage indicator.

TUESDAY MORNING, 5 DECEMBER 2017 — SALON A/B/C, 8:30 A.M. TO 11:45 A.M.

Session 2aSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Engineering Acoustics: Acoustic Metamaterials I

Christina J. Naify, Cochair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr., Pasadena, CA 91109

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

8:30

2aSA1. On time-dependence of mechanical metamaterials properties. Massimo Ruzzene and Giuseppe Trainiti (Daniel Guggenheim School of Aerosp. Eng., Georgia Inst. of Technol., 270 Ferst Dr., Atlanta, GA 30332, ruzzene@gatech.edu)

The study focuses on wave propagation in time-periodic mechanical matamaterials, such as beams with time-modulated elastic modulus. The time-dependence of the elastic modulus results in a system with a frequency-periodic dispersion diagram. Flat bands originate at the intersections of the dispersion branches, signaling zero group velocity modes. Through an analytic approach, we obtain approximate expressions for the range of wavenumbers associated to flat bands, as well as their dependence on the modulation parameters. It is shown that the flat bands lead to unbounded growth of the displacement field, which is due to the interaction between the structure's dynamics and the energy provided to it by means of the elastic modulus variation. If such phenomenon is properly confined over a limited time window, it provides a way to filter wave propagation over a ultra-wide frequency range by property choosing the modulation parameters. Numerical simulations, obtained by implementing a FDTD scheme, verify the theoretical findings. Finally, an experimental validation of the results is discussed, which employs piezoelectric patches and time-dependent negative capacitance shunted circuits.

8:50

2aSA2. A metamaterial design method applied to topologically protected mechanical metamaterials. Kathryn Matlack (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801, kmatlack@illinois.edu), Marc Serra-Garcia (Inst. for Theor. Phys., ETH Zürich, Zurich, Switzerland), Antonio Palermo (Dept. of Civil, Chemical, Environ. and Mater. Eng. - DICAM, Univ. of Bologna, Bologna, Italy), Sebastian D. Huber (Inst. for Theor. Phys., ETH Zürich, Zurich, Switzerland), and Chiara Daraio (California Inst. of Technol., Pasadena, CA)

Mass-spring models provide a straightforward way to describe complex dynamic behavior of mechanical systems, such as topologically protected and backscattering-free surface phonons. Such discrete models also provide a means to translate these properties, based on the unique class of electronic materials of topological insulators, to the mechanical domain. This talk will discuss a systematic design process to take such a mass-spring model and implement its functionality directly into a metamaterial. The design method uses

combinatorial searches plus gradient-based optimizations to determine the configuration of the metamaterial's building blocks that replicates the behavior of the target mass-spring model. A key aspect of the design method is the use of "perturbative metamaterials"—i.e., metamaterials with weakly interacting unit cells. The weak coupling enables the design space to be divided into smaller sub-spaces that can be independently tuned, which results in an exponential speed-up of the design process. It also allows us to isolate the interactions between vibrational modes of its unit cells within the frequency range of interest, simplifying the inverse problem. The design method is illustrated through the example of engineering a 2D mechanical metamaterial that supports topologically protected and defect-immune edge modes, directly from the mass-spring model of a topological insulator.

9:10

2aSA3. Active acoustic metamaterials using sensor-driver architecture. Bogdan Ioan Popa (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu)

Acoustic metamaterials can, in principle, overcome the fundamental limitations of passive structures and can enable novel applications that are hard or impossible with passive media. However, active acoustic metamaterials remain elusive, and the few implementations going beyond basic material parameter tunability lag significantly behind the true potential of active media. We will present a design method that results in active metamaterials whose acoustic response is electronically programmed to cover a very large range of non-linear acoustic responses. Specifically, metamaterials obtained with this approach can be configured to have almost any second and higher harmonic response, making them very effective and versatile non-linear acoustic media. We demonstrate experimentally this design methodology in an application in which an acoustically thin metamaterial funnel responds to incident sound waves by producing collimated second harmonics. Past demonstrations of this type of active metamaterials will also be briefly reviewed.

Contributed Papers

9:30

2aSA4. Design and measurement of an acoustic bi-anisotropic metasurface for scattering-free manipulation of the refracted wavefront. Junfei Li, Chen Shen (ECE, Duke Univ., 101 Sci. Dr., Rm. 3417, FCIEMAS Bldg. Duke Univ, Durham, NC 27708, junfei.li@duke.edu), Ana Díaz-Rubio, Sergei Tretyakov (Dept. of Electronics and NanoEng., Aalto Univ., Aalto, Finland), and Steven Cummer (ECE, Duke Univ., Durham, NC)

Relying on the generalized Snell's law which defines a phase gradient to the surface, extraordinary control of the reflected and transmitted wavefronts can be achieved by metasurfaces with subwavelength thickness. However, a fundamental limitation for such metasurfaces is their power transmission efficiency, especially at large deflection angles. Building on the theoretical requirements for the boundary conditions at an ideal metasurface, we designed and fabricated the necessary bi-anisotropic cells for a wave-front transformation acoustic metasurface that overcomes the fundamental limits of conventional designs, allowing us to steer the energy flow in an arbitrary manner without parasitic scattering. The design with a discretized structure is verified both numerically and experimentally and an energy conversion efficiency of 93% is observed for the transmitted wave into the desired direction of 60 degrees, higher than the corresponding theoretical upper limit in generalized Snell's law based designs (89%). Our experimental demonstration confirms the theoretical results and opens a new way of designing practical and highly efficient metasurfaces for different functionalities, allowing nearly ideal control over the energy flow through metasurfaces.

9:45

2aSA5. Passive acoustic metasurfaces for high-efficiency wavefront transformations. Li Quan and Andrea Alu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu)

Metasurfaces have offered new degrees of freedom to control the propagation properties of acoustic waves. Gradient metasurfaces, first proposed in optics, and recently extended to acoustics, add an extra transverse momentum to the incident plane wave so that reflected and refracted waves can be rerouted without having to rely on conventional Snell's Law. However, recent studies have shown that gradient metasurfaces cannot transfer all incident energy into the desired direction without causing unwanted parasitic diffraction and limiting the overall transformation efficiency. More sophisticated metasurface designs have considered impedance matching to address this issue, but this inherently requires the use of active and lossy materials for extreme wave transformations. In this talk, we present our recent efforts in designing passive acoustic metasurfaces with unitary efficiency in beam steering, even when considering extreme steering angles. We consider nonlocal surface impedance profiles and the use of bianisotropic inclusions, which are both effective routes to realize efficient metasurfaces without relying on gain elements. Our work opens new design approaches for metasurfaces, and it can improve the operational efficiency of metasurfaces.

10:00

2aSA6. Anomalous scattering effects from bi-anisotropic inclusions and metamaterials. Li Quan and Andrea Alu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu)

The acoustic scattering properties of small particles are typically dominated by weak monopolar or dipolar effects, associated with their nature and geometry. As large collections of them form materials, these features determine the overall response to sound of the object we have around. Recent work on acoustic bi-anisotropic acoustic inclusions has shown that, by breaking geometrical and time-reversal symmetries, one can realize constitutive relations that couple pressure fields to dipole radiation and velocity field to monopole radiation in non-trivial ways. In this talk, we use these properties to induce anomalous scattering properties in small inclusions, including unidirectional scattering cancellation, non-reciprocal scattering responses, and directive or sharp scattering resonances. We also consider the use of active acoustic particles to further extend the available degrees of freedom in scattering engineering, and explore unusual parity-time symmetric responses in acoustics. By combining several of these complex inclusions to form acoustic metamaterials, we also investigate the unusual material responses offered by this design approach, which goes beyond what available in natural materials, offering new degrees of freedom for acoustic engineering.

10:15–10:30 Break

10:30

2aSA7. Asymmetric transmission via lossy gradient-index metasurfaces: Role of diffraction. Chen Shen (Elec. and Comput. Eng., ECE, Duke Univ., Durham, NC 27705, chen.shen4@duke.edu), Yong Li (Inst. of Acoust., School of Phys. Sci. and Eng., Tongji Univ., Shanghai, China), Yangbo Xie, Junfei Li (ECE, Duke Univ., DURHAM, NC), Yun Jing (Mech. Eng., North Carolina State Univ., Raleigh, NC), and Steven Cummer (ECE, Duke Univ., Durham, NC)

The development of acoustic metasurfaces has enabled numerous wave control abilities. The effect of losses in acoustic metasurfaces for sound transmission manipulation, however, is largely unexplored. Here, we show that robust asymmetric transmission can be achieved by harnessing

judiciously tailored losses. Theoretical investigations show that the asymmetric behavior stems from loss-induced suppression of high order diffractions. Multiple reflections occur inside the individual slits for the negative incident direction, which lead to different orders of diffraction and asymmetric responses for opposite directions. Numerical simulations based on effective medium are performed and are in good agreement with theoretical analysis. Real structures based on unit cells with designed internal viscous loss are fabricated and measured in a 2D waveguide. The peak energy contrast is about 10 times in a certain range of incident angles and frequencies. This study may open up new possibilities in lossy acoustic metamaterials and metasurfaces. The theory can also be readily extended to electromagnetic waves.

10:45

2aSA8. Optimal attenuation caused by boundary impedance in acoustic waveguides. Matthew Kelsten and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu)

Non-Hermitian systems can exhibit "exceptional points" (EPs) at which modes coalesce. The connection between EPs and acoustic damping goes back to the observation of Cremer (1953) that optimal attenuation in a duct occurs when the two lowest modes have equal complex-valued eigenvalues, although the physical basis for this effect remains unclear. In an attempt to understand Cremer's observation, we consider the model case of a two-dimensional waveguide with different impedance conditions on the two boundaries. This allows us to determine the complete set of all possible pairs of passive impedance conditions that give rise to EPs, and from these to select impedances appropriate to a particular frequency bands. The non-separable, and generally non-symmetric, mode shapes are described. The theoretical findings are linked to realistic passive impedance values based on various models for boundary impedance, such as perforated and porous panels. These comparisons are discussed to illustrate the feasibility of optimized wall impedances in absorbing sound passing through ducts. [Work supported by NSF.]

11:00

2aSA9. Thermoviscous effect on sound propagation in acoustic metastructures. Likun Zhang, Xudong Fan (Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, zhang@olemiss.edu), Xue Jiang (Nanjing Univ., Nanjing, China), and Yong Li (Tongji Univ., Shanghai, China)

Thermoviscous dissipation plays an important role in sound propagation through acoustic metastructures for a variety of applications. The thermoviscous effects rely on the resonances in the structures of subwavelength features. Our analyses and numerical simulations reveal that thermoviscous dissipation can significantly reduce the transmission of sound waves through acoustic metastructures of hybrid resonances in certain scenarios even when the thickness of the viscous boundary layer is much smaller than the width of the slit in the structure [Jiang *et al.*, *J. Acous. Soc. Am.* (2017)]. To further explore how the resonances affect the significance of the losses, we analyze and simulate sound propagation in acoustic metamaterials of various configurations, including a slit connected to one or multiple resonators at the side of the slit or at the end of the slit. We take into account both wall friction and thermoviscous diffusivity in both the slits and the resonators. The results are compared with that without the resonators to gain insight into the role of the resonances. Optimal designs for both minimal dissipation and maximal absorption are addressed.

11:15

2aSA10. Acoustics of locally bilinear periodic metamaterials. Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, alexey.titovich@navy.mil)

The scope of traditional metamaterials is expanded by analyzing bilinear discontinuities in an otherwise continuous medium. This amplitude independent nonlinearity is sought for its practicality and utility. Wave propagation has to be solved for in the time domain due to the difference in energy transfer from compression to rarefaction across an interface. Various sizes, orientations and shapes of the discontinuities are analyzed on a unit cell level. Periodic chains with the same type of material discontinuity have recently been shown to exhibit a rich dispersion behavior, such as negatively sloped dispersion branches [Titovich (2017), *J. Acoust. Soc. Am.* **141**(5), 3698]. These waves are now investigated in a continuum, where the discontinuity is modelled as displacement-dependent complex impedance relating the interface tractions. The onset of chaotic behavior as well as its effect on absorption is discussed. Also, the ability of such periodic discontinuities to produce nonlinear non-reciprocity is investigated.

11:30

2aSA11. Closed-cell hyperelastic elements with mechanical instabilities and structural negative stiffness. Stephanie G. Konarski and Michael R. Haberman (Appl. Res. Labs. & Dept. of Mech. Eng., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

Metamaterials are subwavelength structures that generate enhanced or exotic properties that are unattainable with conventional materials. One such class of metamaterials utilizes inclusions with mechanical instabilities to generate a non-monotonic pressure versus strain response and thus regions of structural negative stiffness. By embedding these sub-wavelength structures in a lossy matrix material, it is possible to tune and enhance material properties of the mixture such as acoustic and elastic nonlinearity and energy dissipation. Previous theoretical work on this topic by the authors has focused on a dilute concentration of these types of hyperelastic inclusions within a fluid or nearly incompressible elastic matrix material to determine the overall response via both quasi-static and dynamic nonlinear homogenization methods. The present work focuses on numerical analysis using the finite element method for closed-cell elements that display non-monotonic pressure-strain behavior. The objective is to design sub-wavelength, hyperelastic inclusions that are easy to embed in a fluid or polymeric background material in an effort to generate elevated parameters of nonlinearity or increased capacity to dissipate vibro-acoustic energy. Emphasis is placed on structures that can be fabricated using additive manufacturing techniques. [This work was supported by the Office of Naval Research.]

Session 2aSC

Speech Communication: Articulatory and Acoustic Characteristics of Nasalization

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Chair's Introduction—9:00

Invited Papers

9:05

2aSC1. Speaker and production characteristics of posterior nasal fricatives. David J. Zajac (Univ. of North Carolina at Chapel Hill, CB# 7450, UNC Craniofacial Ctr., Chapel Hill, NC 27599, david_zajac@dentistry.unc.edu)

Posterior nasal fricatives (PNF) are unusual articulations produced by children both with and without velopharyngeal (VP) anomalies to replace oral fricatives. PNFs are produced by occluding the oral cavity and forcing airflow through a partially closed VP port to create turbulent noise. Often, extra noises occur due to displacement of mucous and/or tissue flutter. The headset of the nasometer was used to record the separate oral and nasal acoustic signals from 18 children who produced PNFs. They ranged in age from 4 to 15 years, 11 were males, and 11 had some form of cleft palate. Examination of the oral acoustic waveforms revealed that all children produced a stop gesture during intended fricative targets—typically a lingual-alveolar or palatal articulation. Sixteen of the children produced a clear flutter (raspberry-like) noise as evidenced by quasi-periodic components in the nasal spectra. All children used PNFs to replace /s/, 13 (72%) to replace /z/, 12 (67%) to replace /tʃ/, and 6 (33%) to replace /ʃ/ and /dʒ/. Most of the children had known histories of conductive hearing loss. Most of the children with cleft palate had adequate VP closure during stop consonant production. Possible causes of PNFs are discussed.

9:25

2aSC2. The relation of auditory perceptual ratings of nasality to nasal port area in connected speech. Brad H. Story and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Modeling can be used to understand the relation of articulatory movement to the acoustics and perception of speech. Published studies of velopharyngeal orifice size and perceived nasality in clinical populations have not shown a clear relationship between the measures. Two reports based on computational modeling, however, have found a high correlation between ratings of nasality and nasal port area (Bunton & Story, 2012; Bunton, 2013). These studies were based on sustained vowel simulations with nasal port areas ranging from 0 to 0.5 cm^2. Expert listeners were able to detect nasality in vowel samples with a nasal port area greater than 0.01 cm^2 and nasality rating plateaued with areas greater than 0.16 cm^2. The present study extends this work by reporting on nasality ratings based on short connected speech samples simulated with varying nasal port areas. The connected speech samples included either obstruent consonants or approximant consonants to examine whether the relationship between perception and nasal port area varies based on the pressure demands of the speech sounds produced. This work allows for a more complete explanation of the relation between auditory perceptual ratings of nasality and nasal port area. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

9:45

2aSC3. Using oropharyngeal articulation to compensate for nasalization: Acoustic and perceptual evidence. Panying Rong (Speech-Language-Hearing: Sci. & Disord., Univ. of Kansas, 1000 Sunnyside Ave., Lawrence, KS 66045, prong@ku.edu), David P. Kuehn (Speech and Hearing Sci., Univ. of Illinois, Champaign, IL), and Ryan Shosted (Linguist, Univ. of Illinois, Urbana, IL)

A variety of oropharyngeal articulatory differences have been identified between oral and nasal vowel pairs in different languages, which either enhance or reduce the acoustic effects of nasalization. The aim of this study is to simulate oropharyngeal articulatory adjustment strategies to maximally compensate for the acoustic effects of nasalization. Two articulatory models were used for the simulation—one speaker-independent (SI) model and one speaker-dependent (SD) model. The results suggested that (1) the oropharyngeal articulatory strategies generated by both models effectively reduced the acoustic effects of nasalization on the low-frequency spectrum of the vowels; and (2) the oropharyngeal articulatory strategies generated by the SD model provided a better compensatory effect in the spectral region of F2 and higher formants compared to the strategies generated by the SI model. Moreover, a comparison of nasality ratings of the synthetic nasal vowels with and without oropharyngeal articulatory adjustments showed a significant reduction of nasality following oropharyngeal articulatory adjustments. These findings provide acoustic and perceptual evidence in support of motor equivalence. With further research on other speech sounds and the temporal pattern of nasalization warranted, the findings might have potential clinical implications for reducing hypernasality in persons with resonance disorders.

10:05

2aSC4. Articulatory insights on the evolution of nasal vowels in Slavic. Ryan Shosted (Linguist, Univ. of Illinois at Urbana-Champaign, 4080 FLB MC-168, 707 S Mathews Ave., Urbana, IL 61801, rshosted@illinois.edu)

The complex articulatory-acoustic mapping of the vocal tract suggests that the acoustic effects of nasalization are tied to other articulatory mechanisms besides velopharyngeal opening; this has been borne out in many recent studies. But do these findings settle on any consensus regarding the articulatory mechanisms (besides velopharyngeal opening) that differentiate nasal and oral vowels and cause shifts in vowel inventories across time? Do articulatory results mirror or contradict the relatively larger corpus of findings regarding the acoustics of nasalization? Are these results idiosyncratic reflections of language differences or do they represent universal properties of phonological development? To approach these questions, results of a series of magnetic resonance and electromagnetic articulography studies of French, Portuguese, and Hindi are compared to historical sound changes in Slavic, a language family where most nasal vowels are now lost. Nasal vowels have been implicated in a number of diachronic Slavic vowel shifts, particularly the lowering of nasal /i/ and nasal /e/ and the raising of nasal /a/ in the development of Late Middle Slavic. Height-related phenomena are well-attested in the articulatory studies and seem predictive of Slavic changes; however, frontness changes, also attested in the articulatory studies, do not find clear Slavic counterparts.

10:25–10:45 Break

Contributed Papers

10:45

2aSC5. Articulatory correlates of phonemic and coarticulatory nasalization. Marissa Barlaz, Zhi-Pei Liang, and Brad Sutton (Linguist, Univ. of Illinois at Urbana-Champaign, 707 S. Mathews Ave., MC 168, Urbana, IL 61820, goldrch2@illinois.edu)

Phonological theory distinguishes nasal and oral vowel counterparts by velopharyngeal port opening, neglecting other phonetic differences between phonemic and coarticulatory nasalization. Recent articulatory work provides evidence of oropharyngeal distinctions, in addition to velic lowering. This study (12 Brazilian Portuguese speakers) uses real-time MRI to investigate oropharyngeal differences between oral, phonemically nasal, and phonetically nasalized vowels /a, i, u/. Tissue boundaries in midsagittal vocal tract images were automatically detected to reveal each vowel repetition's aperture function. Principal Components Analysis determined vocal tract regions responsible for the greatest variance in the data. Time-dynamic analyses of vocal tract area in these regions used smoothing spline ANOVA. Results show the tongue body and/or hyperpharynx as the most important articulators. For /a/, nasal vowels demonstrate wider hyperpharyngeal and narrower tongue body regions compared to oral vowels. For /u/, oral vowels show wider hyperpharyngeal and narrower tongue body regions. For /i/, nasal vowels demonstrate wider tongue body and narrower hyperpharyngeal regions. Nasalized vowels manifest apertures intermediate between oral and nasal vowels for /a/ and /u/, and similar to oral vowels for /i/. Results are largely in line with expected acoustic effects of nasalization, and demonstrate that phonetic differences exist between phonemic and coarticulatory nasalization.

11:00

2aSC6. Spontaneous nasalization after glottal consonants in Thai: An rt-MRI investigation of rhinoglottophilia. Sarah E. Johnson (Linguist, Univ. of Illinois at Urbana-Champaign, 508 S First St. Apt. 403, Champaign, IL 61820, sjohnson2052@gmail.com), Brad Sutton (BioEng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Zhi-Pei Liang (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The affinity between glottal and nasal articulations ("rhinoglottophilia") facilitates spontaneous nasalization. In Thai, low vowels nasalize after /h/ and to a lesser degree after glottal stop. Nasalization after /h/ may occur because breathiness and nasalization share high energy at low frequencies and a raised first harmonic. Glottal consonants generally may cause nasalization because aerodynamically they do not require a closed velopharyngeal port. We investigated whether Thai vowels after /h/ and glottal stop exhibit similar degrees of velopharyngeal opening (VPO) and compared these results with acoustic measures of nasalization / breathiness. We calculated nasalization / breathiness by the energy ratio of low and high harmonics; we measured VPO by processing oblique real-time magnetic resonance images of the velopharyngeal port. Four Thai speakers (two females, two males) produced relatively large VPO and high acoustic nasalization / breathiness after /h/. Female speakers nasalized vowels after glottal stop, though they produced overall less VPO and lower acoustic nasalization / breathiness when compared to vowels after /h/. In Thai, vowels after /h/ exhibit more physiological nasalization than vowels after glottal stop; furthermore, low VPO is associated with low acoustic nasalization / breathiness. We conclude VPO is primarily responsible for impressions of nasalization in this context.

11:15

2aSC7. Nasal rustle: An evidence-based description. Michael Rollins (Biomedical Eng., Univ. of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267, rollinmk@mail.uc.edu), Liran Oren (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Ann W. Kummer (Speech-Lang. Pathol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), J. P. Willging (Velopharyngeal Insufficiency Clinic, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and Suzanne E. Boyce (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

"Nasal rustle" (also known as "nasal turbulence") is a loud nasal distortion that can be heard during speech production in certain children with velopharyngeal insufficiency (VPI). It occurs when there is a leak of airflow into the nasal cavity through a small velopharyngeal opening. While the perception of audible nasal emission—including nasal rustle—is a standard means for the diagnosis of VPI, there is no consensus on the sound generation mechanism. Current hypotheses include aerodynamic turbulence, velar flutter, and bubbling of mucus secretion. This study investigates the correlation between the acoustic signal of nasal rustle and physical movement at the superior velopharyngeal port. Several pediatric VPI patients were recorded via high-speed video nasopharyngoscopy and simultaneous nasometry during production of speech sounds susceptible to nasal rustle. Instances of perceived nasal rustle in the acoustic signal were identified and compared with the high-speed video. A high correlation was found between the bubbling frequency of mucus secretion above the velopharyngeal port and frequencies in the nasal acoustic signal, suggesting that secretion bubbling is a mechanism for generating nasal rustle. From this analysis, an integrated description of nasal rustle invoking physiological, physical, and acoustic principles is proposed.

Session 2aSP

Signal Processing in Acoustics and Underwater Acoustics: Detection, Classification, Localization, and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) I

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Chair's Introduction—7:55

Invited Papers

8:00

2aSP1. Detection in an uncertain underwater waveguide. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu) and Ali Abdi (Elec. and Comput. Eng., New Jersey Inst. of Technol., Newark, NJ)

Detection of signals in oceanic waveguides is complex because of uncertainties in the underwater channel and potentially suboptimal location of receiving phones. Accurate detection and estimation are a direct result of optimization with respect to frequency, environmental parameters, and receiver location. We here study the impact of these parameters on detection and propose methods for improvement, including detection processor calibration for optimal performance. We complement our work by investigating detection performance with real data collected at different sensors in a pool. In addition to conventional hydrophones, we also use vector sensors. The rationale behind using a vector sensor is that particle velocity components may show different noise and channel characteristics such as a different number of modes and different dispersive behavior, which are tightly related to detection. We study several detectors and evaluate their performance. [Work supported by ONR and NSF.]

8:20

2aSP2. Achieving the transparent ocean. Peter J. Stein (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pstein@scisol.com)

Our common impression is that radar systems monitor the vast majority of the airspace above the earth's surface. Safe travel is nearly taken for granted, and for the purposes of national security there is a constant close eye on the comings and goings through our airspace. The same cannot be said for under the ocean's surface, from way out at sea, to near our coast, and within our harbors and waterways. The need for the long-coveted "Transparent Ocean" is evident, and in this paper we explore the visionary aspects of achieving this goal. This includes reviewing concepts and a framework around which state-of-the-art often stove-piped techniques in underwater acoustics, signal processing, and acoustical oceanography, along with non-acoustic modalities including radar and EOIR, are combined into an integrated system.

8:40

2aSP3. Effect of beamwidth on detection of near-bottom targets with multibeam echosounders. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org)

In multibeam echosounding, beamwidth and sidelobe levels constrain detection of echoes from near-bottom targets that are often masked by echoes from the bottom received in the mainlobe of individual beams or through their sidelobes. Acoustic backscatter data collected with multibeam echosounders from various manufacturers indicate that the customary half power point (−3 dB) metric for the angular width of individual beams underestimates the masking effects of bottom echoes. A better fit of the detected bottom echo trace is obtained with beamwidths estimated 10 dB below the maximum response of each beam. The corresponding beamwidths are roughly 66% larger at −10 dB than at −3 dB for a canonical sinc squared beam pattern. Therefore, using the manufacturers' nominal −3 dB beamwidth specifications yields underestimates by the same percentage of the area ensonified by the pulse within a beam, and corresponding overestimates of the detected acoustic backscatter level used to infer target strength or bottom backscatter coefficients.

Contributed Paper

9:00

2aSP4. The research of a frequency-domain adaptive matched filter. Juan Hui (Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn)

The most classical detector of active underwater acoustic is the matched filter (MF), which is the optimal processor under ideal conditions. Aiming at the problem of active sonar detection, we propose a frequency-domain adaptive matched filter (FDAMF) with the use of a frequency-domain adaptive line enhancer (ALE). The FDAMF is an improved MF. In the simulations in this paper, the signal to noise ratio (SNR) gain of the FDAMF is about 18.6 dB higher than that of the classical MF when the input SNR is −10 dB. In order to improve the performance of the FDAMF with a low input SNR, we propose a pre-processing method, which is called frequency-domain time reversal convolution and interference suppression (TRC-IS). Compared with the classical MF, the FDAMF combined with the TRC-IS method obtains higher SNR gain, a lower detection threshold, and a better receiver operating characteristic (ROC) in the simulations in this paper. The simulation results show that the FDAMF has higher processing gain and better detection performance than the classical MF under ideal conditions. The experimental results indicate that the FDAMF does improve the performance of the MF, and can adapt to actual interference in a way.

Invited Papers

9:15

2aSP5. The hybrid Cramer-Rao bound of direction finding by a uniform circular array of isotropic sensors that suffer stochastic dislocations. Zakayo N. Morris, Kainam Thomas Wong (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., BC 606, Hung Hom KLN, Hong Kong, zakayo.morris@connect.polyu.hk), Dominic M. Kitavi (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., Kowloon, HungHom, Hong Kong), and Tsair-Chuan Lin (Dept. of Statistics, National Taipei Univ., New Taipei City, Taiwan)

Consider azimuth-elevation direction finding by a uniform circular array of isotropic sensors. In the real world, the sensors may dislocate from their nominal positions. These dislocations could be modeled as random variables having an a priori known distribution. This paper investigates how the dislocations would affect azimuth-elevation direction finding by deriving the corresponding hybrid Cramer-Rao bounds. Maximum a posteriori estimators are derived and Monte Carlo simulations are conducted to validate the derived hybrid Cramer-Rao bounds.

9:35

2aSP6. Angle estimation using a moving PV sensor: Performance assessment. Edmund Sullivan (EJS_Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

It is well known that bearing estimation performance, using a moving line array of pressure sensors, can be enhanced by using a Kalman filter to jointly estimate the bearing and source frequency. This is because the bearing dependence of the Doppler can be exploited when the source frequency is known. However, it is also known, based on an observability analysis, that it cannot be done using a single moving pressure sensor. Here, it is shown that if a single moving pressure-vector sensor is used, both the bearing and roll angle can be estimated. If only the vector sensor is used, estimation of these angles can still be done, but the performance is poor. The addition of the pressure sensor allows the motion to be exploited, thus significantly enhancing the performance. This phenomenon is shown theoretically by using a Bayesian Cramer-Rao lower bound calculation. An example using simulated data is shown where the improvement is clearly demonstrated.

9:55–10:10 Break

10:10

2aSP7. Intelligent active sonar via tuning of transmit waveform and detection threshold. Jill K. Nelson (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr. MSN 1G5, Fairfax, VA 22030, jnelson@gmu.edu) and Steven Schoenecker (Naval Undersea Warfare Ctr., Newport, RI)

We consider active sonar systems that translate high-level tasks to a set of parameter adaptations and arbitrate among competing demands, thereby incorporating cognitive processing in the system. We propose adopting the goal-driven autonomy (GDA) architecture presented in Klenk *et al*. [*Computational Intelligence*, May 2013] to realize intelligent active processing. Using the GDA architecture, the sonar system uses its observations to inform how system parameters are tuned to achieve a set of surveillance goals. In addition, the system is able to reason about its actions, identifying discrepancies between predicted and observed performance and modifying both parameters and goals accordingly. In this work, we focus on a tracking task and consider the transmit waveform and detection threshold as tunable parameters. We describe an active sonar system that tunes these two parameters based on observations of the physical environment and target characteristics, as well as current goals. The intelligent system uses its observations of the environment to generate estimates of target Doppler and of clutter density to tune waveform and detection threshold, respectively. Through simulation, we demonstrate the performance improvement achieved by the cognitive system when tracking a maneuvering target that is traveling through spatially non-uniform clutter.

10:30

2aSP8. Littoral Acoustic Demonstration Center passive acoustic localization and tracking of marine mammals in the Northern Gulf of Mexico. Juliette W. Ioup (Dept. of Phys., Univ of New Orleans, New Orleans, LA 70148, jioup@uno.edu)

The Littoral Acoustic Demonstration Center (LADC) consortium, founded in 2000, has collected underwater acoustic data in the northern Gulf of Mexico since 2001 using Environmental Acoustic Recording Systems (EARS) buoys. These hydrophone systems were developed by the U.S. Navy, and originally LADC could record underwater signals only up to 11.7 kHz. Since then the EARS have been upgraded several times and currently record up to 192 kHz, thus allowing recordings of not only sperm whale clicks and codas, but also signals of beaked whales and dolphins. One goal of LADC is to develop an acoustic method for identification of individual animals, with the goal of an "acoustic catalog" similar to the existing photographic catalog of individual whale flukes. Localization and tracking is an important component for acoustic identification. Some localization and tracking techniques and procedures developed and used by LADC will be presented and discussed.

10:50

2aSP9. Incorporating image sources in the time-domain beamforming model for the localization of sound sources in reverberant rooms. Alice V. Lam, Murray Hodgson (Dept. of Mech. Eng., Univ. of Br. Columbia, 6250 Appl. Sci. Ln., Vancouver, BC V6T 1Z4, Canada, avl@alumni.ubc.ca), and Vincent Valeau (Institut PPRIME, Université de Poitiers-ENSMA-CNRS, Poitiers Cedex 9, France)

In this study, we use a small hemispherical microphone array to locate impulsive or stationary sound sources in reverberant rooms. A modified version of the conventional time-domain beamforming model was used, which incorporates physical information from the early specular reflections of the sound source, as well as the geometric and absorption characteristics of the room. In other words, the propagation model takes into account the potential image sources up to a given image order. We show that the technique can be seen as a time-reversal process, the measured pressure field being simultaneously re-emitted from both the actual array position and its virtual image positions. We then present results applying this modified model to simulated data (generated using an image-source propagation model), as well as measurements in real rooms, using both impulsive and continuous broad-band noise. In all cases, the modified model shows an improvement in localization over conventional beamforming. The issue of appropriately adjusting the image order and mixing time when performing the processing for optimal results is also addressed.

11:10

2aSP10. Detection localization and tracking of aircraft using Sparse distributed passive acoustic system. Alexander Sedunov, Hady Salloum, Alexander Sutin, Nikolay Sedunov (Maritime Security Ctr., Stevens Inst. of Technol., One Castle Point on Hudson, Hoboken, NJ 07030, asedunov@stevens.edu), and David Masters (Sci. and Technol. Directorate, DHS, Washington, DC)

Stevens Institute of Technology has conducted extensive long-term testing of acoustic systems designed to track low-flying small aircraft in remote location and recorded over 2 years of data. The system consisted of 4 nodes placed in difficult remote terrain with separation ranging 1–4 km, each node comprising a pyramid-shaped volumetric cluster of 5 microphones an embedded computer, and a pan-tilt-zoom camera steered to detected targets in real time. and communication device. Each nodes' computer performed direction of arrival finding communicated to a central computer collected that data and processed it to generate tracks and classify targets. The duration and the scale of the deployment allowed to identify and solve many problems, including the effects of propagation delays between station and on cooperative localization and tracking, the seasonal changes in environmental noise, persistent and transient noise sources, and the diversity of targets of opportunity and their signatures. The propagation delay effects led to development of separate trackers for review of target trajectories and for immediate action such as steering the camera. An overview of the algorithms is presented along with the long-term observations. [This work was funded by DHS's S&T Directorate.]

Contributed Papers

11:30

2aSP11. Inference regarding the state of mobile underwater object from a small aperture vertical array. Abner C. Barros and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, abarros1@umassd.edu)

A narrowband source ensonifies an area of interest with an oncoming submerged target in a refractive undersea environment. A vertical linear hydrophone array is employed to infer the depth, range, and speed of the target by resolving a direct path and one interacting surface path. Tracking the target by means of a continuous wave transmission is challenging due to the difficulty of inferring the frequencies and angles of the two returned closely spaced wave vectors. The complex propagation of sound in a refractive medium presents additional challenges for inversion of the wave vectors to range and depth. A Gibbs sampler is employed to construct the posterior joint density of all parameters taking full advantage of the analytic tractability of the conditional densities of the received amplitudes and phases and of the ambient acoustic noise power. The conditional densities of the ordered wave vectors however are constructed numerically by 2-dimensional inverse quantile sampling. The inferred joint posterior density of the target state is obtained by constructing an inverse transformation of the acoustic propagation model that accounts for the depth dependent sound speed. Simulation results demonstrate the approach at received signal to noise ratios below −3dB.

11:45

2aSP12. Compressive spectral estimation methods using a noise reference. Alexander S. Douglass and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu)

Compressive beamforming methods have shown promise for use in high resolution direction finding and source localization. However, these methods often fail in scenarios with low SNR, particularly if only a few snapshots are available. The Spectral Estimation Method (SEM) (Blacodon and Elias, 2004, J. Aircraft **41** (**6**), 1360–1369) provides a means of source localization and source power estimation by minimizing the mean squares error between cross-spectral-density matrices of a measured signal and modeled signals at each location of interest. SEM With Additive Noise (SEMWAN) (Blacodon, 2011, Applied Acoustics, **72**, 11–21) does the same but also utilizes the cross-spectral-density matrix of one or more noise references. In this presentation, the performance of a compressive version of SEMWAN is assessed for use in low SNR scenarios when the solution is sparse and a noise reference is available. Compressive SEMWAN is analyzed first using far-field, free-space simulations with varying SNR and multiple sources. Compressive SEMWAN is then applied to laboratory measurements taken with real noise sources at varying SNR and including some multipath propagation. The performance in each scenario is compared directly to conventional beamforming, a standard compressive beamforming method, and traditional SEMWAN. [Sponsored by NAVSEA through the NEEC.]

Session 2aUW

Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques I

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

Invited Papers

8:20

2aUW1. An overview of the Seabed Characterization Experiment 2017. Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu) and David P. Knobles (KSA, LLC, Austin, TX)

A multi-institutional, multi-disciplinary ocean acoustics experiment, Seabed Characterization Experiment 2017 (SCE2017), was conducted 95 km south of Martha's Vineyard, MA, USA, in March and April of 2017. Three research vessels and more than a dozen principle investigators conducted several types of experiments to improve our understanding of the acoustics of fine grained sediments, forward modeling of sound propagation in shallow water over fine grained sediments, and inverse and statistical inference processes used to characterize the seabed in such environments. The measurements included impulsive and tonal source tows received on vertical and horizontal line arrays, direct measurement of sediment bulk acoustic waves and interface waves, and supporting oceanographic observations. The 2017 experiment was preceded by survey cruises in 2016 and 2015 in which more than 200 sediment cores were obtained and detailed sub-bottom profiling was conducted. Here, an overview of the 2017 experiment is presented. [Work supported by ONR.]

8:40

2aUW2. Sound speed and attenuation in muddy sediments from low-frequency acoustic measurements. Lin Wan, Mohsen Badiey (Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu), D. P. Knobles (KSA, LLC, Austin, TX), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas a Austin, Austin, TX), and Justin Eickmeier (Univ. of Delaware, Newark, DE)

While there have been numerous theoretical and experimental studies on the properties of marine granular sands, there are significantly fewer studies on sediments classified as muds. The validity of geoacoustic models for muddy sediments has not been successfully tested due to the lack of inverted low-frequency sound speed and attenuation from acoustic measurements. The ONR-sponsored Seabed Characterization Experiment (SBCE), conducted in a mud patch on the New England continental shelf in the spring of 2017, provides an opportunity to make substantial improvements in understanding the physical mechanisms controlling sound propagation in muddy sediments. Acoustic signals (e.g., 31g explosive and combustive source signals) detonated at various ranges, depths and azimuths were measured in SBCE. This paper utilizes these measured signals to extract the acoustic normal mode characteristics including modal dispersive curve with Airy phase structure, modal amplitude, modal attenuation coefficient, and mode depth function. These normal mode characteristics are used in geo-acoustic inversion algorithms to estimate low-frequency sound speed and attenuation in muddy sediments as a function of frequency. The performance of different inversion methods using different normal mode characteristics is discussed. [Work supported by ONR Ocean Acoustics.]

9:00

2aUW3. Vector sensor observations of the 2 + mode pattern on the New England Mud Patch at 57 Hz. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng. Dept. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

We present recordings made by an Intensity Vector Autonomous Recorder (IVAR) deployed on the seafloor during the Sediment Characterization Experiment (SCE17) conducted on the New England Mud Patch [40°28' N, 70°35' W] in March of 2017. IVAR continuously and coherently records four channels of acoustic data, three from a tri-axial accelerometer embedded in a neutrally buoyant sphere and one from an omnidirectional hydrophone positioned 10 cm above the centroid of the sphere 1.2 m above the seafloor. Here we focus on IVAR recordings of a 57 Hz continuous wave tone generated by a low-frequency acoustic source (J-15) that was towed at 30 m depth by the R/V Endeavor along transects roughly parallel and perpendicular to bathymetric contours. Over the entire tow track deep signal fades occur in 1000 m intervals, suspected to be caused interference between two modes trapped by the sediment basement. A signature in the vector data identifies an additional interference structure superimposed on this pattern (a 2 + mode pattern at 200 m intervals) which fades when the source moves to the northeast of IVAR, where sediment surveys indicate thinning of the overlaying mud layer. Comparison with a mode based model provides an estimate of the sediment properties.

9:20

2aUW4. The Intensity Vector Autonomous Recorder (IVAR) as deployed in the Sediment Characterization Experiment. Peter H. Dahl and David R. Dall'Osto (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu)

The IVAR system (Intensity Vector Autonomous Recorder) is a bottom deployed system developed for first-use in the Sediment Characterization Experiment (SCE17), conducted on the New England Mud Patch [40°28' N, 70°35' W] in spring 2017. IVAR continuously and coherently records four channels of acoustic data, three from a tri-axial accelerometer embedded in a neutrally buoyant sphere (diameter 10 cm) and one from an omnidirectional hydrophone positioned 10 cm above the centroid of the sphere positioned 1.2 m above the seafloor. Despite operations being significantly impacted by two Gale-force storms, IVAR obtained 72 h of data over 2 deployments that included low frequency signals from a towed source (J-15) deployed from the R/V Endeavor and broad band SUS charges deployed from R/V Neil Armstrong. Interesting features of the intensity (Umov) vector field emerge in both bandwidth regimes, but here we focus on measurements of the 100+ SUS charges. The measured Umov vector for each SUS charge is first compared with the known source bearing, and is followed by an analysis of precursor arrivals and dispersive water-borne arrivals in the context of the received vector field. The source locations provide complementary information on range and azimuth dependence of the sediment properties.

9:40

2aUW5. Statistical inference for sediment parameters using ship noise and a horizontal array. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no), David P. Knobles (KSA LLC, Austin, TX), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper considers the use of broadband noise from ship sources of opportunity in statistical inference for geoacoustic parameters of a layered seabed, with applications to data collected with a bottom-moored horizontal array in the 2017 Seabed Characterization Experiment conducted on the New England Shelf. Statistical inference via a maximum entropy—Bayesian approach that determines the most appropriate model parameterization (number of seabed layers) and provides uncertainty estimates of all model parameters (sediment geoacoustic profiles) is applied. The geoacoustic information content of noise due to large commercial ships in a nearby shipping lane and of noise due to a research vessel traversing the array is quantified and compared. Parameter estimates from the inversions are compared with direct measurements from sediment cores and other geophysical data collected in the experiment area.

10:00–10:15 Break

10:15

2aUW6. Bayesian-Maximum Entropy method applied to seabed geoacoustic models using broadband and narrowband acoustic measurements. D. P. Knobles (KSA, LLC, P.O. Box 27200, Austin, TX 78755, davidknobles@gmail.com), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas a Austin, Austin, TX), William S. Hodgkiss (MPL, Scripps Inst. of Oceanogr., La Jolla, CA), Lin Wan, and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE)

Conditional probability distributions $P(\boldsymbol{H}|\boldsymbol{D},\boldsymbol{M})$ are computed using a Bayesian-Maximum Entropy method applied to acoustic measurements collected during the Seabed Characterization Experiment in the spring of 2017. Marginal distributions are computed for both seabed geoacoustic and source parameter values. A prior range-dependent seabed model $\boldsymbol{M}$ is derived from CHIRP survey measurements made in 2015. The prior bounds of parameter values forming the hypothesis vector **H** are obtained from an extensive set of sediment core measurements made in the region. The acoustic data, $\boldsymbol{D}$, were produced by small explosive and combustive sources and towed tonal sources. Two acoustic models are employed to sample the N-dimensional $\boldsymbol{H}$ space: Range-dependent parabolic equation RAM and a normal mode method that includes shear wave effects. The dimensionality of $\boldsymbol{M}$ is optimized via a Gaussian Mixture Model (GMM) and compared to the Bayesian Information Criterion (BIC) and the Akaike Information Criterion (AIC). [Work supported by Office of Naval Research.]

10:35

2aUW7. The measurement of muddy seabed properties using ambient noise coherence. Dieter A. Bevans (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), David R. Barclay (Oceanogr., Dalhousie Univ., Dept. of Oceanogr., Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca), and Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The autonomous passive acoustic lander Deep Sound was deployed five times during the Seabed Characterization Experiment and collected ambient noise data on four hydrophones, arranged in a inverted "T" shape, with three spaced in the horizontal and two in the vertical. The lander was deployed with the bottom-most phones approximately 30 cm above the seafloor, recording over an acoustic bandwidth of 5 Hz–30 kHz. Pressure time series, vertical and horizontal noise coherence (directionality), and the local temperature and conductivity were recorded continuously for periods of 9 hours. Wind-driven surface ambient noise coherence was used to estimate bulk acoustic seabed properties. An analytical Pekeris-waveguide noise model was fitted to the data in order to determine the bulk sound speed, density, and frequency dependent attenuation in the bottom fluid half-space. [Research supported by ONR.]

10:55

2aUW8. Hybrid geoacoustic inversion method and its application to different sediments. Zhenglin Li and Renhe Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 Beisihuan West Rd., Beijing 100190, China, lzhl@mail.ioa.ac.cn)

Bottom acoustic parameters have large effects on sound propagation both in shallow water and in deep water with an incomplete sound channel. A hybrid geoacoustic inversion method is proposed to invert for sound speed, density and attenuation coefficient based on the facts that the bottom acoustic parameters have different sensitivities to sound field at different ranges. The hybrid inversion method is applied to inversion for bottom parameter with different sediments in the Yellow Sea, East China Sea, and South China Sea and verified by using the core sampling measurements. The relationships between the sediments types and the bottom acoustic parameters are given. The differences of the sediment acoustic parameters between sandy silty(or silty sand) and clay(or mud) sediment at low frequency are discussed in the end. [Work supported by the National Natural Science Foundation of China under Grant No. 10434012 and Grant No. 41561144006.]

Contributed Papers

11:15

2aUW9. Path losses in sandy sediments with upward refracting sound speed profiles. Alexander G. Soloway (Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, alexander.soloway1@gmail.com) and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, Seattle, WA)

In this work, receiver-to-receiver path losses calculated from measurements of underwater explosions in shallow water (water depth $\sim$15 m) off the coast of Virginia Beach, VA will be presented. Within the approximate frequency range of 40 to 70 Hz, these data have path losses that suggest energy increasing with range by up to 10 dB. Above this frequency range, these data show energy decreasing with range as expected. An analytical model [Frisk, George V., "Determination of sediment sound speed profiles using caustic range information," *Bottom-Interacting Ocean Acoustics*. Springer, US, 1980, 153–157] and numerical propagation models will be used to show that these path losses can be attributed to an upward refracting sound speed gradient in the sediment. This upward refraction of sound waves propagating in the seabed results in the formation of caustics that define shadow zone boundaries. Below the cutoff frequency of the waveguide, receivers in the shadow zone will exhibit lower sound levels than receivers positioned farther from the source and outside of the shadow zone. While frequencies above cutoff will also undergo refraction in the seabed, the observability of this effect will be masked by trapped modes propagating in the water.

11:30

2aUW10. A full-field scattering coefficient for rough interfaces and a modified sonar equation for long-range reverberation. Anatoliy Ivakin (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A full-field perturbation approach [A.N. Ivakin (2016), *J. Acoust. Soc. Am.*, **140**(1), 657–665] allows modifying two terms of sonar equation for reverberation in complex environments and waveguides with rough interfaces. The first one describes a two-way transmission loss for the full-field intensity in the vicinity of the interface. The second term is a full-field scattering coefficient related to but different from the conventional scattering cross-section per unit area. In particular, the coefficient has a non-zero limit value at small grazing angles of incident and scattered waves defined by the contrast of acoustic parameters at the interface and the power spectrum of roughness. The modified sonar equation allows extremely fast estimating the long-range reverberation based on only one (central) frequency calculation of the transmission loss averaged over a certain span of ranges defined by the used frequency bandwidth, and only one value of the scattering coefficient (at zero grazing angles). No multiple calculations of the reverberation pressure field resulted from Monte-Carlo simulations of random rough interfaces are required. Numerical analysis of ocean reverberation is performed considering different types of rough interfaces: sea-surface, water-sediment interface, buried sediment interfaces, and bottom basement. Model-data comparisons are presented based on results of a recent shallow water reverberation experiment TREX2013. [Work supported by ONR.]

11:45

2aUW11. Azimuthal dependence of sound propagation due to seabed variability in shallow water. Mohsen Badiey, Lin Wan, Justin Eickmeier (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Newark, DE 19716, badiey@udel.edu), David P. Knobles (KSA LLC, Austin, TX), and Preston S. Wilson (Mech. Eng., Univ. of Texas, Austin, TX)

In shallow water regions, the environment has complicated temporal and spatial variability including changes of bathymetry, sediment layer structure, bottom property, and physical oceanographic spatial and temporal changes due to processes like internal waves. All these effects can influence sound propagation in the waveguide. The azimuth angle dependence of sound propagation has been studied using the broadband acoustic signals measured at the Atlantic Generation Station site on the New Jersey continental Shelf, where two distinctive geologic/geoacoustic regions exist [*J. Acoust. Soc. Am.* **96**(6), 1994]. The current paper revisits this idea by analyzing the modal dispersion of broadband acoustic signals deployed along circular tracks at the site of the Seabed Characterization Experiment 2017, where the seabed shows strong azimuthal dependent sub-mud layer ridges overlaid by a relatively uniform mud layer with variable thickness along different directions. The results of this paper can be utilized to assess the azimuthal dependence of the sound propagation in the mud patch region. [Work supported by ONR Ocean Acoustics.]

Session 2pAA

Architectural Acoustics and, Psychological and Physiological Acoustics: Perceived Diffuseness II

Jin Yong Jeon, Cochair
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Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th St., Troy, NY 12180

Chair's Introduction—1:15

Invited Papers

1:20

2pAA1. Novel scattering coefficient for characterizing diffusely reflecting surfaces. Jean-Dominique Polack (Institut d'Alembert, Sorbonne Universités/UPMC, 4 pl. Jussieu, Paris 75242 Cedex 05, France, jean-dominique.polack@upmc.fr) and Hugo Dujourdy (Institut Langevin, PSL Res. Univ., Paris, France)

In a recent publication (Acta Acustica **103**(2017) 480–491), the authors developed an innovative model for describing diffuse sound fields. Developing the diffusion equation one step further, this model makes use of the full stress-energy tensor to provide supplementary relations between sound intensity and sound energy. In the case of non-Sabine spaces (narrow or flat rooms), the supplementary relations naturally introduce new boundary conditions that can be interpreted as scattering coefficients on the walls, in analogy to the traditional absorption coefficients associated with the boundary conditions of the energy equation. The paper explains how the novel coefficient is obtained, compares its significance with the widespread diffuse field wall diffusion coefficient, and proposes an experimental technique for measuring it. Preliminary results will be presented, the expected range of values for the scattering coefficient explained as well as its link to wall impedance in the case of a locally reacting boundary.

1:40

2pAA2. Finding an appropriate headphone-based stereo track playback methods by referencing acoustic characters of professional recording studios. Myoung woo Nam (Dept. of Trans-disciplinary Studies, Seoul National Univ., D406, Iuidong 864-1, Yeongtonggu, Suwon, Gyeonggi 443-270, South Korea, mnam@snu.ac.kr)

Most Red Book CD or streaming 2-track music mixed and mastered in a professional recording studio. However, headphone listener experience different to its original studio condition. The recent headphone has decent frequency response, but the sounds do not match especially in soundstage (width and depth of music mix in a horizontal plane) perspective. Although Recording studios are designed to minimize unwanted reflections and resonances, every studio has its own room characteristics of natural room frequency response. In other words, "the acoustical room characteristic" added on "finalized mix" is the real representation of the mastered 2-track music. In this paper, we tried to find the ultimate headphone-based listening methods by referencing the professional recording studios.

2:00

2pAA3. Perceptual evaluation of a virtual acoustic room model. Gabriel Vigliensoni (Music Res. Dept., McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A1E3, Canada, gabriel@music.mcgill.ca), David Romblom (Dysonics, San Francisco, CA), Marc-Pierre Verge (Appl. Acoust. Systems, Montreal, QC, Canada), and Catherine Guastavino (School of Information Studies, McGill Univ., Montreal, QC, Canada)

Two experiments were conducted to perceptually evaluate a virtual acoustic room model implemented using a virtual microphone technique for the direct path and reflections, the image-source model for the reflection geometry and a Feedback Delay Network reverberator. Experiment 1 aimed to determine the best equalization profile for different room sizes. Anechoic recordings of 7 solo instruments were processed to generate 5 room sizes with profiles corresponding to 5 different absorption settings. Fourteen sound engineers and musicians were asked to rate the resulting stimuli in terms of how realistic each setting was for the instrument in that space. The highest ranked absorption setting for each room size was selected for Experiment 2 to investigate whether listeners could identify different rooms sizes across different instruments. Using a free sorting task, 15 musicians were presented with 7 different solo instruments in 5 different room sizes and asked to group them by type of space in which the instruments were performed. The analysis of the dissimilarity matrices indicates that participants were able to organize sounds by room sizes, but the two largest rooms tended to be clustered together. The results show that the virtual acoustic model is capable of rendering realistic room effects.

Contributed Papers

2:20

2pAA4. Room acoustics: Idealized field and real field considerations. Ernesto Accolti and Fernando di Sciascio (Instituto de Automática, National Univ. of San Juan - National Sci. and Tech. Res. Council, Av Libertador San Martín oeste, San Juan, San Juan 5400, Argentina, eaccolti@inaut.unsj.edu.ar)

How is an acoustically diffuse field defined? To what extent are valid the equations of diffuse field theory? These are the questions addressed in this presentation. The answers are explained through more general theories, in turn explained with figures instead of formulae. The starting point is the idealization of diffuse sound field, from where the basic calculation tools used in architectural acoustics are derived. Then, we go through the physical-mathematical models of wave theory and ray theory assuming diffuse field simplifications and analyze the scope of diffuse field models. Wave models and ray models are presented in a simple format with visual support and reference to the underlying mathematical models. The criteria used to define a diffuse field in frequency domain as well as in temporal domain are analyzed. Finally, we present a review of several state of the art tools used to address the real cases when diffuse field cannot be assumed.

2:35

2pAA5. Development of highly diffuse surfaces from architectural concept to listening experience. Gregory A. Miller, Carl Giegold, John T. Strong, and Laura Brill (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

Many different surface shapes can scatter sound effectively, but not every surface can scatter sound beautifully. When employed over large areas within performance spaces, reflections off sound diffusive surfaces must maintain much of the timbre of the original signal in order to preserve a sound quality pleasing to both performers and audience members. At the same time, these surfaces must be developed in ways that express the architect's visual aesthetic of the room. This paper present means of evaluating and optimizing the sound diffusive capabilities of surfaces within performance spaces, from architectural concept through iterative computer modeling processes and ultimately to physical mockups that can be evaluated both objectively and subjectively.

2:50–3:05 Break

3:05

2pAA6. Energetic wave equation for diffuse sound fields. Jean-Dominique Polack (Institut d'Alembert, Sorbonne Universités/UPMC, 4 pl. Jussieu, Paris 75242 Cedex 05, France, jean-dominique.polack@upmc.fr), Hugo Dujourdy (Institut Langevin, PSL Res. Univ., Paris, France), Baptiste Pialot (Institut d'Alembert, Sorbonne Universités/UPMC, Paris, France), and Thomas Toulemonde (Impedance, Paris, France)

The diffusion equation for modeling diffuse sound fields was proposed some fifty years ago on heuristic principles as an extension to Sabine's diffuse field model, and still receives much attention. In a recent publication (Acta Acustica **103** (2017) 480–491), the authors developed the model one step further, using the full stress-energy tensor to provide the missing relations between sound intensity and sound energy. This introduces some extra terms that, in case of non-Sabine spaces (narrow or flat rooms), can be defined with the help of the boundary conditions in terms of absorption and scattering coefficients on the walls. Integrating the divergence of the stress-energy tensor across the shortest dimensions of the space leads to a propagation equation of the telegrapher type, which can be solved using finite difference time domain simulation. Schemes for one-dimensional (corridors) and two-dimensional (open-space) spaces are proposed, and numerical results compared to measurements in real spaces. The comparison makes it possible to evaluate the absorption and scattering coefficients by an adjustment procedure. The paper discusses the range of values taken by these coefficients and compares them to more traditional building-acoustical coefficients. It also presents under which further assumptions the diffusion equation is recovered.

3:20

2pAA7. Diffusion: When phase and energy can become more important than directivity to the perception of "space". Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This paper will present the newest research on the properties of diffusion that are becoming more important than just directivity to the perception of "space." This includes phase response and energy levels remaining after the act of diffusion.

3:35

2pAA8. A spherical harmonics basis for quantifying the isotropy of sound fields in reverberant enclosures. Mélanie Nolan, Jonas Brunskog, and Cheol Ho Jeong (Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, melnola@elektro.dtu.dk)

This study examines an experimental method for evaluating isotropy in reverberant sound fields, based on an analysis in the spherical harmonics domain. The methodology relies on estimating the wavenumber (or angular) spectrum of the sound field in the room, to characterize the magnitude of the waves arriving from definite directions at the observation point. Subsequently, the obtained wavenumber spectrum is expanded into a series of spherical harmonics, and the multipole moments from the spherical expansion are used to characterize the isotropy of the sound field. This spherical harmonic basis is best suited for characterizing isotropy, as it provides an unequivocal characterization of the symmetry of the wave field. The work examines how theoretical considerations compare with experimental results obtained in various rooms with diverse diffuse field conditions. The experimental results are based on automated measurements using a scanning robot. In addition, the corresponding spatial distribution of the active sound intensity field is determined, making it possible to benchmark the proposed methodology with direct observations of the energy flows in the sound field.

3:50

2pAA9. Factors differentiating the 22.2- and 2-channel reproduced sound fields through an acoustic modeling of three listening rooms. Madhu Ashok (Univ. of Rochester, 500 Comput. Studies Bldg., P.O. Box 270166, Rochester, NY, mashok@ur.rochester.edu) and Sungyoung Kim (RIT, Rochester, NY)

We have simulated two loudspeaker configurations (22.2- and 2-channel reproduction) using the CATT-Acoustic software and analyzed the influence of room acoustics on the perception of multichannel-reproduced music. With the rapid growth of virtual reality (VR) and mixed reality (AR), there is a need for an immersive acoustic system that can maintain compatibility of auditory impressions in various acoustic conditions. The research question of the authors' project is whether an increased number of reproduction channels would reduce the room-induced perceptual difference. To answer this question, we have analyzed physical characteristics from calculated impulse responses (IRs) of three distinct room models (varying dimensions and reflecting surfaces). Among many characteristics, the early decay time (EDT) and clarity (C_{80}) values covary with the loudspeaker configurations. The IRs calculated from a 22.2-channel reproduction system had different EDT and C_{80} values for all three rooms. The change was more evident in a reverberant room (a normal listening room) than a relatively dry room (such as studio control room), possibly due to boosts of the ratio of direct sound to late reflections. No salient factor associated with room differences was observed. The subsequent subjective evaluations, performed by eleven listeners, support that they weighted the perceptual difference associated with the reproduction format more over all other perceptual dimensions.

Session 2pAB

Animal Bioacoustics, Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography: In Memory of George Ioup: Acoustics in the Gulf of Mexico II

Natalia Sidorovskaia, Cochair
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David K. Mellinger, Cochair
Coop. Inst.for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365

Contributed Papers

1:00

2pAB1. Decadal assessment of the sperm whale population trends in the northern Gulf of Mexico using acoustics. Kun Li, Natalia Sidorovskaia (Physics, Univ. of Louisiana at Lafayette, Broussard Hall, Rm. 103, 240 Hebrard Blvd., Lafayette, LA 70503, kxl1737@louisiana.edu), Christopher Tiemann (R2Sonic LLC, Austin, TX), and Azmy S. Ackleh (Mathematics, Univ. of Louisiana at Lafayette, Lafayette, LA)

Passive acoustic monitoring data have been used to assess population trends of sperm whales in the northern Gulf of Mexico. In this paper, the variability of sperm whale abundance in the Mississippi Canyon area derived from acoustic data collected between 2001 and 2017 is discussed in relation to seasons, habitat type, ambient noise levels, and environmental disturbances. The results have shown that sperm whales were present in the region throughout the entire monitoring period with lower activity in winter months. A considerable habitat shift was observed after the 2010 oil spill with sperm whale activity higher at the sites further away from the spill site. The results clearly indicate the importance of long-term spatially distributed acoustic monitoring in characterizing changes in Gulf of Mexico marine mammal population and their habitat.

1:15

2pAB2. Comparing performance of bottom-moored and unmanned surface vehicle towed passive acoustic monitoring platforms for sperm whale detection. Sakib Mahmud, Natalia Sidorovskaia, Kun Li (Physics, Univ. of Louisiana at Lafayette, 611, W Taft St., Lafayette, LA 70503, sxm3227@louisiana.edu), Chris Pierpoint (BioSci. Group, Seiche, Ltd., Devon, United Kingdom), Christopher Tiemann (LLC, R2Sonic, Austin, TX), and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Newport, OR)

Passive acoustic monitoring (PAM) is a more efficient method of monitoring the distribution and abundance of deep-diving cetaceans than conventional visual surveys. Many species produce identifiable acoustic signals during echolocation and communication which makes it possible to identify and classify based on their acoustical cues. Three different PAM platforms recorded data in overlapping time periods in the vicinity of the 2010 oil spill site: unmanned surface vehicle towed array, bottom-moored buoys, and Seaglider-mounted hydrophone. The detection rate of unmanned surface vehicle towed array and bottom-moored buoys were compared for their efficiency in detecting marine mammals. Detection events were obtained using independent detectors for each platform and then compared by feeding data through a common detector. The results of this study aid in the development of cost-efficient PAM methodology for mitigation and environmental impact assessment purposes. [This research was made possible by a grant from The Gulf of Mexico Research Initiative.]

1:30

2pAB3. Species-level classification and clustering of beaked whale echolocation recordings. Jack G. LeBien and Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, UNO, New Orleans, LA 70148, jlebien@uno.edu)

The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) consortium has collected passive acoustic monitoring data in the northern Gulf of Mexico since 2001. Recordings were made in 2007 near the Deepwater Horizon oil spill, which have provided a baseline for an extensive study of regional marine mammal populations in response to the disaster. Beaked whales are of particular interest as they remain one of the least understood groups of marine mammals, and relatively few abundance estimates exist. Efficient classification and clustering algorithms are demanded for mining the long-term passive acoustic data. Three algorithms using k-means, self-organizing maps, and spectral clustering are tested with various features of detected echolocation transients. Several methods are observed to effectively isolate recorded echolocation clicks of regional beaked whale species. The waveform fractal dimension is introduced as a feature for marine biosonar classification and shown to improve accuracy. A feedforward neural network classifier is also evaluated and shows performance with high accuracy under various noise conditions. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

1:45

2pAB4. Calculating sperm whale lengths in the Northern Gulf of Mexico. George Drouant and Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, george.drouant@selu.edu)

Sperm whales (*Physeter macrocephalus*) produce short duration acoustical clicks while diving to search for food, each click composed of several pulses. The time interval between consecutive pulses, the interpulse interval (IPI), can be used to estimate the length of the whale. IPIs have been estimated from whales of unknown orientation by processing several hundred clicks and averaging the results. In this work, discrete wavelet transforms and autocorrelation-based methods are used to obtain IPI estimates with improved consistency. The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015. Results presented here are produced using data recorded by the LADC Environmental Acoustic Recording Systems (EARS) at the site closest to the BP oil spill. Calculated lengths can be compared to lengths from

previous measurements as well as to future measurements in order to determine changes in average whale size, useful in determining population health. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

2:00

2pAB5. Dolphin target detection processes investigation using finite element model. Chong Wei, Matthias Hoffmann-Kuhnt (Acoust. Res. Lab., Tropical Marine Sci. Inst., National Univ. of Singapore, 12A Kent Ridge Rd., Singapore 119222, Singapore, weichong3310@foxmail.com), Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kailua, HI), Darlene R. Ketten (Dept. of Otology and Laryngology, Harvard Med. School, Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), and Abel Ho (Acoust. Res. Lab., Tropical Marine Sci. Inst., National Univ. of Singapore, Singapore, Singapore)

Dolphins and porpoises use their sophisticated biosonar systems for targets detection, within a range of a few meters to about 200 m, there is not a better sonar on the planet. In this study, the high resolution computer tomography (CT) scan data were used to create the detecting click signal propagation models of Atlantic bottlenose dolphins (*Tursiops truncatus*) and harbor porpoise (*Phocoena phocoena*). The finite element methods (FEM) were used to simulate the processes of the clicks emitted from phonic lips and transmit to the water through animals' heads. The biosonar beam forming in the nearfield and farfield including the amplitude contours were determined and compared to the prior measurement results. There were no evidences of convergence in the farfield, which were consistent with measurement results for *Tursiops truncatus*. Additionally, in a cross-modal matching experiments with *Tursiops*, we found that the accuracy of the successive match was significantly different when the following subjects with same shape were used (water-filled PVC pipes, air-filled PVC pipes, foam ball array, and PVC pipes wrapped by foam) the results of FE model shed some light on the reasons why the animal has significant difference in performances when detecting different targets in the experiments.

2:15

2pAB6. Influences of spatial variability in pelagic scattering layers on sperm whales behavior. Adrienne M. Copeland (Univ. of Hawaii at Manoa, 1315 East West Hwy., SSMC3, Rm. 10246, Silver Spring, MD 96734, acopelan@hawaii.edu), Ladd Irvine, Bruce Mate (Marine Mammal Inst., Dept. of Fisheries and Wildlife, Oregon State Univ., Newport, OR), and Whitlow Au (Univ. of Hawaii at Manoa, Kailua, HI)

The central slope of the Gulf of Mexico (GOM) is home to over 20 species of marine mammals. Prior to the Deep Horizon Oil Spill in 2010, sperm whales were the predominately sighted large whale in the area. This study investigated the distribution and utilization of the north-central GOM by sperm whales two years after the spill by comparing sperm whale presence to micronekton distribution. Micronekton, organisms 2–20 cm, might be a key component in understanding sperm whale behavior as they are potentially an essential link in the food web between primary producers and higher trophic levels, including cephalopods—primary prey items of sperm whales. A Simrad EK60 echosounder operating at 38 kHz recorded the backscatter of micronekton throughout the northern GOM. The spatial distribution of micronekton backscatter was highly variable and may be driven by proximity to the Mississippi Delta. Sperm whales were sighted by trained visual observers and foraging was detected using a towed hydrophone array. The sperm whales were sighted most often in shallow slope waters less than 900 meters in areas with higher micronekton backscatter, suggesting that the waters above sloping bottoms with dense patches of mesopelagic micronekton support the prey of sperm whales.

2:30

2pAB7. Bottom characterization in the northern Gulf of Mexico using chirp sonar and ship noise data. Altan Turgut (Acoust., Naval Res. Lab., Naval Res. Lab., 4555 Overlook Ave., S.W., Washington, DC 20375, alturgut@yahoo.com), Warren Wood (Marine GeoSci., Naval Res. Lab., Stennis Space Ctr., MS), and Raymond J. Soukup (Office of Naval Res., Arlington, VA)

As a part of Prof. George Ioup's multi-institutional research initiative, chirp sonar bottom surveys were conducted near the Mississippi Canyon in the northern Gulf of Mexico during the Littoral Acoustic Demonstration Center (LADC) 2001 experiment. Bottom geoacoustic properties were estimated using a shallow-towed 2–12 kHz chirp sonar that provided sub-meter resolution and 60 m penetration. The inversion results showed that the experimental site is mainly covered with clayey sediment. In 2010, a second bottom characterization experiment was conducted at $\sim$25 miles NE of the LADC 2001 experiment site using a deep-towed chirp sonar. Deep-towed chirp sonar provided geoacoustic property estimations as well as low-grazing-angle bottom-loss measurements. As a source of opportunity, own-ship noise data were also used for geoacoustic parameter estimation and bottom-loss measurements. Geoacoustic parameter estimations and bottom loss measurements from the chirp-sonar and ship-noise data provided similar results that were also in agreement with those from the LADC 2001 experiment. [Work supported by ONR.]

2:45

2pAB8. Population densities of dolphins in the Northern Gulf of Mexico. Kendal Leftwich, Jack G. Labien, SydiCharise Austin, Evan Wellmeyer, and Juliette W. Ioup (Physics, Univ. of New Orleans, 1021 Sci. Bldg.,New Orleans, LA 70148, kmleftwi@uno.edu)

The Littoral Acoustic Demonstration Center–Gulf Ecological Monitoring and Modeling (LADC-GEMM) consortium collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015, returning to sites previously surveyed by LADC (2007, 2009, and 2010). Results presented here are produced using data recorded by the LADC-GEMM Environmental Acoustic Recording Systems (EARS) at the site closest to the BP oil spill, specifically for Rizzo, bottlenose, Clymene, and Pantropical-Spotted dolphins. Clustering and trained classifier techniques were used to isolate dolphin species. A population model developed by LADC-GEMM based on acoustic data collected from the EARS buoys is used to estimate the strength of delphinid recovery in the northern Gulf of Mexico after the BP oil spill. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

Session 2pBA

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

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

Chair's Introduction—1:00

Invited Papers

1:05

2pBA1. Quantitative analysis of the angiogenic microvasculature in tumor using multiple scattering and dual-frequency transducers. Aditya A. Joshi, Sibo Li (MAE, NCSU, 911 Oval Dr., Raleigh, NC 27695), Sunny Kasoji (BME, UNC, Chapel Hill, NC), xiaoning jiang (MAE, NCSU, Raleigh, NC), Paul Dayton (BME, UNC, Chapel Hill, NC), and Marie M. Muller (MAE, NCSU, Raleigh, NC, mmuller2@ncsu.edu)

The objective is to characterize angiogenic networks using contrast-enhanced multiple scattering. The methods relies on the measurement of the diffusion constant from the time evolution of the incoherent backscattered intensity. Dual-frequency arrays allow to emit a pulse at a frequency close to resonance frequency of microbubbles, and to receive the backscattered echoes at higher harmonics, resulting in better contrast-to-tissue ratio compared to single frequency transducers. At 8 MHz, we demonstrated that the diffusion constant enables the quantification of the vascular density and anisotropy in tumor-related vasculature in rat models *in vivo*. We show here that the results could be further improved using a dual-frequency approach, using two linear arrays with different central frequencies (7 MHz/18 MHz) and using a custom made dual frequency transducer (3 MHz/15 MHz). The minimum detectable distance between two vessel-mimicking cellulose tubes filled with contrast agents was measured via the diffusion constant. The dual-frequency approach drastically reduced the smallest detectable distance between tubes, which was found to be 25μm using a single frequency, 20μm using two linear arrays and 10μm using the custom dual-frequency array, respectively. These results show that measuring the diffusion constant with a dual frequency array can enable the quantification of vessel density with high accuracy

1:25

2pBA2. Ultrasound molecular imaging of the secreted tumor marker Netrin-1 in multiple breast cancer models. Jennifer Wischhusen (LabTAU U1032, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, 151 Cours Albert Thomas, Lyon 69003, France, jennifer.wischhusen@inserm.fr), Jean-Guy Delcros (Cancer Res. Ctr. Lyon, U1052, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Katheryne E. Wilson (Radiology, MIPS, School of Medicine, Stanford Univ., Stanford, CA), Benjamin Gibert (Cancer Res. Ctr. Lyon, U1052, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Rodolfo Molina (LabTAU U1032, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Patrick Mehlen (Cancer Res. Ctr. Lyon, U1052, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Juergen K. Willmann (Radiology, MIPS, School of Medicine, Stanford Univ., Stanford, CA), and Frederic PADILLA (LabTAU U1032, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France)

With regard to biomarker expression, tumors are heterogeneous media displaying intra- and interpatient variability. Ultrasound molecular imaging (USMI) is a newly developed tool that can characterize heterogeneous tumors. Here, we focus on Netrin-1-overexpressing breast cancer. A newly developed targeted therapy interfering with Netrin-1 requires patient stratification. We developed Netrin-1-targeted microbubbles (MBs) and assessed USMI for identification of Netrin-1-positive breast cancer. *In vitro* flow chamber assays showed specific binding of anti-Netrin-1-MBs to human and murine breast cancer cell lines overexpressing Netrin-1. *In vivo*, in Netrin-1-positive transgenic or implanted mouse breast cancer models, USMI showed significantly increased signals with Netrin-1-targeted MBs, compared with isotype control MBs and targeted MBs after blocking of Netrin-1:—44.8%, 32.0%, 33.7% intensity for $MBs_{Netrin-1}$, $MBs_{Isotype}$, or after blocking, resp., in MMTV tumors;—28.2%, 13.7%, 13.4%, resp., in SKBR7 tumors;—60.9%, 47.8%, 47.9%, resp., in 4T1 tumors. In Netrin-1-negative tumors and in normal glands, targeted and control MBs showed no significant differences. Immunohistochemistry confirmed the expression of Netrin-1 in endothelial cells. In conclusion, USMI allowed the characterization of tumor heterogeneity and the differentiation between Netrin-1-positive and -negative tumors, and has the potential to become a companion diagnostic for breast cancer patient stratification.

Contributed Papers

1:45

2pBA3. Quantitative-ultrasound-based prostate-cancer imaging by means of a novel micro-ultrasound scanner. Daniel Rohrbach (Riverside Res. Inst., 156 William St., 9th Fl., New York City, NY 11215, drohrbach@RiversideResearch.org), Brian Wodlinger, Jerrold Wen (Exact Imaging, Markham, ON, Canada), Jonathan Mamou, and Ernest Feleppa (Riverside Res. Inst., New York, NY)

Currently, transrectal ultrasound (TRUS) guided biopsy is the only method for definitive diagnosis of prostate cancer. Our current study used a high-frequency (i.e., 29-MHz), transrectal, micro-ultrasound system (ExactVu^{TM} micro-ultrasound, Exact Imaging) to acquire RF data from 163 patients immediately before needle firing during 12-core biopsy examinations. Quantitative ultrasound (QUS) estimates of effective scatter diameter (ESD), effective acoustic concentration (EAC), midband (M), intercept (I), and slope (S) were calculated. Additional QUS estimates were derived including envelope statistics employing a Nakagami distribution and the envelope signal-to-noise ratio (ESNR). Estimate values were used to train linear-discriminant classifiers and performance was assessed using area-under-the-curve (AUC) values obtained from receiver operating characteristic (ROC) analyses based on 10-fold cross validation. A combination of ESS- and EAC-related parameters produced an AUC value of 0.75. When ESNR or PSA value was added as a feature, the AUC increased significantly to 0.77 or 0.78, respectively. The best classifier performance was obtained by combining envelope statistics, PSA, ESD, and EAC, which produced an AUC of 0.80. Our initial results with AUC values of 0.80 are very encouraging for developing a new tool for prostate-cancer biopsy guidance and treatment.

2:00

2pBA4. Anechoic lesion detection in a strongly scattering medium: Application to pulmonary nodules. Kaustav Mohanty and Marie M. Muller (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., 3147 B, 911 Oval Dr., College of Eng., EB-3, Raleigh, NC 27606, kmohant@ncsu.edu)

We present an imaging methodology to detect the presence of an anechoic lesion in a highly scattering medium consisting of spherical air scatterers. Classical imaging methods have failed to image such media. In the method presented here, the incoherent backscattered intensity is extracted and the linear growth of the diffusive halo is tracked. Sudden changes in this growth indicates the presence of a target. This algorithm combined with the algorithm of Winslow *et al.* on contour detection enables us to predict the presence and the location of such a lesion. This method has been developed to detect the presence of pulmonary nodules and ground glass opacities in the lung parenchyma. Using a 128-element linear array transducer operating at 5 MHz, experimental results were obtained from a sponge phantom with an air volume fraction of 50%, in which gelatin nodules of diameter 5 mm and 8 mm were implanted at depths of 15 mm and 20 mm, respectively. The diameter of the nodules could be predicted within an error margin of $\pm$25%. FDTD simulations were also carried out with nodule sizes of 6mm and 11 mm, in media with 20% and 50% air volume fractions. The coordinates of the center of the lesion could be predicted within a tolerance of $\pm$7%.

2:15

2pBA5. Development of tissue-mimicking phantom of the brain for ultrasonic studies. Somayeh Taghizadeh, Cecille Labuda, and Joel Mobley (Dept. of Phys. and Astronomy, Univ. of MS, 123 Lewiss Hall, University, MS 38677, staghiza@go.olemiss.edu)

Constructing tissue-mimicking phantoms of the brain for ultrasonic studies is complicated by the low backscatter coefficient of brain tissue, causing difficulties in simultaneously matching the backscatter and attenuation properties. In this work, we report on the development of a polyvinyl alcohol (PVA) based tissue-mimicking phantom with properties approaching those of white matter tissue. PVA was selected as the base material for the phantom as its properties can be varied by temperature cycling, variations in concentration and the addition of scattering inclusions, allowing some independent control of backscatter and attenuation. The ultrasonic properties (including speed of sound, attenuation, and backscatter) were optimized using these three methods with talcum powder as a scatterer. It was determined that the ultrasonic properties of the phantom produced in this study are best matched to brain tissue in the frequency range 1.0–2.5 MHz, indicating its utility for benchtop ultrasonic studies in this frequency range.

2:30–2:45 Break

Invited Papers

2:45

2pBA6. Quantitative ultrasound for the dynamic biomechanical analysis of tongue-food interface during oral processing: An *in vitro* study. Mathieu Mantelet, Isabelle Souchon, Maud Panouillé, François Boué (UMR 782 GMPA, INRA - AgroParisTech - Université Paris Saclay, Unit GMPA, INRA-AgroParistech, 1 Ave. Lucien BrÕtigniÒres, Thiverval-Grignon 78850, France), Frédéric Restagno (UMR 8502 LPS, CNRS - Université Paris Sud - Université Paris Saclay, Orsay, France), and Vincent Mathieu (UMR 782 GMPA, INRA - AgroParisTech - Université Paris Saclay, Thiverval-Grignon, France, vincent.mathieu@inra.fr)

The development of non-invasive methods is critical for a better understanding of the biomechanical phenomena involved in the dynamic mechanisms of food texture perception during oral processing. The aim of the present study is to investigate *in vitro* the potential of a quantitative ultrasound device in order to monitor the mechanical properties of tongue-food interface during a compression. A tongue-palate bio-mimicking set-up was designed, consisting in a traction-compression machine equipped with tongue and palate phantoms. Gels and emulsion filled gels were considered as model foods for their wide ranges of texture properties. Finally, a 1 MHz ultrasonic transducer positioned under the tongue records in real-time the pulse-echo response of the tongue-food-palate system during a compression. The results put in evidence *in vitro* the potential of tongue-food interface reflexion coefficient to monitor the competitive and collaborative contributions (i) of the tongue-palate system (roughness, lubrication, and velocity) and (ii) of food properties (composition, structure, and rheology) on the mechanical properties of the tongue interface during occlusion cycles. The study paves the way for the use of quantitative ultrasound methods to monitor tongue-food-palate system during oral processing.

3:05

2pBA7. Ultrasound propagation in bone. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

Bone sonometry is a rapidly evolving diagnostic technology for managing osteoporosis. Bone sonometers that measure ultrasound propagation through calcaneus or along long bones (*parallel* to the long axis) have been used for many years. However, the Food and Drug Administration recently cleared two new, innovative designs that measure ultrasound propagation *perpendicular* to long axes of tibia (2016) and radius (2017). A basic understanding of wave propagation in bone is required in order to compare devices based on different physical principles and skeletal sites. Phantom studies elucidate the relative importance of components of signal loss during propagation through bone: absorption, longitudinal-to-shear scattering, and longitudinal-to-longitudinal scattering (Wear, *IEEE Trans. UFFC,* 55, 2418–2425, 2008). Comparison of experimentally-measured ultrasound properties (attenuation, sound speed, and backscatter coefficient) and finite-element-analysis-derived mechanical properties *in vitro* (n = 25) indicate that ultrasound measurements provide additional information regarding fracture risk beyond that provided by bone quantity alone (Wear *et al., Bone,* **103**, 93–101, 2017). Bone can support two longitudinal waves, which often overlap in time and frequency domains but can be separated using Bayesian probability theory or the modified least-squares Prony's (MLSP) method (Wear *et al*., *J. Acoust. Soc. Am.*, **136**, 2015–2024, 2014).

3:25

2pBA8. A new dual-mode ultrasonic technique for assessing cortical bone. Jonathan J. Kaufman and Gangming Luo (CyberLogic, Inc., 611 Broadway, Ste. 707, New York, NY 10012, jjkaufman@cyberlogic.org)

Osteoporotic fractures are a major public health problem; diagnosis is based on x-ray densitometry (DXA); however, DXA is not able to accurately predict who will and will not suffer a fracture. An alternative to DXA is ultrasound, which is viewed as having the potential to better characterize fracture risk. Our approach is based on "dual-mode" ultrasound in which 2 distinct modes are used together to assess cortical bone. In particular, measurements are made at the mid-shaft tibia in both axial-transmission and pulse-echo modes. Axial transit time (τ_A) and pulse-echo transit time ($\tau_{P/E}$) are obtained at the same tibial cortical site. These 2 transit times can be used in a classification scheme to identify individuals at greatest risk of fragility fracture. A pilot set of measurements on 5 individuals has been carried out to demonstrate the basic feasibility of the proposed technology. For the 5 subjects, each subject data point was found to be located in a distinct region (i.e., a hypothesized distinct bone quality state) of the quadrant associated with the 2 transit times. Because the 2 transit times are each affected in distinct fashions by a variety of bone quality factors (such as degree of mineralization, porosity, cortical thickness, and biomechanical stiffness), the bivariate feature has the potential to accurately identify those individuals at increased risk of fracture. Future studies on individuals with and without fragility fractures will elucidate the capabilities and potential of the dual-mode ultrasound technology.

Contributed Paper

3:45

2pBA9. Diffusion constant and frequency dependent attenuation for the assessment of cortical bone porosity. Omid Yousefian, Yasamin Karbalaeisadegh (Mech. and Aerosp. Eng., North Carolina State Univ., 2704 Brigadoon Dr., Apt. A, Raleigh, NC 27606, ykarbal@ncsu.edu), Kay Raum (Charité - Universitätsmedizin Berlin, Berlin, Germany), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

The objective of this study is to assess cortical bone porosity through measurements of the diffusion constant and attenuation as a function of frequency. Using FDTD simulations, the diffusion constant was calculated for 45 modified human femur shaft bone geometries obtained from a reference binarized 100-MHz Scanning Acoustic Microscopy image. The diffusion constant was sensitive to cortical porosity and demonstrated around 85% decrease for 10% increase in porosity. This suggests that lower diffusion constant values could be associated with more severe cases of osteoporosis. Using a spectroscopy approach in the 1–8 MHz range, we demonstrate that the attenuation coefficient as a function of frequency can be approximated as:$\alpha(f) = Af^B + C$, where "A", "B" and "C" are related to pore diameter and pore concentration. For a constant pore density (10 mm^2), "A" increased from 0.03 to 19.79 when pore diameter increased (Po.Dm $\in$40–130μm), while "B" and "C" were found to decrease from 2.654 to 0.5741 and from −0.061 to −21.43, respectively. A similar trend was observed when pore diameter was kept constant. A model was derived to quantify pore size and concentration through constrained nonlinear optimization method. This parametric model combined with estimation of the diffusion constant has the potential to enable the quantification of pore size and concentration.

Invited Papers

4:00

2pBA10. *In vivo* radius bone evaluation in their teens by two longitudinal wave propagation. Mami Matsukawa (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), Isao Mano (Doshisha Univ., Joyo, Japan), Yutaro Yoneda, Kaoru Horii (OYO Electric Co., Joyo, Japan), Shiori Umemura, and Etsuko Ozaki (Kyoto Prefectural Univ. of Medicine, Kyoto, Japan)

An ultrasonic bone measurement system, LD-100 (OYO electric),[1] has been developed for the evaluation of the distal 5.5% site of the radius of the non-dominant hand. The system measures two longitudinal waves (fast and slow waves) which propagate in the inside cancellous bone and echo waves reflected from the interface between the cortical and cancellous bones. The properties of these waves can give us the information of cancellous bones. We can also estimate the cortical thickness from the echo waves. For the small radius of teenagers, we have improved the system using an annular type transducer to avoid the effects of guided waves in the cortical bone. The radius bones of 654 teenagers were measured. The cortical thicknesses of female students in their late teens were around 95 % of the young adult mean (YAM), where those of male students showed variation from 90 to 100%. The cancellous bone densities in their late teens were 82–94% (female) and 66–85% (male). The growth of cancellous bone was late, which was clearer in men. In addition, the total bone growth of men seemed slower than that of women. [1]H. Sai, *et al*., Osteoporos Int. (2010) **21**:1781.

4:20

2pBA11. Associations between ultrasonic backscatter, bone density, and microstructure in cancellous bone characterization. Dean Ta (Dept. of Electron. Eng., Fudan Univ., 220 Handan Rd., Shanghai 200433, China, tda@fudan.edu.cn) and Chengcheng Liu (Inst. of Acoust., Tongji Univ., Shanghai, China)

Ultrasonic backscatter is related to the microstructure of cancellous bone. However, whether backscatter measurement can provide additional information independently of BMD remains speculative. In this study, we analyzed the independent contribution of cancellous bone densities and microstructure to apparent backscatter signals *in vitro*. The results showed that ultrasonic backscatter and the statistics of backscatter envelope were significantly correlated with bone densities and microstructure. After adjustment for BMD, some trabecular structures still contributed significantly to the adjusted correlations, with moderate additional variance explained. Multiple linear regressions revealed that both bone density and structure contributed significantly to the prediction of ultrasonic backscatter (adjusted $R^2 = 0.75–0.93$, $p<0.05$), explaining an additional 14.0% of the variance at most, compared with that of BMD measurements alone. We also demonstrated that the Nakagami model had great potential in trabecular microstructure characterization. The results proved that ultrasonic backscatter was primarily determined by bone apparent density, but bone densities plus microstructure structure could achieve encouragingly better predictive performance than BMD alone. This study implied that ultrasonic apparent backscatter might provide additional density and structural features unrelated to current BMD measurement. Thus, we suggest that ultrasonic backscatter measurement could play a more important role in cancellous bone evaluation.

Contributed Paper

4:40

2pBA12. Assessment of dental implant stability: Coupling modeling and experimental approaches. Romain Vayron, Vu-Hieu Nguyen, and Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr)

Dental implants are widely used clinically but implant failures, which may have dramatic consequences, still occur and remain difficult to anticipate. Accurate measurements of implants biomechanical stability are of interest since they could be used to improve the surgical strategy. The aim of this work is to develop a medical device capable of estimating dental implant stability using quantitative ultrasound. To do so, our approach consists in coupling modeling and experimental methods. Firstly, an implant was initially completely inserted in the proximal part of a bovine humeral bone sample. The 10 MHz ultrasonic response of the implant is then measured and a quantitative indicator is derived. Then, the implant is unscrewed by 2π radians and the measurement was realized again. The value of indicator significantly increases as a function of the number of rotation. The results show that bone quantity in contact with the implant has a significant influence on its ultrasonic response. Secondly, a 3D finite element model is employed and the geometrical configuration is assumed to be axisymmetric. The numerical results show that the implant ultrasonic response changes significantly when a liquid layer is located at the implant interface.

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics V

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

1:00

2pEA1. Timbre analysis from monotone rearranging of the acoustic signal: Application to fountains. Laura Velardi (LISA-Environ. HydroAcoust. Lab, ULB, av. F.D. Roosevelt 50, Brussels 1050, Belgium, lvelardi@ulb.ac.be), Jean-Pierre Hermand (LISA-Environ. HydroAcoust. Lab, ULB, Brussels, Brussels Capital, Belgium), and D'Autilia Roberto (Matematica, Università degli Studi di Roma Tre, Rome, Italy)

Public fountains are an example of a soundmark that is both non musical and pleasing to the ear due to their specific timbre. Standard audio descriptors designed for musical sounds are limited in describing such sounds. A new method is proposed for identifying and classifying fountains sounds as well as surrounding urban noises. The method consists in sorting discrete-time signal samples in a monotonically descending order. Features of the resulting curve are extracted in an attempt to define a timbral evaluation metric. Reference signals including natural water sounds and synthesized pink noise are also analyzed and the results are compared to those of fountain sounds. Outcomes are discussed in the framework of urban soundscape studies, in order to improve acoustical urban planning.

1:15

2pEA2. Assessment of granular distortion in digital sound recorders by statistical analyses of recorded sound and analytically distinguishing quality of 24-bit and 16-bit analog to digital converters. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Modern sound recording systems are mostly digital type, using at least one or several analog to digital converters that commonly provide 24-bit or 16-bit resolution and accordingly limits the recording quality. The use of such digitizers inherently add some granulation distortion. A 16 bit digitizer is theoretically limited to granulation step size of 1/65,536. Whereas a 24 bit digitizer is theoretically limited to granulation step size of 1/16,777,216. Smaller granulation step size provides better fidelity of the recording. This study explored the operational resolution of a few of such digital recorders by statistical analysis of samples of recorded sound tracks. The histogram of the digital samples of the sound recordings showed that a digitizer may not always be digitizing to the finest precision.

1:30

2pEA3. A micromachined microphone based on the membrane with bias voltage and field-effect-transistor mechano-electrical transduction. Junsoo Kim, Hoontaek Lee, Donghwan Seo, Jaehyeok Jin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), San 31 Hyoja-dong, Nam-gu, Pohang, Gyeongsangbuk-do 37666, South Korea, chak7258@postech.ac.kr), Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyungbuk, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

Recently, capacitive-type microphone dominates the MEMS (Micro-Electro-Mechanical-System) microphone market. Since sizes of the electrodes on the membrane and backplate determine the sensitivity of the microphone, there is a limit in reducing the size of the MEMS microphone. Here, we propose a micro-machined microphone consist of the membrane with bias voltage and field-effect-transistor (FET). The difference between the conventional capacitive MEMS microphone and proposed microphone is the Mechano-Electrical transduction mechanism. By biasing the voltage on membrane, the gate voltage is induced depending on the vibration of the membrane, eventually changes the source drain current. In case of a MEMS microphone using the proposed FET and biased membrane microphone, low frequency roll-off according to the energy conversion does not occur, and the size of the membrane and backplate can be reduced as compared with the conventional MEMS microphone, without the degradation of sensitivity. In addition, depending on the biasing condition, amplitude modulation technique to reduce the flicker noise of the FET can be applied, which leads to the higher SNR (Signal to Noise Ratio). Conventional metal-oxide-semiconductor fabrication process and micromachining process were used. In this study, design, fabrication and performance test of the proposed FET microphone are conducted. [Work supported by CMTC, UM15304RD3.]

1:45

2pEA4. A micromachined microphone based on the electret membrane and field-effect-transistor mechano-electrical transduction. Junsoo Kim, Hoontaek Lee, Donghwan Seo, Jaehyeok Jin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyeongsangbuk-do, South Korea), Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyungbuk, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 405, POSTECH, San31, Hyoja-dong, Nam-gu, Pohang city, Kyungbuk, South Korea, Pohang, Kyungbuk 37666, South Korea, wkmoon@postech.ac.kr)

Most commercially available MEMS (Micro-Electro-Mechanical-System) microphones use a capacitive method, and their structure and performance are saturated to some extent. However, due to the limitation of the capacitive transduction method, the roll-off at the low frequency is inevitable, and there is a limit in reducing the mechanical thermal noise caused by the squeeze film damping that occurs between membrane and backplate structure. Proposed electret membrane and FET (Field Effect Transistor) based MEMS microphone detects a change in the source-drain current according to the gate voltage change of the FET induced by vibrating the membrane with the fixed charges. In the case of a MEMS microphone using the proposed FET, low frequency roll-off according to the energy conversion does not occur, and the size of the backplate can be drastically reduced as compared with the conventional MEMS microphone, thereby further reducing the mechanical thermal noise, leading to the possibility of achieving the higher SNR (Signal to Noise Ratio) than 65 dBA. Conventional metal-oxide-semiconductor fabrication process and micromachining process were used. In this study, design, fabrication and performance test of the proposed FET based MEMS microphone are conducted. [Work supported by CMTC, UM15304RD3.]

Session 2pEDa

Education in Acoustics: Acoustics Education Prize Lecture

Chair's Introduction—2:00

Invited Paper

2:05

2pEDa1. Student-centered acoustical engineering education at the University of Hartford. Robert Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu)

The University of Hartford has provided the author with a fulfilling métier, shepherding two undergraduate engineering programs in the area of acoustics: (1) the Bachelor of Science in Mechanical Engineering (BSME) with Acoustics Concentration and (2) the Bachelor of Science in Engineering with a major in Acoustical Engineering & Music. The first is part of our BSME degree program that has required courses in Vibration as well as Engineering Acoustics since the 1960's. The Acoustical Engineering & Music degree is a unique program instituted in 1976, where applicants must meet engineering's math and science entrance requirements as well as pass the audition requirements of our music conservatory (The Hartt School). Both ABET-accredited programs encompass the same engineering vibrations and acoustics courses, as well as the same acoustics projects sequence, beginning in the sophomore year. Alumni of both undergraduate programs have successfully obtained positions in consulting (architectural and environmental), audio product and A/V design, musical instrument design, hearing- and psychoacoustic-related design, noise/vibration control of machines, as well as graduate degrees. The use of real world industry-sponsored acoustic projects for engineering design courses throughout the curriculum will be described, as well as the programs' Service Learning components and High Impact Practices.

TUESDAY AFTERNOON, 5 DECEMBER 2017 BALCONY I/J/K, 3:15 P.M. TO 4:15 P.M.

Session 2pEDb

Education in Acoustics: General Topics in Acoustics Education

Contributed Papers

3:15

2pEDb1. Writing an acoustics textbook under the ASA Press/Springer contract. Steven L. Garrett (None, Appl. Res. Lab, P. O. Box 30, State College, PA 16804, sxg185@psu.edu)

The Acoustical Society's Books + Committee decided to add first editions to their "classic reprints." After a search process, the ASA decided that it was in the best interests of that project to team with Springer. A contract between ASA and Springer was signed in May 2013 creating a mechanism by which ASA members could propose and write new books. Members, student members, or registered meeting attendees could purchase paperback versions of those books produced under that contract for $24.99. Having just published my acoustics textbook [*Understanding Acoustics*; ISBN 978-3-319-49976-5], I will report on my experiences during the entire process, which begins with a book proposal. If the Committee believes that the proposed book would serve the interests of the membership, an individual contract is signed. Springer provides a "template" in MSWord[â] or L[a]T$_E$X format. I used the Word version that operated like any *.docx. Once the manuscript was complete, two ASA members were assigned as content editors for the technical review. Upon their approval, the layout, some redrawing of artwork, and a "grammar and spelling" review were done very quickly by Springer's affiliate in India. I corrected my "author proofs" at a Springer web site that was amazingly user friendly.

3:30

2pEDb2. Development of a simple replica of the middle ear pathology for demonstration of tympanometric measurements. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

This study evaluates a simple replica of the middle ear pathology for demonstration of tympanometric measurements. Common tympanometers often use small passive cavities for calibration of basic measurements such as equivalent ear canal volume. But the real ear normally presents an inverted V shaped tympanogram. The pathologic ears present various

characteristic shapes of tympanograms such as unusually high or low peaks. In some patients the tympanometric peak may be shifted away from the zero differential pressure point. Often such patients may not be readily available during a classroom demonstration of the procedures. Utility of this study to replicate such pathologic conditions with a simple model will be discussed.

3:45

2pEDb3. Engaging high school students in studying marine mammals observed near the BP oil spill. Kendal Leftwich, Juliette W. Ioup, C. Gregory Seab, Simeon P. Benit, and Matthew Firneno (Physics, Univ. of New Orleans, 1021 Sci. Bldg., New Orleans, LA 70148, kmleftwi@uno.edu)

The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project partnered with Warren Easton Charter High School to analyze underwater acoustic data collected in the northern Gulf of Mexico during the summer of 2015, returning to sites previously surveyed by LADC. Results presented here are produced using data recorded by Environmental Acoustic Recording Systems (EARS) at the site closest to the BP oil spill. The aim of the outreach project described here is for high school students to learn to work independently using scholarly works to analyze and interpret data, as well as to communicate and collaborate in a professional environment. Various marine mammals have been detected, including sperm whales, beaked whales (Cuvier, Blainville, Gervais, and BWG), and dolphins (bottlenose, Rizzo). The numbers and types of marine mammals are possible indicators of the health of the area and the marine mammals around the oil spill. Changes will indicate possible future problems. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

4:00

2pEDb4. Educational practice of linear array techniques at the Acoustics department of the Escuela Superior de Ingeniería Mecánica y Eléctrica campus Zacatenco, Instituto Politécnico Nacional, Mexico. Eduardo Vivas (CIBNOR, La Paz, Baja California Sur, Mexico), Omar A. Bustamante (Acoust., ESIME, IPN, Cuautitlan Izcalli, Mexico), and Sergio Beristain (Acoust., ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

Sound Source Location is a topic that has been studied over the past decades for a wide range of applications whether in air or underwater, resulting in the development of different sophisticated methods and systems. Nevertheless, this topic in not usually included in the curricula of acoustic courses for the requirement of expensive and specialized instrumentation. The purpose of this research is to develop a practical approach looking to involve the students by using common hardware for which they are already familiar with: hardware from the videogames industry. In the Acoustics department of the Escuela Superior de Ingeniería Mecánica y Eléctrica (ESIME Zacatenco), IPN Mexico, a low-cost Sound Source Location system has been implemented for this purpose. The system is built around a PS3 Eye digital camera that includes a linear array of four microphones, and Raspberry Pi microcomputer. The Beamforming algorithms are programmed in Python to provide flexibility and ease of experimentation. It was originally designed for research applications, and now it is a prototype for laboratory practice. The system provides the undergraduate students with the basics of linear arrays and associated processing techniques engaging the students by using familiar hardware from the games industry and real life applications.

TUESDAY AFTERNOON, 5 DECEMBER 2017 STUDIO 7, 2:00 P.M. TO 3:30 P.M.

Session 2pID

Interdisciplinary: Guidance from the Experts: Applying for Grants and Fellowships

Martin S. Lawless, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Michaela Warnecke, Cochair
Dept. of Psychological and Brain Sciences, Johns Hopkins University, 3400 N Charles St., Baltimore, MD 21218

Matthew C. Zeh, Cochair
Mechanical Engineering Graduate Acoustics Program, University of Texas at Austin, 1300 West 24th Street, Apt. 212, Austin, TX 78705

A panel of successful fellowship winners, selection committee members, and fellowship agency members will answer questions regarding grants and fellowships, application advice, and funding opportunities. The panelists will briefly introduce themselves, followed by a question and answer session with the audience.

Session 2pMUa

Musical Acoustics: Cajun Music: Accordians, Culture, and History

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

Chair's Introduction—1:55

Invited Papers

2:00

2pMUa1. Early history of the accordion family: Where did all the accordions come from? James P. Cottingham (Physics, Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

Two hundred years ago, there were no accordions. Free reed instruments were known in Asia for thousands of years, but the free reed instruments of European origin such as the accordion, harmonica, and reed organ were only invented and developed during the last two centuries. In 1780, Kratzenstein published a paper in St. Petersburg describing a speaking machine that produced vowel sounds using free reeds with resonators of various shapes. This event marks a convenient, if arbitrary, starting point for the history of the free reed musical instruments of European origin. These instruments developed rapidly, and by 1850, the accordion, concertina, harmonica, reed organ, and harmonium all had been invented and developed into more or less final form. This paper presents some episodes in the development of these instruments, in particular the accordion-concertina family, along with discussion of their acoustical design characteristics. Also addressed is the question of the influence of the Asian free reed mouth organs on the origin of the Western free reeds.

2:20

2pMUa2. Fluid dynamics approach to free reed physics. Thomas Tonon (Bluesbox, 11 Bolfmar Ave., Princeton Junction, NJ 08550, tsbtonon@gmail.com)

The periodic vibration of a free reed subject to a steady pressure gradient (bellows pressure) is examined from a fluid dynamics approach. In this analytic study, the self-excited forcing function is modeled from the static and dynamic pressures acting on the vibrating tongue surface. A first formulation includes standard mounting of the reed over a cavity, with a hole that accommodates a mean air flow to the outside. A second formulation includes in addition a resonant air geometry that could include non-standard operation, such as pitch bending.

2:40

2pMUa3. Cultural significance of the diatonic single-row button accordion in South Louisiana. Mark F. DeWitt (School of Music and Performing Arts, Univ. of Louisiana at Lafayette, UL Lafayette School of Music, PO Box 43572, Lafayette, LA 70504-3572, dewitt@louisiana.edu)

Musical acoustics are intimately bound up in culture. Musical instruments are vehicles for artistic expression in terms of visual design, timbre, and musical style for musicians who play them and sometimes for the artisans who make them. In complex multicultural societies, certain instruments can also become icons for group identity, similarly marked in visual, timbral, and musical terms. The case of the melodeon, known in Louisiana as the single-row button accordion or the Cajun accordion, richly exemplifies these possibilities. The instrument arrived in Louisiana in the mid-to-late 1800s and became the instrument of choice to play at house dances and dance halls by the 1920s, adopted by two neighboring ethnic groups, French-speaking Creoles of color and Cajuns. Local artisans began making single-row accordions when the supply from Germany ceased during World War II, creating a new, higher-quality version of the instrument with a distinctive appearance that continues to be the instrument of choice for Cajun accordionists and some zydeco players. This paper will elaborate on the history of the melodeon in Louisiana, the musical styles that use it, and its status as a regional and ethnic identity symbol.

3:00

2pMUa4. The craft of building Cajun accordions. Larry G. Miller (Bon Cajun Accordions, 886 McMillan Ave., Iota, LA 70543, boncajun@charter.net)

Craftsmen have been building single-row diatonic button accordions in South Louisiana since the mid-twentieth century. A brief history of this phenomenon will be given, followed by an explanation of how these accordions are constructed, from the perspective of someone who has made these instruments for decades, trained other makers, and served as the parts supplier for several shops. Topics covered will include the types of wood used, other parts and materials, design choices that affect sound quality versus visual appeal, special challenges in assembly, and tunings.

3:20

2pMUa5. Cajun and Zydeco accordion playing styles. Chad Huval (Univ. of Louisiana at Lafayette, 239 Washington St., Breaux Bridge, LA 70517, cchuval@yahoo.com)

The 10-button single-row diatonic accordion is a mainstay in the music of Louisiana's Creoles of Color and Cajuns. From the single-row diatonic accordion's appearance in the earliest recordings of Louisiana music to now, a unique syncopated playing style has developed, leading to today's zydeco and Cajun styles. This talk will focus on how the 10-button single-row diatonic accordion is played, with demonstrations of the nuances that make it Cajun, or make it zydeco.

3:40–4:05 Panel Discussion

TUESDAY AFTERNOON, 5 DECEMBER 2017 SALON E, 5:00 P.M. TO 6:00 P.M.

Session 2pMUb

Musical Acoustics: Cajun Music Concert by the Savoy Family Band

James P. Cottingham, Chair

Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

The Savoy Family Cajun Band plays honed down, hard-core Cajun music laced with sounds of fiddle and accordion rooted in South Louisianan tradition. Marc and Ann Savoy and their sons, Joel and Wilson, work together to create a tight, intense sound. Marc Savoy is widely recognized as a player and builder of the Cajun accordion and has received the National Heritage Fellowship Award. Marc and Ann have been performing all over the world and recording together since 1977. They appeared on the PBS series "American Roots," for which Ann wrote the chapter on Cajun music in the book that accompanied the series. Joel and Wilson are each distinguished musicians, both a soloists and members of other groups. The Savoy Family Band brings the energy of the dancehalls of southwest Louisiana to the stage, peppered with informative anecdotes about life in the Cajun heartland.

TUESDAY AFTERNOON, 5 DECEMBER 2017 STUDIO 2, 1:00 P.M. TO 2:55 P.M.

Session 2pNSa

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

Jay Bliefnick, Cochair

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

Jonathan R. Weber, Cochair

Durham School of Architectural Engineering & Construction, University of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

Chair's Introduction—1:00

Invited Papers

1:05

2pNSa1. Evaluating hospital soundscapes to improve patient experience. Jay Bliefnick, Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, jbliefnick@huskers.unl.edu), and Rebecca Jackson (Nebraska Medicine, Omaha, NE)

Patients routinely perceive hospital soundscapes to be poor when rating their experience on HCAHPS (Hospital Consumer Assessment of Healthcare Providers and Systems) surveys administered after discharge. In this study, sound levels within five hospital units were correlated with HCAHPS noise perception survey data. Acoustic metrics including A-weighted equivalent, minimum, and maximum (LAEQ, LAMIN, & LAMAX) and C-weighted peak (LCPEAK) sound pressure levels, occurrence rate, and speech intelligibility index were evaluated in 15 patient rooms and 5 nursing stations. Average patient room LAEQ values within the five units ranged between 52 dBA and 61 dBA with speech intelligibility ranging from poor (<0.45) to marginal (0.45 to 0.75). The absolute minimum values measured within the patient rooms (LAMIN, LCMIN, & LZMIN) were found to be correlated with HCAHPS data and other metrics revealed trends consistent with patient perception. For example, the lowest rated unit also had higher occurrence rates, indicating this unit was louder more often. Ceiling type was also found to impact sound levels with LAEQ 5 dBA quieter on average for rooms utilizing acoustic tile ceilings. Taken as a whole, these results provide insight into acoustic metrics and design strategies which can ultimately be utilized to improve patient experience.

1:25

2pNSa2. Acoustical evaluation of quiet time and its impact on patient outcomes. Jonathan R. Weber, Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jonryanweber@gmail.com), Myra Rolfes, Heather Cooper (Neonatal Intensive Care Unit, Children's Healthcare of Atlanta, Atlanta, GA), Brooke Cherven (Nursing Res. & Evidence Based Practice, Children's Healthcare of Atlanta, Atlanta, GA), and Ashley Darcy Mahoney (The George Washington University School of Nursing, Autism and Neurodevelopmental Inst., Washington, DC)

As healthcare evolves from focusing on survival to prioritizing patient care, more efforts have been devoted to exploring the characteristics of an optimal environment. Intensive care units (ICUs) often involve more urgent situations that require more medical equipment, in-unit procedures, and staff that all contribute to the noisy soundscape. Recently, administrative noise reduction strategies such as Quiet Time (QT) and behavioral modification and noise awareness programs have gained popularity. The literature typically demonstrates some positive effects from these interventions; however, there are inconsistencies in magnitude and sustainability. We conducted an 18-month, longitudinal study of QT aimed at (a) characterizing the Neonatal Intensive Care Unit (NICU) soundscape more extensively and with newly developed metrics and (b) determining the impacts of sound on patient physiological responses. Results show some evidence of trends toward decreased noise levels after QT implementation. For example, QT generally resulted in lowered occurrence rates, indicating a decreased number of high-level noise events. Additionally, spectral data showed significant decreases in level across the vocal frequency range and improved speech intelligibility index scores with QT implementation. Ultimately, these results begin to provide a template for determining the optimal soundscape to aid more targeted administrative and architectural design strategies.

1:45

2pNSa3. An overview of the key misperceptions that are preventing any significant improvements when addressing operational noise in acute care hospitals. Gary Madaras (Making Hospitals Quiet, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

The Making Hospitals Quiet program has been working with healthcare providers inside their existing and noisy acute care hospitals since 2011. An overview will be given on why misperceptions by healthcare professionals and acousticians have resulted in little improvement in the collective patient perception of quietness while admitted in these facilities. Tactics for overcoming these misperceptions will be discussed.

2:05

2pNSa4. Acoustics and the built environment in pediatric hospital units. Ian R. Hough, Yoshimi Hasegawa, and Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, ian.r.hough@gmail.com)

Previous studies have indicated that noise in hospitals can present challenges for patients and staff such as sleep disruption and increased stress symptoms. However, few studies have looked more holistically at the built environment to include evaluations of aspects in addition to noise, such as lighting, thermal comfort, and air quality. In this study, built environment measures and staff perception surveys were conducted in two units: a pediatric intensive care unit (PICU) and a pediatric medical-surgical unit (MedSurg). Results were evaluated to understand physical and perceptual differences based on the time of day, type of unit, and location within the unit. Results revealed interesting trends about acoustics and the built environment. For example, perception of noise, lighting, thermal comfort, and odor differed depending on the location within the unit (nurse stations, patient rooms, and corridors). Many perceptual results tracked with trends in the built environment measures. For example, day shift workers were significantly more annoyed with noise. Likewise, acoustic occurrence rates were generally found to be higher during the day, indicating the units were louder more often in the daytime with shorter periods of restoration. Viewed holistically, these findings reveal potential opportunities to improve the overall built environment in hospitals.

Contributed Papers

2:25

2pNSa5. Noise in the recovery area of a large hospital. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

A noise study was made in a very large government hospital which includes eight surgery rooms and a very large recovery area where patients are placed from as little as half an hour, either before or after a surgery, to about ten hours after some given surgery. In this area, physicians, nurses and bed moving personnel activity is almost continuous, and it has plenty of "Life Equipment" monitoring each and every patient. The noise measurements were concentrated in the recovery area, which is a very large room with very hard boundaries, that includes over 25 beds and aisle areas with no more than curtains separating them in order to allow for easy movement of rolling beds in and out from the surgery area and to the different specialty floors of the hospital. Results of these measurements are presented and commented.

2:40

2pNSa6. A study on 7 color marking of impact noise. Uk-Jin Song and Myungjin Bae (Dept. Information and TeleCommun., Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul 156743, South Korea, imduj@ssu.ac.kr)

Interlayer noise is a noise pollution that occurs mainly in a space where a large number of households, such as multi-family houses or apartments, are together. There are various types of interlayer noise depending on what kind of sound is generated, but there are impact sounds as a great stimulus to human auditory sense. However, to date, the issue of interlayer noise has been the subject of concern because the interlayer noise criterion limits the average size of a certain section and there is no standard for impact noise that appears temporarily. Therefore, in this paper, A7B shows the number of impulse sounds in a row in order to make objective visual data about the interlaminar impact noise. In this case, the criterion of the impact sound is defined as a portion where a large change in sound volume occurs within 0.1 second. Experimental results show that the results similar to the number of impact sounds of the corresponding interlaminar impact noise source can be visually confirmed, and furthermore, it is possible to present clear and objective data in the interstory noise dispute.

TUESDAY AFTERNOON, 5 DECEMBER 2017 STUDIO 2, 3:30 P.M. TO 5:10 P.M.

Session 2pNSb

Noise, Architectural Acoustics, Speech Communication, Psychological and Physiological Acoustics, and ASA Committee on Standards: Acoustics and Its Role in Accessibility (e.g., Americans with Disabilities Act)

David Manley, Chair
D.L. Adams Associates, Inc., 1536 Ogden St., Denver, CO 80218

Chair's Introduction—3:30

Invited Papers

3:35

2pNSb1. Disability rights aspects of ambient noise under the Americans with Disabilities Act for people with auditory disorders. Daniel Fink (The Quiet Coalition, The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

The goals of the disability rights movement are full enjoyment, active inclusion, and equal participation in society for people with disabilities. The Americans with Disabilities Act (ADA) guarantees full and equal enjoyment of places of public accommodation for them. People with partial hearing loss, tinnitus, and hyperacusis meet the ADA definition of having a disability but currently have no legal protection under ADA. People with 25–40 decibel hearing loss cannot understand speech in noisy places, with or without hearing aids. They require ambient noise levels below 60 A-weighted decibels with reverberation times under 0.50 seconds to understand speech. Noise worses symptoms for those with tinnitus and hyperacusis. Noisy restaurants and other indoor places deny full enjoyment and equal participation in public life to people with these auditory disorders. Legislative and regulatory actions are needed to provide quiet environments, with established noise standards vigorously enforced. Technologies and environmental modifications to control noise and reverberation are well known, readily available, and relative inexpensive. The simplest modification is merely turning down the volume of amplified sound. Universal design for quiet facilitates communication for everyone and prevents hearing loss, tinnitus, and hyperacusis in those without auditory disorders.

3:55

2pNSb2. Preliminary reports of New York city restaurant and bar noise levels via crowdsourced measurements show a low number of restaurants and bars suitable for various disabled communities, notably those with hearing loss. Gregory Farber (SoundPrint, PO Box 533, Lincoln, MA 01773, greg@soundprint.co)

Noise causes hearing loss, tinnitus, and hyperacusis, and interferes with understanding speech for those with normal hearing and for those with hearing loss. Media reports document high sound levels in restaurants and bars, but accurate sound measurements are lacking. Noise pollution is an important health and social issue and the public needs a reliable way to determine whether certain venues are safe for auditory health and conducive to conversation. This is important for the disabled, the blind, the autistic and the hearing impaired, who include a large number of older Americans. This report is of sound level measurements in more than 2250 venues in New York City, using the novel SoundPrint smartphone app. The average sound level was 78 dBA in restaurants and 81 dBA in bars. These sound levels do not allow ready conversation, even for those with normal hearing and certainly not for those with hearing loss. There were variations in sound levels in restaurants in different neighborhoods and in different types of restaurants. The reported sound levels by venue managers generally underestimated actual sound levels. This report is a proof-of-concept study of crowdsourced sound measurements, which can provide valuable data for the general public and health officials.

4:15

2pNSb3. Electric vehicles—Safe and quiet with acoustic vehicle 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)

One of the most important environmental issues in densely populated areas is the problem of noise. Traffic noise from cars, railway vehicles and airports located in close proximity to cities is not only annoying for residents; it also leads to serious health issues and has an enormous negative economic impact. 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 and regulation. Currently there is a strong debate here in Germany—as in many other parts of the world—about road vehicles that rely, in whole or in part, on alternative drive trains. On the one hand, there are positive environmental benefits expected from these "hybrid or pure electric" road vehicles. On the other hand, these vehicles are thought to pose a risk to blind and low vision pedestrians, due to their lower noise emissions at approach. 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.

4:35

2pNSb4. Induction loop assisted listening system case study in a Catholic church. David Manley and Benjamin Bridgewater (D.L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

A Catholic church in Metro Denver, Colorado area, was undergoing major renovations for the first time since the 1970s, and wished to incorporate an induction loop assisted listening system as an amenity for their aging members. The pastor indicated the Church has lost a number of parishioners due to the lack of an assisted listening system. While the Church installed a temporary radio frequency assisted listening system until renovations could be complete, a permanent and integrated system was desired. The case study presents coordination with Church, the loop system manufacturer, testing of the existing space for metal loss, and design results.

Contributed Paper

4:55

2pNSb5. On identifying the accidental sudden acceleration of an SUV vehicle by engine sound. Jihye Bae and Myungjin Bae (Commun. Eng., Soongsil Univ., 369 sangdoRo, DongjakGu, Seoul 156-743, South Korea, jbae5@ssu.ac.kr)

A major accident occurred when an SUV vehicle of the company A that recently ran the road was flooded with roadside trees on the road. It was suspected of a sudden accident, but Company A judged the driver immature because the driver did not step on the brake paddle at that moment. The situation before and after the accident was recorded in the black box video, but the engine sound was not heard well. However, we analyzed the curve of the change in engine sound while emphasizing the change of the engine sound at the moment of the accident by recording the voice in the black box video. In particular, when comparing the curve of the engine sound change below 1,000 Hz with the rapid acceleration curve of the existing engine sound, for 5 seconds before the accident, the rotation sound of the engine changed 1.7 times faster than the rapid acceleration, and the highest rpm lasted 3 seconds. After all, in terms of the engine sound change curve of the black box video, this was a sudden velocity accident. In this paper, we proposed a new method to identify the accident through engine sound.

TUESDAY AFTERNOON, 5 DECEMBER 2017 — BALCONY L, 1:00 P.M. TO 3:00 P.M.

Session 2pPA

Physical Acoustics: General Topics in Physical Acoustics I

Kyle S. Spratt, Chair
Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713

Contributed Papers

1:00

2pPA1. Champagne bubble acoustics. Kyle S. Spratt, Kevin M. Lee, and Preston S. Wilson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, sprattkyle@gmail.com)

Sparkling wine, such as the variety coming from the Champagne region of France, is a beverage that is at least partially famous for its carbon dioxide bubbles, a byproduct of the secondary fermentation process that occurs after bottling. A well-known theory, though hardly accepted universally, posits that the quality of a sparkling wine can be ascertained from the characteristics of its bubbles, such as bubble size distribution and rate of production. This talk describes a preliminary investigation to monitor the characteristics of sparkling wine bubbles using passive acoustic measurements, wherein bubble parameters are estimated from the power spectral density of the ambient bubble noise. Measurements made on a variety of sparkling wines will be presented.

1:15

2pPA2. Temperature and frequency dependent behavior of high intensity focused ultrasound (HIFU)-induced shear waves in a viscoelastic micellar fluid. E. G. Sunethra K. Dayavansha, Cecille Labuda, and Joel Mobley (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, MS 38677, sdayavan@go.olemiss.edu)

Wormlike micellar (WM) fluids, which flow when subjected to long term stresses, are mechanically viscoelastic over shorter durations. These fluids are birefringent under shear, allowing the study of shear wave propagation using both optical and acoustic modalities. In this work, the thermal and spectral behavior of ultrasonically generated shear waves in a WM fluid are studied. A fluid consisting of hexadecyltrimethylammonium bromide (CTAB) and sodium salicylate (NaSAL) combined in a 5:3 ratio is used in a 200 mM concentration. A high intensity focused ultrasound (HIFU) beam generates radiation pressure in the fluid and can induce shear waves of sufficient amplitude to be visualized optically when the beam is modulated. By pulsing the HIFU beam, a train of shear waves are generated which propagate laterally from the focal region. The temperature and frequency dependent behavior of the HIFU generated shear waves are correlated with the rheological and microstructural properties of the fluid.

1:30

2pPA3. Phase transition characterization based on acoustic reverberation time. Hossep Achdjian, Julien Bustillo, Laurianne Blanc, Andres Arciniegas, Nicole Doumit (GREMAN, Tours Univ., CNRS, INSA-CVL, Blois, France), and Marc Lethiecq (GREMAN, Tours Univ., CNRS, INSA-CVL, Bat E, 20 Ave., Monge, Parc de Grandmont, Tours 37000, France, marc.lethiecq@univ-tours.fr)

Non-destructive monitoring of a material's state during its physicochemical transformations is of interest for several industrial fields including food processing, such as milk-derived products, or cosmetics. The use of ultrasound to provide reliable information about physicochemical properties is becoming increasingly popular. Indeed, ultrasonic techniques have the main advantage of being rapid and non-invasive methods that allow parameters such as product composition, structure, and physical state to be obtained. Yet, classical techniques are limited to the characterization of the medium along the propagation path using the first wave packets. Here, we propose an alternative technique based on the study of the reverberated waves, classically used in room acoustics. In previous works, this method has shown its capability to characterize materials as well, with the advantage over classical techniques to address a medium in its whole structure. In this context, the determination of sol-gel phase transition time of Salol using this method is presented. Measurements are performed in an aluminum mold using five piezoelectric (PZT) patches, one being used as emitter and the others as receivers. The mean reverberation time over four receivers has been studied and its evolution is shown to lead to a good estimation of the phase transition time.

1:45

2pPA4. Rapid and simple measurement of hypersonic wave velocity by Brillouin scattering with induced phonons. Yoshiaki Shibagaki (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, duq0358@mail4.doshisha.ac.jp), Shinji Takayanagi (Nagoya Inst. of Technol., Nagoya, Japan), Masahiko Kawabe (Doshisha Univ., Kyotanabe, Japan), Takahiko Yanagitani, Masashi Suzuki (Waseda Univ., Tokyo, Japan), and Mami Matsukawa (Doshisha Univ., Kyotanabe, Japan)

Brillouin scattering is a non-contact method to measure wave velocities in the GHz range. One problem of the Brillouin scattering technique is weak light scattering from thermal phonons, which results in the long measurement time and necessity of a complex tandem Fabry-Perot interferometer. To overcome this problem, we have proposed techniques to make use of induced strong coherent phonons from a high frequency transducer. In this study, we have tried to induce strong longitudinal coherent phonons by a ScAlN film transducer (composition of the film : $Sc_{0.41}Al_{0.59}N$), which has a high electromechanical coupling coefficient. The transducer was fabricated on the quartz sample and composed of the ScAlN film grown by an RF magnetron sputtering and electrodes. The transducer was deposited on one side of the sample. Due to the induced phonons, the scattered light became much stronger than those of thermal phonons and could be observed by a simple confocal Fabry-Perot interferometer. The measured frequency shift of the Brillouin scattering peak was equal to the excitation frequency of the ScAlN transducer (883 MHz). This technique enables easy, rapid and simple measurement of wave velocity in the GHz range, which can be applied for the 2D velocity imaging of the sample.

2:00

2pPA5. Resonant ultrasound spectroscopy measurement of elastic properties of SnSe. Ashoka Karunarathne, Josh R. Gladden, Gautam Priyadarshan (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of MS, 145 Hill Dr., University, MS 38677, atholang@go.olemiss.edu), Pai-Chun Wei, Yang-Yuan Chen (Inst. of Phys., Academia Sinica, Taipei, Taiwan), Sriparna Bhattacharya, and Apparao M. Rao (Dept. of Phys. and Astronomy, Clemson NanoMater. Inst., Clemson Univ., Clemson, SC)

Resonant Ultrasound Spectroscopy (RUS) is a precise experimental approach for investigating the elastic properties of solid materials at different temperatures and hydrostatic pressures. In RUS, the elastic stiffness tensor of crystalline solids with different crystal structures are determined from their vibrational resonance spectra. A study of the mechanical properties of thermoelectric materials can provide insights to their high efficiency and low thermal conductivity. SnSe is a thermoelectric material which exhibits a high efficiency and low thermal conductivity due to its anharmonicity and low symmetric crystal structure. SnSe exhibits a structural phase transition from *Pnma* to *Cnma* at ~800 K, and the single layered orthorhombic crystal structure of SnSe results in nine independent elastic constants. In this study, RUS was used to determine the temperature dependent elastic properties of polycrystalline SnSe through its phase transition temperature. Use of RUS technique to better understand the temperature dependent elastic properties of low symmetry crystals will be discussed. The elastic constants of single crystal SnSe calculated by RUS at room temperature are in agreement of within ~35% of theoretical reported values.

2:15

2pPA6. Focal zone characteristics of stepped Fresnel and axicon acoustic lenses. Robert L. Lirette (Phys., Univ. of MS, 2400 Anderson Rd., Apt. 4, Oxford, MS 38655, rlirette@go.olemiss.edu) and Joel Mobley (Phys., Univ. of MS, University, MS)

Flat Fresnel and axicon acoustic lenses were developed and characterized both numerically and experimentally. Flat lenses have a compact profile leading to less attenuation and phase distortion in the bulk lens material. Fresnel lenses approximate spherical focusing to a fixed point whereas the axicon focuses to a narrow focal line, producing a tighter lateral beam over a longer depth. Numerical models of the sound pressure field from both types of lenses were done using the angular spectrum method. The lenses were 3d printed with polylactic acid (PLA) and also lathe cut with transparent polystyrene. Pulse-echo measurements of the sound speed were done for both materials. Pressure field scans were conducted using a 1.2 MHz planar transducer in a hydrophone scanning tank. These scans demonstrate the focusing effect of both types of lenses and are in agreement with the numerical model.

2:30

2pPA7. Focus control in the radial direction using an ultrasound liquid crystal lens. Yuki Shimizu (Graduate School of Sci. and Eng., Graduate School of Doshisha Univ., Tataramiyakodani1-3, Kyotanabe, Kyoto 610-0321, Japan, duq0360@mail4.doshisha.ac.jp), Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), Yuki Harada (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), Akira Emoto, and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

Nematic liquid crystal is widely used in optical devices such as liquid crystal displays and controlled by electric fields through the liquid crystal layer. Our group has proposed a control technique of liquid crystal molecular orientation using ultrasound vibration without indium tin oxide electrodes [1]. In this study, using ultrasound and nematic liquid crystal, we realized a variable focus liquid crystal lens. The liquid crystal lens has no

transparent electrodes and consists of a simple structure; a nematic liquid crystal layer with a thickness of 50 μm is formed between two circular grass plates. An annular PZT ultrasonic transducer was bonded on the glass plate. By exciting the transducer at the resonance frequencies (28 kHz), the flexural vibration modes were generated. The acoustic radiation force from the vibration changed the molecular orientation of the liquid crystals, which induced the distribution of the execution refractive indices. The liquid crystal layer then worked as a lens. The focal points could be controlled by the radiation force and moved in the radial direction. [1] S. Taniguchi, D. Koyama, Y. Shimizu, A. Emoto, K. Nakamura, and M. Matsukawa, *Appl. Phys. Lett.* **108**(2016), 101103.

2:45

2pPA8. Porous lead zirconate titanate ceramics with optimized acoustic properties for high-frequency ultrasonic transducer applications. Marc Lethiecq (GREMAN UMR 7347, Tours Univ., CNRS, INSA-CVL, Bat E, 20 Ave. Monge, Parc de Grandmont, Tours 37200, France, marc.lethiecq@univ-tours.fr), Danjela Kuscer (Electron. Ceramics, Jozef Stefan Inst., Ljubljana, Slovenia), Julien Bustillo (GREMAN UMR 7347, Tours Univ., CNRS, INSA-CVL, Blois, France), Andre-Pierre Abellard (Ctr. des Matériaux UMR7633 CNRS, Mines Paris Tech, Evry, France), Tina Bakaric (Electron. Ceramics, Jozef Stefan Inst., Ljubljana, Slovenia), and Franck Levassort (GREMAN UMR 7347, Tours Univ., CNRS, INSA-CVL, Tours, France)

Acoustic properties of porous PZT ceramics intended to be used as backing in high-frequency transducer applications are investigated using a novel method where an electroded piezoelectric thick film is deposited on the backing under test. Two backings with pore sizes 1.5 μm and 10 μm were obtained by sintering a mixture of ceramic powder and an organic template, their porosity was evaluated by scanning electron microscopy at 15%, leading to a density around 6.5 g/cm^3. The electroacoustic impulse responses of these devices were measured considering the backing as a propagation medium, the initial thickness of which was chosen small enough to allow back-wall echoes to be detected and large enough to be able to separate the signals in time domain. Then the thickness of the backing was reduced (from around 2 mm to less than 1 mm) and the measurements were repeated. Acoustic properties were then deduced: attenuation coefficients reaching 4 dB/mm/MHz and group velocities around 3400 m/s were obtained, leading to an acoustic impedance around 22 MRa. Such combination of high attenuation and moderate acoustical impedance make these materials an interesting solution for high-resolution ultrasonic imaging transducers.

TUESDAY AFTERNOON, 5 DECEMBER 2017 SALON A/B/C, 1:00 P.M. TO 4:30 P.M.

Session 2pSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Engineering Acoustics: Acoustic Metamaterials II

Christina J. Naify, Cochair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr., Pasadena, CA 91109

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

1:00

2pSA1. Zone folding induced topological insulators in phononic crystals. Yuanchen Deng (North Carolina State Univ., 911 Oval Dr., EB III, Campus box 7910, Raleigh, NC 27695), minghui lu (Nanjing Univ., Nanjing, Jiangsu, China), and Yun Jing (North Carolina State Univ., Raleigh, NC, yjing2@ncsu.edu)

This study investigates a flow-free, pseudospin-based acoustic topological insulator. Zone folding, a strategy originated from photonic crystal, is used to form double Dirac cones in phononic crystal. The lattice symmetry of the phononic crystal is broken by tuning the size of the center "atom" of the unit cell in order to open the nontrivial topological gap. Robust sound one-way propagation is demonstrated both numerically and experimentally. This study provides a flexible approach for realizing acoustic topological insulators, which are promising for applications such as noise control and waveguide design.

1:15

2pSA2. Acoustic performance of phononic crystals for underwater coating applications. Gyani Shankar Sharma (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, Sydney, NSW 2052, Australia, gyanishankar.sharma@student.unsw.edu.au), Alex Skvortsov, Ian MacGillivray (Maritime Div., Defence Sci. and Technol. Group, Melbourne, VIC, Australia), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, Sydney, NSW, Australia)

Acoustic coatings on maritime vehicles can significantly reduce the transmission of machinery noise in the ambient marine environment as well as absorb external acoustic waves. In this work, the performance of two types of phononic crystals with steel backing is investigated for acoustic coating applications. The first type of phononic crystal comprises periodic voids embedded in a soft elastic medium. The second type of phononic crystal comprises hard scatterers arranged periodically in an elastic medium.

The voids exhibit monopole resonance, leading to low sound transmission through the coating in a broad frequency range. In contrast, the hard scatterers exhibit dipole resonance which results in low sound reflection. The ratio of bulk to shear moduli of the elastic medium governs the monopole resonance, whereas the dipole resonance is governed by the ratio of the density of the scatterers to the density of the host medium. The effect of steel backing on the transmission side of the elastic medium results in high values of sound absorption attributed to Fabry-Perot resonance. The advantages and limitations of the two types of acoustic coatings are discussed.

1:30

2pSA3. Optimizing sound transmission through 3D pentamode materials in water. Xiaoshi Su and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, xiaoshi.su@rutgers.edu)

Pentamode materials (PMs) are engineered isotropic or anisotropic elastic structures that mimic the acoustic behavior of ideal fluids. Recently, many applications such as negative refraction lens, focal lens, and a cloak have been designed and demonstrated using 2D PMs. We consider metallic lattice structures in which only one wave mode propagates over a wide frequency band. Unlike the 2D PM which naturally has a boundary, the three dimensional PM needs to be sealed to isolate the interior structure from water. However, the material used for encapsulation alters the effective surface impedance of the material. In this talk, we present 3D PMs with acoustic properties matched to water and show that the transmission is sensitive to the thickness of the boundary plate. An analytic model is established to provide some insights into the boundary transmission. Iterations were performed in FEM simulations to achieve optimal transmission within the frequency range of interest (4 kHz–10 kHz). The model PM with partial boundary is made using 3D metal printing, and the remaining boundary is added post fabrication. Preliminary experimental results will be discussed. [Work supported by ONR/MURI.]

1:45

2pSA4. Numerical investigation of metafluid behavior in pre-strained negative stiffness honeycombs. Benjamin M. Goldsberry and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu)

Negative stiffness honeycombs (NSH) are doubly periodic arrays of curved beams that have recently been shown to exhibit a reversible pseudoplastic large deformation response and are therefore interesting candidates for impact isolation applications. The authors previously used finite element analysis to show that NSH exhibit three types of propagating modes described by (i) longitudinal, (ii) transverse, and (iii) rotational displacements with respect to the propagation direction. The phase and group velocities of each mode can be significantly altered by applying a uniaxial pre-strain imposed on the top and bottom boundaries of the NSH. The present work will show that the transverse and rotational mode phase speeds are significantly more sensitive to strain than the longitudinal modes for small but finite pre-strain levels. Consequently, for some pre-strain levels there exists a large frequency band where only longitudinal waves propagate. The pre-strained NSH therefore behaves as a pentamode metamaterial, or metafluid, and may represent a new type of structure to realize devices designed using transformation acoustics. The effect of the geometry and pre-strain on the existence and frequency range of unimodal propagation in pre-strained NSH will be investigated. [Work supported by NSF.]

2:00

2pSA5. Enhancement of low-frequency sound emission by metamaterial enclosures. Likun Zhang (Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, zhang@olemiss.edu), Jiajun Zhao (King Abdullah Univ. of Sci. and Technol. (KAUST), Austin, Texas), Ying Wu (King Abdullah Univ. of Sci. and Technol. (KAUST), Jeddah, Saudi Arabia), and Maryam Landi (Dept. of Phys. and Astronomy, Univ. of MS, Oxford, MS)

Emission of low-frequency sound by a source is limited by the size of the source relative to sound wavelength. We propose to place the source in a subwavelength enclosure of anisotropic metamaterials which enhances the emission around Mie resonant frequencies of that enclosure [J. Zhao, L. Zhao, and Y. Wu, *J. Acoust. Soc. Am.* **142**(1), EL24–29, July 2017]. The enclosure has a relatively small sound speed along the radiation direction, enabling the resonances to occur at low frequencies. Our numerical simulations provide evidence of the proposed low-frequency enhancement for both monopole and multipole sources. The common Mie resonant frequencies increase with the order of the multipoles. Here, we introduce an extreme anisotropy in the azimuth of the enclosure to degenerate the resonant frequencies of higher-order multipoles down to the same frequencies as the lower-order multipoles. The degeneracy associated with the anisotropy of the enclosure is theoretically analyzed. The results guide the experimental realization of the enclosure for low-frequency enhancement.

2:15

2pSA6. Experimental realization of enhancing low-frequency sound emission by metamaterial enclosure. Maryam Landi, Likun Zhang (Univ. of Mississippi/National Ctr. for Physical Acoust. (NCPA), 145 Hill Dr., University, MS 38677, mlandi@go.olemiss.edu), Jiajun Zhao (Div. of Comput., Elec. and Mathematical Sci. and Eng., King Abdullah Univ. of Sci. and Technol. (KAUST), Austin, Texas), and Ying Wu (Div. of Comput., Elec. and Mathematical Sci. and Eng., King Abdullah Univ. of Sci. and Technol. (KAUST), Jeddah, Saudi Arabia)

A solution to the low radiation rate of a sound source has been proposed by using metamaterial enclosures of degenerated Mie resonances [J. Zhao, L. Zhao, and Y. Wu, *J. Acoust. Soc. Am.* **142**(1), EL24–29, July 2017]. This talk presents our experimental realization of this enhancement for a monopole source. The enclosure with a low sound speed is created by a coiling-space structure fabricated by 3D printing. A small balanced-armature speaker with a 1 mm speaker mouth is placed at the enclosure center to radiate impulsive sound signals. The diameter of the structure is only one seventh of the wavelength of the sound at the maximum enhancement, corresponding to the lowest resonance. The enhanced sound pressure is more than two orders of magnitude larger than the sound pressure without the enclosure. The measurements of sound pressure at different angles in the space show the omnidirectionality of the enhancement. This enhancement is demonstrated as a result of enhanced density of states, i.e., the increased number of modes per unit frequency range and per unit volume, which we relate to Fermi's Golden rule and Purcell effect in quantum systems and find the analog in acoustics.

2:30–2:45 Break

2:45

2pSA7. Compressibility-near-zero acoustic supercoupling. Matthew S. Byrne (Elec. and Comput. Eng., Univ. of Texas at Austin, 1616 Guadalupe St., Austin, TX 78751, mbyrne@utexas.edu), Hussein Esfahlani (Ecole Polytechnique Fédérale de Lausanne, Lausanne, Switzerland), Matthew J. McDermott (Mech. Eng., Univ. of Texas at Austin, Austin, TX), and Andrea Alu (Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX)

Epsilon-near-zero supercoupling is a widely-researched topic in electromagnetics. This phenomenon takes advantage of media with near-zero dielectric permittivity to build waveguide coupling channels which can, in principle, support unitary transmission and complete phase uniformity, independent of the length and height of the coupling channel. In this effort, we present the possibility that an analogous, extraordinary coupled-mode phenomenon exists in acoustics, which we call compressibility-near-zero (CNZ) supercoupling. Prior works have required the presence of periodic membrane resonances in order to observe near-zero index coupling. We have shown, for the first time, theoretically and experimentally, that the phenomenon can be observed without the use of membranes. We envision this development as a first step towards experimentally realizing spatiotemporally-modulated acoustic non-reciprocal devices which take advantage of these physical principles.

2p TUE. PM

3:00

2pSA8. Acoustic cloak by inverse design and gradient based optimization. Feruza Amirkulova (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, feruza.amirkulova@wne.edu) and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

We present a gradient based minimization of the total scattering cross section (TSCS) from a set of cylindrical obstacles by incrementally repositioning them so that they eventually act as an effective cloaking device. The idea differs from earlier inverse designs that use topology optimization tools and generic algorithms. We use the optical theorem to define the position dependent TSCS in terms of the forward scattering amplitude for an incident plane wave. The gradient-based optimization algorithm reduces the TSCS by evaluating its derivative with respect to the cylinder positions and then perturbatively optimizing the position of each cylinder in the cloaking device while taking into account acoustic the multiple scattering between the cylinders. The method is illustrated by examples of sets of hard cylinders of uniform size.

3:15

2pSA9. Unidirectional sound manipulation with acoustic metamaterials. Jie Zhu (Hong Kong Polytechnic Univ., Kowloon 00000, Hong Kong, jiezhu@polyu.edu.hk)

Abstract: Unidirectional sound propagation introduces strong direction-dependent responses. Such effect has drawn considerable attentions in acoustics. In this presentation, we introduce some of the interesting unidirectional sound manipulation phenomenon that is achieved by acoustic metamaterials. The experimentally observed phenomenon is associated with the enhanced wave-structure interactions. We can predict and tune the operating frequency by further trimming the material's geometry. Our demonstration provides a new degree of freedom to the realization of unique wave dynamics for applications like noise control, acoustic sensing, and imaging. [This work was supported by the Early Career Scheme (ECS) of Hong Kong RGC (Grant No. 25208115).]

3:30

2pSA10. Double-decorated membrane structure: Initial efforts towards real-time phase modulation. Reza Ghaffarivardavagh (Mech. Eng., Boston Univ., 8 saint Mary St., Photonic Bldg., Rm. 832, Boston, MA 02215, ghaffari@bu.edu), Stephan Anderson (Radiology, Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., Boston, MA)

One of the fundamental challenges in the practical implementation of acoustic metamaterials science is the issue of tunability. To date, proposed acoustic metamaterial structures and devices suffer from narrow-band working frequency are typically passive structures. For example, with regard to phase modulation using metasurfaces, the tunability and capability of real-time phase modulation is critical in numerous applications ranging from biomedical ultrasound to acoustic communication. The study presented herein represents an initial step toward realizing acoustic metamaterials with real-time tunability intended for phase modulation applications. Herein, the double-decorated membrane structure (DDM) will be introduced, in which fundamental resonance of two membranes have been tailored to generate constructive interference, enabling phase modulation across a large range, while the amplitude variation has been mitigated. In this study, analytical and numerical investigations have been conducted to explore and optimize the design parameters. The proposed structure can be also used for the aim of low frequency sound energy harvesting in which the DDM structure will broaden the working frequency band and, consequently, enhance the energy conversion efficiency.

3:45

2pSA11. A new approach to generate local resonator for the application of acoustic/elastic metamaterials. Sheng Sang (System Eng., Univ. of Arkansas at Little Rock, 1701 Westpark Dr., Apt. #110, Little Rock, AR 72204, sxsang@ualr.edu)

In this paper, we proposed a type of locally resonator which works by introducing more than one vibration modes, and thus can provide a new approach to formulate acoustic or elastic metamaterial. Such new concept of resonator is fundamentally different from traditional translational and rotational resonators. The potential application and design of acoustic metamaterial based on this kind of structures are also studied both analytically and numerically. The acoustic metamaterial filter we developed has proved to have unique property that only allow the waves with specific frequency to pass while other waves will be attenuated. We also proposed a kind of plane wave lens which can transfer plane wave with uniformly distributed amplitude profiles into pure plane wave. With this novel resonator, the research and application regarding it are highly anticipated in the near future.

4:00

2pSA12. Active vibration control by a tunable piezoelectric shunt metamaterial. Aaron J. Stearns (Mech. Eng., Applied Res. Lab., Penn State Univ., 225 Sci. Park Rd., State College, PA 16803-2213, ajs6037@psu.edu) and Benjamin Beck (Structural 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. Primarily through periodic subwavelength resonant inclusions, acoustic metamaterials can enable steering, cloaking, lensing, and frequency band control of acoustic waves. However, a common drawback of acoustic metamaterials is that effectiveness is limited to narrow frequency bands. Thus, investigation of practical active and adaptable acoustic metamaterials is valuable in achieving wider operation frequency bands. Here, a metamaterial consisting of active tunable piezoelectric shunts is investigated numerically and experimentally from the unit cell level. A physical model of the unit cell is developed using the finite element method. From the finite element model, the wave finite element method is applied to compute the dispersion and forced response of the periodic structure. It is demonstrated that the shunts introduce an additional degree of freedom by which adaptable bending wave attenuation can be accomplished. Since the periodic shunts are only effective at certain frequency bands, a known optimization method is implemented to tune the shunts. Additionally, a new optimization scheme is compared to the existing scheme found in literature.

4:15

2pSA13. Digital design of cellular solids for noise and vibration mitigation. Mark J. Cops, James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, mcops@bu.edu), and Elizabeth A. Magliula (NAVSEA, Newport, RI)

Cellular solids are of large interest in structural vibration due to their high strength-to-weight ratio and ability for high energy absorption. This presentation describes concepts for digitally designing cellular solids and demonstrates the ability of this design method for tuning effective material properties for noise and vibration mitigation applications. The designs can be created by defining a complete topology mathematically or by specifying section cut-outs from a solid host material. The digital designs are then analyzed with finite element software to determine effective material properties. This approach is advantageous because it allows for an automated computer procedure for designing and characterizing materials. In addition, resulting designs can be fabricated through either 3D printing processes or CNC milling. In this presentation, the relationship between geometrical/physical properties and effective static material properties (such as Young's modulus and Poisson ratio) of cellular solids will be discussed for the digital designs, with reference to existing theory.

Session 2pSC

Speech Communication: Speech Production (Poster Session)

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

Contributed Papers

2pSC1. Tracking larynx movement in real-time MRI data. Miran Oh (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu), Asterios Toutios (Elec. Eng., Univ. of Southern California, Los Angeles, CA), Dani Byrd, Louis Goldstein (Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth S. Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Ejective and implosive consonants involve rapid raising or lowering, respectively, of the larynx coordinated with an oral constriction formation and release. Techniques such as contour tracking and region-of-interest (ROI) direct image analysis are becoming established for examining supralaryngeal constriction formation in real-time MRI data, but quantification of larynx movement from dynamic vocal-tract magnetic resonance imaging has not been tackled. We will evaluate methods of indexing the movement of the larynx in fast real-time MRI data collected in the midsagittal plane. We analyze data from USC's real-time MRI IPA database produced by several phoneticians and data from speakers of the Hausa language producing both ejectives and implosives. Kinematic profiles of the laryngeal movements are obtained by outlining the arytenoid structure and applying principal component analysis on the vocal-tract outlines. In addition, vertical movement of the larynx is indexed by calculating the time-varying pixel intensity centroid (i.e., intensity-weighted average spatial position) in a rectangular ROI defined for larynx movement. Once direct measurement of the larynx actions is validated for this type of imaging data, further work will focus on the temporal coordination of the laryngeal raising/lowering gestures and the supralaryngeal constriction gestures that occur in glottalic airstream consonants. [Work supported by NIH.]

2pSC2. Toward a dynamic theory of vowel production. Fang Hu (Inst. of Linguist., Chinese Acad. of Social Sci., 5 Jian Guo Men Nei St., Beijing 100732, China, hufang@cass.org.cn)

Vowels are traditionally classified into monophthongs, diphthongs, triphthongs, and in some rare cases, tetraphthongs. However, there is a long debate on the complexity of vowels in the literature. Some phoneticians view diphthongs as a single vowel with phonetically complex nucleus, while others treat diphthongs as a sequence of two vowel elements. This paper reports recent developments in vowel researches in Chinese dialects, and argues for an integral account for vowel production. First, fine-gained phonetic details from Mandarin, Wu, Jin, Hui, Min, and Hakka reveal that falling diphthongs and rising diphthongs differ in temporal organization, spectral property, and spectral dynamics. Second, there is no dichotomy, but a continuum between monophthongs and diphthongs. Comparisons between four Hui dialects reveal how the process of diphthongization develops from a more monophthong-like stage to a more diphthong-like stage. In conclusion, the production data from Chinese dialects support an integral account for vowel production. Monophthongs are composed of a static spectral target, diphthongized vowels and falling diphthongs are composed of a dynamic spectral target, and rising diphthongs are sequences of two spectral targets.

2pSC3. Using machine learning to identify articulatory gestures in time course data. Will Styler, Jelena Krivokapic (Dept. of Linguist, Univ. of Michigan, 611 Tappan St., 440 Lorch Hall, Ann Arbor, MI 48109, wstyler@umich.edu), Ben Parrell (Linguist and Cognit. Sci., Univ. of Delaware, Newark, DE), and Jiseung Kim (Dept. of Linguist, Univ. of Michigan, Ann Arbor, MI)

One difficulty in working with articulatory data is objectively identifying phonological gestures, that is, distinguishing targeted gestural movement from general variability. Although human annotators are generally used, an automated approach to identifying meaningful patterns offers advantages in speed, consistency, and objective characterization of gestures (cf. Shaw and Kawahara 2017). This study examines Electromagnetic Articulography (EMA) data from seven American English speakers, aiming to identify and characterize pause postures (specific vocal tract configurations at prosodic boundaries; Katsika *et al.* 2014). Supervised machine learning using kernelized Support Vector Machine Classifiers (SVMs) took as training data 852 trajectories from three speakers analyzed to date, containing 104 pause postures identified by a human annotator, and classified the between-words lip aperture (LA) trajectory to identify tokens containing the pause posture, while also providing token-by-token gesture probability. Features of the curvature were extracted using Principal Component Analysis (PCA) and Discrete Cosine Transform (DCT) of both the actual LA trajectory and of the deviation from a direct word-to-word interpolation. The SVM achieves 94.0% classification accuracy in cross-validation tests, with Cohen's Kappa showing machine-to-annotator agreement of 0.978. These methods of machine-learning-based curve classification are potentially useful and applicable to any time-course articulatory data. [Work supported by NIH and NSF.]

2pSC4. Distributional factors in Telugu sibilant production. Charles Redmon, Allard Jongman, and Jie Zhang (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., Rm. 427, Lawrence, KS 66046, redmon@ku.edu)

Telugu is one of a small set of languages described as exhibiting three or more place contrasts among sibilant fricatives (less than 4% of languages in UPSID; Maddieson and Precoda, 1990). In a pilot study of alveolar, palatal, and retroflex sibilant productions in VCV sequences, Telugu speakers were found to show consistent inter-speaker variation in the production of the palatal sibilant, with half showing evidence (in production and perception) of a complete merger with the alveolar, and the other half showing evidence of a merger with the retroflex, and no clear demographic differences underlying the delineation of the two groups. Following up on this result, we present acoustic data from native speakers of Telugu recorded in Hyderabad producing words with the three sibilants in multiple vowel contexts,

positions (CV, VC), and lexical neighborhood structures (with vs. without sibilant competitors in a second order neighborhood), focusing in the acoustics in particular on formant transition cues to palatal and retroflex places. This data are presented both to clarify the present state of the Telugu sibilant system and to explore what effect the lexical distribution of these sounds has on the acoustic feature mapping of the contrast.

2pSC5. Uniformity of inherent vowel duration across speakers of American English. Colin Wilson (Johns Hopkins, Krieger 247, 3400 North Charles St., Baltimore, MD 21218, colin.wilson@jhu.edu) and Eleanor Chodroff (Northwestern Univ., Evanston, IL)

Vowel duration is determined by a number of factors in American English (e.g., Klatt, 1976), including the tense vs. lax distinction (e.g., /i/-/ɪ/, /u/-/ʊ/) and the relation between duration and vowel height (e.g., Lehiste, 1970). A large body of research has identified segment-internal factors in data averaged over many speakers (e.g., Crystal and House, 1988; Hillenbrand *et al*., 1995), but few studies have investigated the inherent vowel duration patterns of individuals (cf. House, 1961). Stressed vowel productions (>2 million tokens) were identified by forced alignment in connected speech recordings of 390 speakers (209 female) from the Mixer 6 corpus (Chodroff *et al*., 2016). For each speaker, mean durations of ten stressed vowels (/i ɪ e ɛ æ a ʌ o ʊ u/) were calculated after outlier exclusion. The resulting duration patterns were strongly correlated across pairs of speakers (Pearson *r*: mean = 0.936, 95% CI [0.935, 0.936], range [0.559, 0.999]), and PCA identified a single component, plausibly indexing global speaking rate, that accounted for more than 82% of the variance. These results establish that inherent vowel duration is highly uniform across speakers, a type of structured phonetic variation that has implications for models of perceptual adaptation.

2pSC6. Phonological neighborhood density and intra-speaker variation in vowel production. Benjamin Munson, Tait Anderson, and Jayanthi Sasisekaran (Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Words' phonetic characteristics are affected by numerous lexical factors. Wright (2004) showed that words' phonological neighborhood density (PND) affects the production of vowels. Vowels in words with high PND are hyperarticulated relative to the same vowels in low-PND words. Wright interpreted this as evidence that speech production accommodates listeners' presumed needs: high-PND words are hyper-articulated because they are generally harder to perceive than low-PND words (Vitevitch & Luce, 1998). This effect has been the subject of numerous recent studies, including ones by Munson and Solomon (2004) and Munson (2013) that replicated the effect, and ones by Buz and Jaeger (2016) and Gahl, Yao, and Johnson (2012) that did not. One possible explanation for Wright's original effect is that the ease with which low-density words are perceived allows speakers more freedom to produce the sounds in them variably, including under-articulating them. If this is true, then we would predict that the acoustic characteristics of low-PND words would show more trial-to-trial variability than those of high-PND words. To examine this, we recorded 19 speakers saying 10 repetitions each of high- and low-PND CVC words. The F1 and F2 frequencies of the vowels were measured. For each token, the distance from the center of each speaker's F1/F2 space was computed. The coefficient of variation for this measure was higher (i.e., more variable) for low-PND words than for high-PND words, $t[18]=-2.80$, $p=0.011$.

2pSC7. Acoustic features of spontaneous and read conversational and clear speech produced in simulated acoustic environments. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), and Jessica J. Staples (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Speech production can differ depending on how speech is elicited (e.g. spontaneous speech, read text, speaking style instructions, the speaking environment). In most studies in which different speaking styles have been elicited via instruction (e.g., clear speech) or via the speaking environment (e.g., Lombard speech), the talkers have read printed materials. There is evidence that the acoustic features of clear speech elicited by reading are similar to those observed in semi-spontaneous interaction between two interlocutors, but that clear speech changes are of a greater magnitude in the read speech than the semi-spontaneous speech [V. Hazan and R. Baker, *J. Acoust. Soc. Am.* **130**(4), 2139–2152, 2011]. The present study examines the effects of two speaking style instructions (conversational and clear) and four simulated listening environments (quiet, 55 dB SPL of white noise, 63 dB SPL of white noise, and a reverberant environment) presented via earphones for three types of speech materials: read sentences, read passages, and spontaneously produced picture descriptions. Acoustic features relevant to clear speech and Lombard speech will be compared among the three material types.

2pSC8. Vocalic processes in Southern Ute. Viktor Kharlamov (Florida Atlantic Univ., 777 Glades Rd., CU-97, Ste. 280, Boca Raton, FL 33431, vkharlamov@fau.edu) and Stacey Oberly (Southern Ute Indian Montessori Acad., Ignacio, CO)

Most of the Native American languages are severely endangered and lack detailed phonetic descriptions. Our research aims to document the sound system of Southern Ute, a Numic language of the Ute-Aztecan family spoken by approximately 40 elders in southwestern Colorado, for which only basic impressionistic descriptions are available in the published grammar of the language (Givon, 1980). In this acoustic study, we focus on Southern Ute vowels. We analyze over 6,000 vowel tokens produced by 8 fluent speakers and present our findings on a variety of vocalic processes in Southern Ute, including positional allophony, vowel harmony, as well as phonetic aspects of lexical stress and durational differences related to the phonological length distinction.

2pSC9. Determining the place of articulation of Arabic voiceless dorsal fricatives. Sherman D. Charles, Abdullah Alfaifi, Noor Abo-Mokh, Sarah Robinson, and Kenneth de Jong (Linguist, Indiana Univ., 1610 S Dorchester Dr. Apt. 49, Bloomington, IN 47401, sdcharle@indiana.edu)

Previous, mostly impressionistic, examinations of Arabic varieties provide varied descriptions of the dorsal fricative /χ/ place of articulation, ranging from velar to uvular. While some descriptions attribute this variation to dialectal differences, descriptions of the same dialect also vary. 3D ultrasound data were collected from six native speakers of different dialects of Arabic, one each from Syria, Palestine, Faifi (in Saudi Arabia), Egypt, Algeria, and Morocco. Apart from the dorsal fricatives, the corpus included productions of palatal, pharyngeal, and contrasting velar and uvular stops to provide comparative standards for various points of articulation. The Syrian, speaker showed very similar articulations for /χ/ with uvular stops, while the other speakers variably showed a more anterior articulation between the velar and uvular stops. The most anterior articulations were apparent in the Moroccan and Algerian speakers, giving some suggestion of a dialectal difference; however, point of articulation of the dorsal fricative was variable for most of the speakers and not obviously restricted by dialect. The effect of these articulatory observations on the acoustic noise spectra are ongoing.

2pSC10. Vowel-to-vowel coarticulation in Spanish non-words: Effects of stress and consonantal context. Jenna Conklin and Olga Dmitrieva (Linguist, Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, jconkli@purdue.edu)

In spoken language, gestural overlap in speech production regularly leads to coarticulation between neighboring segments, resulting in assimilation measurable by changes in the acoustic parameters. Vowels in adjacent syllables can coarticulate in a phenomenon called vowel-to-vowel coarticulation, which is subject to variation based on environmental factors such as surrounding consonantal context and the placement of stress. This study investigates vowel-to-vowel coarticulation in Spanish in order to better understand the effect of stress and consonantal context on coarticulation. Formant analysis of vowels produced by 20 native speakers of Spanish was used to determine the presence and direction of coarticulation in trisyllabic nonce words with varying stress (/CVCVCV/ words, vowels /e, o/ as targets and /e, i, o, u/ as triggers, /k/ and /p/ contexts). Results showed that both anticipatory and carryover vowel-to-vowel coarticulation was present at vowel

edges in all contexts, but only carryover coarticulation in backness extended to vowel midpoints. Stress placement mediated the coarticulatory effect with select target and trigger combinations in carryover coarticulation, while consonantal context played a greater role in anticipatory coarticulation. When stress played a role, unstressed target vowels were more susceptible to coarticulation, while stressed targets resisted coarticulatory effects.

2pSC11. The phonetic effects of onset complexity on the English syllable. Anna Mai (Linguist, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, acmai@ucsd.edu)

In English, onset segments of the syllable do not participate in the categorical criteria which determine phonological syllable weight, yet they probabilistically impact stress assignment in novel words and influence the composition of weight-based metric verse (Kelly 2004, Ryan 2014). Together these observations suggest a phonetic motivation for the role that onsets play in English's phonological syllable weight system, investigated here. Using 20 rhyming sets of single syllable words minimally differing in onset material (i.e., *rap, trap, strap*), this production study replicates results from the literature showing that the addition of segmental material to the onset accompanies a decrease in vowel duration (Gordon 2005, Ryan 2014). Furthermore, results show that the pitch and amplitude maxima of the syllable occur earlier within syllables containing more onset segments. Pitch and amplitude maxima have previously been implicated in perceptually based accounts of syllable weight (Goedemans 1998), and since stimuli were controlled for stress and categorical weight, these results suggest that the phonetic effects observed reflect general acoustic properties of onset complexity in the language. For these reasons, findings reported here suggest that perceptual acoustic effects intrinsic to onset complexity are exploited by the weight system of English despite exclusion from its categorical criteria.

2pSC12. Acoustics of the tense-lax stop contrast in Semarang Javanese. Scott Seyfarth (Linguist, NYU, New York, NY), Jozina Vander Klok (Linguist and Scandinavian Studies, Univ. of Oslo, Oslo, Norway), and Marc Garellek (Linguist, UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu)

Javanese has a contrast between tense and lax stops. While both tense and lax stops are voiceless and unaspirated, the contrast at least in word-initial position is realized through acoustic differences in the following vowel, including lower f0, breathier voice quality, and higher F1 for the lax stops relative to their tense counterparts. However, previous reports have indicated substantial cross-speaker variation, and in some cases involve differing characterizations of the acoustic contrast, possibly due to small sample sizes. Moreover, it is still unclear whether (and how) this contrast is maintained in word-final position. In this study, we investigate the tense-lax contrast based on audio recordings of 27 speakers of Central Javanese from Semarang, Indonesia who each read 30 or more items with a word-initial or word-final stop in a carrier phrase. Stops and their adjacent vowels were hand-annotated, and measurements were taken including voice onset and offset time, stop closure and release duration, f0, formant frequencies, and spectral tilt and noise levels. Discussion will focus on how the tense-lax contrast is realized in both word-initial and word-final positions, as well as how different acoustic correlates co-vary within and across speakers.

2pSC13. An electromagnetic articulography study of stop-/s/ and /s/-stop clusters in Greek. Evdoxia Doli, Patrick Reidy, and William F. Katz (Callier Ctr. for Commun. Disord., The Univ. of Texas at Dallas, 1966 Inwood Rd, Dallas, TX 75235, exd160330@utdallas.edu)

Previous kinematic studies on consonant clusters suggest that many factors affect their articulatory timing, including speech rate, frequency of occurrence, and prosody. Other possible factors include the direction of articulatory movement (front-to-back, back-to-front), whether independent articulators (e.g., tongue and lips) or single articulators (e.g., tongue tip and tongue body) are involved, as well as the relative sonority of the consonantal segments. In this study, five talkers of Modern Greek produced the clusters /sp/, /ps/, /sk/, and /ks/ in the carrier phrase "Ipa_____pali" (I said______again). These clusters varied in articulatory direction, articulatory independence, and sonority patterns. We used four methods to determine the degree of articulatory overlap. Each method yielded durations (ms) between gestural landmarks, which were used to compute consonant overlap. The methods differed in the ways gestures were interpreted from the velocity signal (e.g., using extrema or 20% threshold) and how consonant overlap was computed. Preliminary results indicate that articulatory direction and relative sonority exerted the strongest effects on speakers' productions, while articulator dependence/independence played no apparent role.

2pSC14. A case report of nasal fricatives with grimacing: Evidence for spectral marking of the /s/-/ʃ/ contrast. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu) and David J. Zajac (Dental Ecology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

A three-year old girl with repaired bilateral cleft lip and palate was observed to produce anterior nasal fricatives (ANFs) with grimacing during video administration of GFTA-3. ANFs are learned maladaptive articulations used to replace oral fricatives. They are produced by occluding the oral cavity and forcing all airflow through an open velopharyngeal (VP) port that results in turbulent noise at the anterior liminal valve of the nose. Although the girl had normal hearing at the time of testing, she had a history of conductive hearing loss and ventilation tubes. Pressure-flow testing confirmed VP dysfunction and a small nasal area during breathing. The girl appeared to grimace more severely during production of targeted /ʃ/ as compared to /s/ sounds. Greater grimacing on /ʃ/ was confirmed by a forced-choice task with three raters who independently viewed randomized video segments of /s/-/ʃ/ words. Spectral moment analysis revealed higher spectral mean, more negative skewness, and higher kurtosis for ANFs substituted for /s/ as compared to /ʃ/ sounds. We conclude that the girl used a nasal grimace as an articulatory gesture—perhaps learned during speech therapy—to spectrally mark the /s/-/ʃ/ contrast by modulating the length and cross-sectional area of the anterior nasal valve. [Research reported in this publication was supported by the National Institute of Dental & Craniofacial Research of the National Institutes of Health under Award Number 1R01DE022566-01A1.]

2pSC15. Indexing tongue profile narrowing for English lateral consonants using 3D volumetric MR imaging. Mairym Llorens, Dani Byrd (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu), Nancy Vazquez (Linguist, California State Univ., Long Beach, CA), Louis Goldstein, Tanner Sorensen (Linguist, Univ. of Southern California, Los Angeles, CA), Asterios Toutios, and Shrikanth S. Narayanan (Eng., Univ. of Southern California, Los Angeles, CA)

Production of lateral consonants in many languages involves separate but coordinated tongue tip and tongue rear actions—raising of the tongue tip and retraction of the tongue body. Given these gestures and the presence of lateral airflow, it has been speculated that horizontal (i.e., side-to-side) narrowing of the tongue may occur during production of these laterals, either as a passive result of the anterior-posterior lingual stretching or as an actively controlled movement. This study uses 3D volumetric MR scans of speakers producing a variety of sustained English sounds to examine horizontal tongue width as a function of consonant laterality/centrality and as a function of vowel height in front vowel contexts. Multiplanar reconstruction of volumetric data for each token and subject permitted imaging of the static postures on the axial plane. We determine a protocol for identifying a specific oblique axial slice in terms of mid-sagittal anatomical landmarks that allows for a stable and informative index of horizontal tongue width. This index demonstrates utility for identifying and quantifying tongue profile narrowing specific to English lateral consonants. [Work supported by NIH.]

2pSC16. Quantifying labial, palatal, and pharyngeal contributions to third formant lowering in American English /ɹ/. Sarah Harper, Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, skharper@usc.edu), and Shrikanth S. Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Although the acoustic profile of /ɹ/ is largely stable across speakers and contexts in rhotic dialects of American English (e.g., Hagiwara, 1995), significant variability is observed in the articulatory implementation of the palatal, pharyngeal, and labial constriction gestures involved in its production (Delattre & Freeman, 1968; Alwan *et al*., 1999; Tiede *et al*., 2004). In order to quantify each gesture's relative contribution to F3 lowering, we expand upon previous work on articulatory-acoustic relations in American English /ɹ/ by examining the acoustic effect of articulatory variation separately for each supralaryngeal constriction gesture. Real-time MRI data from four speakers in the USC-TIMIT corpus (Narayanan *et al*., 2014) were analyzed to determine the location, length, and aperture for each constriction gesture in 668 tokens of /ɹ/. Acoustic data were taken from simultaneous audio recordings of each speaker's real-time MRI capture, with F1-F4 values extracted at the time of maximal constriction for each gesture. Our results suggest that each gesture's acoustic contribution to F3 depends on between- and within-speaker variation in the articulation of /ɹ/, as variation in the length and location of the palatal and pharyngeal gestures is associated with differences in their relative effect size on F3 lowering. [Work supported by NIH.]

2pSC17. On simultaneous electromagnetic articulography and electroglottography data acquisition. Sarah Harper, Sungbok Lee, Dani Byrd, and Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, skharper@usc.edu)

Simultaneous measurements of speech articulator movement and laryngeal activity are desirable for obtaining a picture of the coarticulatory behavior between oral articulators and laryngeal vocal fold behavior. However, since electroglottography (EGG) use the application of a weak current across the larynx, which is in close proximity to the oral vocal tract, there is a possibility that its use may interfere with the sensor voltage fluctuation measurements used in electromagnetic articulography (EMA). In order to investigate any hypothetical interference effect, data was collected using Northern Digital Instruments' Wave articulograph with Glottal Enterprises's EG2-PCX electroglottograph. Datasets were collected from one male and one female speaker—both with simultaneous EMA and EGG and, for comparison, with EMA alone. Means and standard deviations of inter-sensor distances for static (i.e., reference) and moving (e.g., tongue tip, lip, and jaw) EMA sensor trajectories were compared across the two collection conditions for possible interference effects. Preliminary results find no discernable differences in inter-sensor distances for either static or moving EMA sensors between the with- and without-EGG conditions. If this finding is maintained for additional speakers, NDI EMA data and GE EGG signals may be collected simultaneously without adverse effects on the measurement. [Work supported by NIH.]

2pSC18. The distribution of the retroflex lateral allophone of the liquid phoneme in Korean. Nari Rhee (Linguist, Univ. of Pennsylvania, 2930 Chestnut St., APT2405B, Philadelphia, PA 19104, n.rhee94@gmail.com)

The liquid phoneme in Korean is known to have an optional retroflex allophone in syllable-final positions (Lee 1999). However, there lacks literature evaluating when and how this retroflex allophone is realized. To define the distribution of the retroflex allophone of the liquid phoneme in Korean, this study uses a read speech corpus of Seoul Korean to track the rhotacization of the liquid phoneme in coda positions. Rhotacization of the liquid is detected and measured by the lowering of the third formant. Preliminary results show that the liquid phoneme in coda position is realized as retroflex when preceded by a back vowel, and when followed by another liquid phoneme (gemination). The results suggest that the liquid phoneme is not simply optionally rhotacized in coda positions; instead, the retroflex allophone is realized in contexts defined by the preceding vowel and the following consonant. Future research remains to further examine the articulatory and perceptual aspects of the rhotacization.

2pSC19. Articulatory and acoustic correlates of tongue root contrasts in Gua. Samantha Myers, Kelly Berkson, and Kenneth de Jong (Dept. of Linguistics, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine 844, Bloomington, IN 47405, kberkson@indiana.edu)

Vowel systems in West African languages are often noted for using the position of the tongue root (TR) to contrast vowels throughout the vowel space. E.g. X-ray studies of Igbo show that pairs of vowels such as /i/ and /ɪ/ contrast with regards to tongue root position. A similar study of Akan shows that vowel height also gets incorporated into the contrast (Ladefoged and Maddieson 1990). While many languages are noted for having TR contrasts, imaging data are available for only a small subset. Gua, a Kwa language from the Niger Congo family spoken in coastal Ghana (Simons and Fennig 2017, Yeboah-Obiri 2013), is a critically under-documented language which contains TR contrasts in all high and mid vowels (Advanced TR: /i e o u/; Retracted TR: /ɪ ɛ a ɔ ʊ/). Acoustic analysis and articulatory data from 3D ultrasound recordings reveal that RTR vowels show a variety of deformations of the tongue surface, depending on the vowel. However, these deformations are linked by the mechanics of tongue root retraction. Also, images reveal that differences in tongue height in addition to TR advancement are often present.

2pSC20. Phonetic realizations of the post-consonantal liquid in North and South Korean dialects. Suyeon Yun and Yoonjung Kang (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, Humanities Wing, H427, Toronto, ON M1C 1A4, Canada, suyeon.yun@utoronto.ca)

This study presents large-scale production data of consonant-liquid sequences in North and South Korean dialects. It is known that in South Korean, the liquid /L/ is not allowed in post-consonantal onset position and is nasalized not only after a nasal (e.g., /kɨmLi/ → [kɨmni] "interest") but after a stop, involving the nasalization of the stop (e.g., /sʌpLi/ → [sʌmni] "providence"). We investigate whether this holds (i) for Seoul Korean speakers who become familiar with the onset liquid through exposure to English and (ii) for North Korean speakers who retain the onset liquid in their dialect. Thirty five North Korean defectors speaking Northern Hamkyeong dialect and 20 Seoul Korean speakers read 236 Korean words including a nasal-liquid (/mL, nL, ŋL/) or a stop-liquid sequence (/pL, kL/). Acoustic measurements for the sequences include formant frequencies, high/low frequency energy ratios, and presence of closure and release. Results show that the most common pattern is the nasalized output, with speaker and lexical variation. The post-consonantal /L/ was also realized as the lateral [l] or the tap [ɾ], depending on the speaker's age (old vs. young) and the dialect (north vs. south).

2pSC21. Oral and nasal vowels effects on subglottal pressure. Didier Demolin, Roland Trouville, Ruolan Wang, and Rosario Signorello (Laboratoire de Phonétique et Phonologie, Université Sorbonne Nouvelle, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr)

This paper investigates the effect of oral and nasal vowels on subglottal pressure. The vowels /i, e, a, o, u, ø, œ̃, õ/ were produced on a specific tone by two Belgian French speakers (male and female). The tone frequency was given to the speakers while producing the vowels through a set of headphones connected to a synthesizer. Three frequencies were given at comparable intensities (male speaker: A, C, E; female speaker: C, F, A). Both speakers produced the set of vowels in a mask, with the mouth at a quasi-constant distance from the microphone. In addition to the acoustic signal, the subglottal pressure was recorded by tracheal puncture with a needle inserted between the cricoid cartilage and the first tracheal ring. Measurements show that both speakers produced each of the vowels with stable F0 (2 Hz difference variation from the given tone). One interesting observation is that there is a substantial difference in subglottal pressure between oral and nasal vowels. Both speakers produced nasal vowels with a lower subglottal pressure when compared to oral vowels. Mean differences between both set of vowels were quantified at 2.15 hPa. Nasal vowels were found to have lower intensity than oral vowels.

2pSC22. Cross-contextual consistency of /s/ length and spectral quality in gay men's speech. Dominique A. Bouavichith (Linguist, Univ. of Michigan, 611 Tappan St., #455C, Ann Arbor, MI 48109, dbouavichith@gmail.com)

Previous literature on gay(-sounding) speech has shown that gay men tend to produce longer /s/ tokens with higher spectral centers of gravity (Munson & Babel (2010), Levon (2006, 2007), and Linville (1998)). The existing body of work, however, has only shown this difference using word-initial, non-cluster /s/ tokens or aggregate average values across all contexts. This study specifically investigates these questions in /sC/ clusters, both word-initally and -finally. Participants produced words in a carrier phrase containing either /s/ or an /s/-stop cluster, in either word-initial or word-final position (N = 16). To control for speech rate variation, /s/ tokens were measured with respect to participants' syllable length. Preliminary findings show that gay speakers did in fact lengthen /s/ tokens in all contexts, at similar rates across cluster and singleton contexts. Significant differences between gay and straight men were found for both /s/-to-syllable ratio and spectral center of gravity at /s/ midpoints. These findings contribute to the depth of gay speech acoustics research and provide a basis for context-specific socio-phonetic perceptual studies.

2pSC23. Speaking of sexuality: Analyzing [s] as an index of speaker identity in Japanese. Ryan C. Redmond (Linguist, Univ. of California Davis, 2505 5th St., Apt. 117, Davis, CA 95618, rcredmond@ucdavis.edu)

While variation in the production of the voiceless alveolar fricative [s] has been proposed as a marker of alternative sexualities in some languages, this phenomenon has yet to be tested in Japanese. In the present study, data from Japanese television shows and self-uploaded online "coming out" videos are used to sample 600 tokens of word-initial/intervocalic [s] from 20 different speakers: 10 queer men (older camp-gay "onee" celebrities, and younger Youtubers), and 10 men and women who do not identify as gay. Following a moments analysis, spectral center of gravity and skew measurements arose as the clearest delineating variables (i.e., these factors contributed to significant variation in [s] production) between the queer vs. non-queer male groups, as well as the non-queer male vs. non-queer female groups, but not the queer male vs. non-queer female groups. These findings may suggest alignment between Japanese women's language and Japanese "gay" speech, as has been noted in previous research, but two counter theories are posed. One theory notes the social benefits associated with hyperarticulation that could arise from the usage of these variant phones, while the other theory suggests that non-queer men could be the ones deviating from the norm.

2pSC24. Dialect contact and word-specific phonetics: North Koreans in Seoul. Yoonjnung Kang and Suyeon Yun (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, HW427, Toronto, ON M1C 1A4, Canada, yoonjung.kang@utoronto.ca)

This study examines the speech of North Korean refugees who currently reside in Seoul. We investigate (a) how the North Korean speakers' stops are affected by contact with Seoul Korean and (b) whether words only commonly used in the North and words newly acquired in the South are realized differently. The data were collected from 35 Hamkyeoung speakers, balanced for age (born before vs. after 1975) and time since arrival in Seoul (0–3 vs. 3–15 years). Twenty Seoul speakers also participated as a comparison group. Participants produced 78 stop-initial words with two repetitions, evenly divided into North vs. South lists, balanced for laryngeal feature, place of articulation, and the following vowel height across the lists. The results show that in line with previous literature, Hamkyeoung aspirated stops are longer in VOT than Seoul stops and Hamkyeoung lenis stops are shorter in VOT than Seoul stops. We found evidence for contact-induced phonetic drift for aspirated stops, but not for lenis stops—earlier arrivals produce aspirated stops with a shorter VOT than recent arrivals. We found no evidence of word-specific phonetic representation and stops were produced with a comparable VOT value regardless of their Northern vs. Southern origin.

2pSC25. A regression approach to vowel normalization for missing and unbalanced data. Santiago Barreda (Dept. of Linguist, UC Davis, Davis, CA 95616, sbarreda@ucdavis.edu) and Terrance M. Nearey (Linguist, Univ. of AB, Edmonton, AB, Canada)

Researchers investigating the vocalic systems of languages or dialects frequently employ normalization methods to minimize between-speaker variability in formant patterns while largely preserving dialectal, between-phoneme variability. One popular method, log-mean normalization, relies on estimating the logarithm of the geometric mean formant-frequency (G_s) produced by a speaker across their vocalic inventory, and then expressing the formant patterns produced by a speaker as deviations from this mean. However, in the face of missing or unbalanced data, the traditional approach to calculating G_s for a speaker will usually lead to biased estimates, which will produce artificial asymmetries in the normalized vowel spaces of different speakers. An alternative method is proposed for the estimation of G_s based on a linear-regression framework, which avoids the biases associated with traditional estimation of G_s when data is unbalanced. The regression method to normalization is described, and simulations are carried out to compare the accuracy of G_s estimates via regression to other estimation methods. Results indicate that the proposed method is substantially more accurate than the traditional approach to estimating G_s in the face of missing or unbalanced data.

2pSC26. Allocation of attention to real-time visual speech feedback in a digital mirror. Elizabeth D. Casserly and David E. Ballenger (Dept. of Psych., Trinity College, 300 Summit St., Hartford, CT 06106, elizabeth.casserly@trincoll.edu)

The role of sensory feedback in speech motor control is typically investigated by observing the behavioral and/or neurological response to experimental feedback perturbation. The function of feedback and its use in typical control is therefore inferred from responses under atypical conditions. The present study avoided perturbation, using eye tracking and a digital mirror (continuous video relay from a webcam) to examine how speakers attend to real-time visual feedback across normal speech and non-speech tasks. Fixation locations and durations were recorded during a period of rest prior to task onset (no-task condition), a recitation of the ABC's, a re-telling of a popular children's story (Goldilocks and the Three Bears), and parallel clear speech versions of the latter two tasks. Analysis of gaze showed that participants overall seemed to avoid fixating on their self-image, and that avoidance increased during speech tasks compared to non-speech "rest." Across the two content areas (ABC's vs. storytelling), participants' self-gaze was concentrated more heavily on speech-relevant areas during storytelling, and no differences in gaze were observed between casual and clear speech. These data suggest an increased role for sensory feedback during complex linguistic tasks, as well as indicating an overall aversion towards realistic visual self-feedback.

2pSC27. Variable vowel convergence in a novel cooperative task. Jennifer Nycz and Shannon Mooney (Dept. of Linguist, Georgetown Univ., 1437 37th St. NW, Washington, DC 20057, jn621@georgetown.edu)

Phonetic convergence is partly automatic, yet mediated by linguistic and attitudinal factors; salient social identity can suppress convergence or lead to divergence (Babel 2010). We assessed convergence across vowels among 12 pairs of speakers using a novel task that minimizes the salience of identity. Each person separately read aloud a 45-item word list, once before the main task and again afterwards. Between readings, the pair played a version of the game Taboo, in which players take turns attempting to elicit specific words from their partner while avoiding forbidden words. Lobanov-normalized formant values were extracted from the word lists. A convergence measure was calculated for each word for each pair, by subtracting the Euclidean distance between the speakers' vowels in that word in post-game lists from that of the pre-game lists. Mixed-effects models of convergence were fit, with random effects for word and pair and fixed effects for vowel and pre-game distance. Greater initial distance was associated with greater convergence. /i/ and /a/ converged most consistently, while diphthongs typically diverged. This suggests that listeners focus on point vowels to model an interlocutor's vowel space, facilitating convergence in these vowels, and that even when identity is not salient, divergence may occur.

2pSC28. Exploring the source of short-term variations in respiratory data. Susanne Fuchs (Leibniz Ctr. for General Linguist, Berlin, Germany) and Laura L. Koenig (Adelphi Univ. and Haskins Labs., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu)

Past studies of speech breathing have observed short-term variations in the respiratory signal. In early kinematic work, Stetson observed "ripples" on the breathing signals, which he interpreted to represent chest pulses for each syllable. Ladefoged and coauthors subsequently reported that brief excursions in respiratory data corresponded closely in time to stressed syllables. Segmental characteristics may also impact respiratory signals, however. In particular, voiceless obstruents have been associated with short-term decreases in breathing data, presumably reflecting rapid airflow venting through an open glottis. Indeed, much of Stetson's speech material consisted of repeated CV syllables with voiceless stop onsets. This study revisits the degree to which voiceless consonants correspond to negative excursions in respiratory signals obtained from several speakers. We collected acoustic data as well as intraoral pressure (to infer subglottal pressure non-invasively) and used inductance plethysmography to obtain displacements of the rib cage and abdomen. We use acoustic and intraoral pressure signals to locate voiceless segments as well as nasal consonants in stressed syllables, and assess the characteristics of the respiratory signals in these regions. The nasals provide a control condition for testing whether voiceless sounds have unique effects on the respiratory data.

2pSC29. Vowel contrasts relative to schwa: Effects of utterance-level fundamental frequency. Christina Kuo (Commun. Sci. and Disord., James Madison Univ., MSC4304, 235 Martin Luther King Jr. Way, Harrisonburg, VA 22807, kuocx@jmu.edu)

The purpose of this study is to characterize the potential effects of utterance-level fundamental frequency (F0) on the acoustic contrasts of vowels as expressed relative to the mid-central vowel schwa. This is a follow-up to a previous study [Kuo, *J. Acoust. Soc. Am.* **141**, 3840 (2017)] based on a hypothesis of schwa as a speaker-defined reference for vowel contrasts. The hypothesized reference schwa is made up of the averaged first and second formant (F1 and F2) frequencies from many tokens of schwa produced by a given speaker. Motivated by the potential interactions among F0, spectral sampling, and articulation, the present study evaluates the effects, if any, of utterance-level F0 on vowel contrasts relative to schwa. Utterance-level F0 is measured for breath groups at the sentence level. Vowel contrasts are expressed as the Euclidean distances between vowels and the reference schwa in the F1-F2 space. Specifically, two questions are of interest. First, are the Euclidean distances between vowels and the reference schwa impacted by the utterance-level F0? Second, does utterance-level F0 affect the variability of schwa productions, and thus influencing the makeup of the reference schwa? Findings will be discussed within the framework of the acoustic theory of speech production.

2pSC30. Pause postures in American English. Jelena Krivokapic, Will Styler (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220, jelenak@umich.edu), Ben Parrell (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, Newark, DE), and Jiseung Kim (Linguist, Univ. of Michigan, Ann Arbor, MI)

Recent work examining articulation during pauses has found that articulatory patterns distinguish grammatical from ungrammatical pauses (Ramanarayanan *et al.* 2009). For Greek, Katsika *et al.* (2014) have identified pause postures (specific configurations of the vocal tract at prosodic boundaries) which are triggered by π-gestures with high activation levels and consequently occur at strong prosodic boundaries. This study investigates pause postures in American English, specifically, whether they occur, and how they are coordinated with other events at prosodic boundaries. In an electromagnetic articulometry (EMA) study, seven speakers produced 7-8 repetitions of 42 types of sentences varying in linguistic structure (stress, boundary, phrasing, and sentence type). Results from two speakers analyzed to date indicate that pause postures exist but that their presence might be speaker dependent. Analyses of gesture lags indicate a stable relationship between the boundary tone and pause posture, that the boundary tone gesture starts later when stress is on the second syllable, and that the π-gesture is shifted towards the stressed syllable, parallel to findings in Katsika *et al.* (2014). We discuss these results in relation to models of prosodic structure and to speech planning processes. [Work supported by NIH and NSF.]

2pSC31. Tracking developmental changes in articulatory strategy during childhood. Tanner Sorensen, Asterios Toutios, Louis Goldstein, and Shrikanth S. Narayanan (Univ. of Southern California, 3740 Mcclintock Ave., Los Angeles, CA 90089, tannersorensen9991@gmail.com)

During development, children learn how to coordinate movements of the speech articulators in order to optimally achieve motor goals. It has been shown that variability in these coordinative patterns, or articulatory strategies, decreases over the course of childhood before ultimately stabilizing at adult-like levels. For example, the jaw becomes more tightly coordinated with the tongue and lips. Recent advances in real-time magnetic resonance imaging (rt-MRI) and analysis provide a means to characterize such articulatory strategies by quantifying how much the jaw, tongue, lips, velum, and pharynx contribute to constrictions of the vocal tract during speech. The articulators are segmented in reconstructed rt-MRI and constriction degrees are measured as the linear distance between opposing structures (e.g., tongue and palate). Change in constriction degree over time is decomposed into articulator contributions to characterize articulatory strategy. In this pilot study, we obtain quantitative biomarkers of articulatory strategies from a 10-year-old participant and compare them against those of 8 healthy adult participants. The study quantifies the difference between child and adult articulatory strategies in terms of how much each articulator contributes to constrictions of the vocal tract during speech and indicates how the articulator movements are coordinated with each other in time.

2pSC32. Production of stop consonants in 3-to-6-year-old Mandarin-speaking children. Jing Yang (Commun. Sci. and Disord., Univ. of Central Arkansas, CSD/ Speech Lang. Hearing Ctr., 201 Donaghey Ave., UCA Box 4985, Conway, AR 72035, jyang@uca.edu)

The present study compared the temporal measurements of stop consonants in 29 3-to-6-year-old Mandarin-speaking children and 12 Mandarin-speaking adults. Each participant produced 18 Mandarin disyllabic words which contained six stops /p, p^h, t, t^h, k, k^h/ each followed by three vowels / a, i, u/, respectively. All stop consonants were located at the word-initial position in the first syllable. The temporal measurements of VOT, overall burst duration, average duration per burst, the number of burst and VOT-lag duration were obtained. The results showed that Mandarin-speaking children in this age range produced all six stops, especially aspirated stops, with longer VOT means than the adults. The older children produced even longer VOTs than the younger children, which evidenced the overshooting pattern of VOT values in children. Although adult-like short-lag VOTs for unaspirated stops were achieved in all children, the long-lag VOTs for aspirated stops were widespread in the younger group and gradually developed to a concentrated distribution in the older children. Further examination of the burst and VOT-lag revealed that these children tended to produce shorter average duration per burst and longer VOT-lag than the adults for both unaspirated and aspirated stops. These results indicate that children in this age range may not have developed adult-like laryngeal-oral timing pattern and airflow control for stop production.

2pSC33. Practicing English medical word pronunciation with a communication robot and a tablet by Japanese medical students. Yoko Sakamoto and Nobuhiro Sakata (Premedical Sci., Dokkyo Medical Univ., 880 Kitakobayashi, Mibu, Shimotsugagun, Tochigi 3210293, Japan, y-saka@dokkyomed.ac.jp)

The aim of this study was to investigate which tool can be more effective way to practice English medical word pronunciation for Japanese medical students: a communication robot or a tablet. The subjects were 8 Japanese medical freshman students. The target words were 10 medical words related to an infectious disease. In the training, the subjects learned the meaning of the word in Japanese and then they practiced the pronunciation in English. Four subjects followed the order; Pretest → Robot → Posttest1 → Tablet → Posttest 2, while the remaining 4 subjects were counterbalanced; Pretest → Tablet → Posttest 1 → Robot → Posttest 2. All subjects answered the questionnaire after Posttest 2. The process was video-recorded and the waveform was analyzed. The results showed that the

postures and voices of the subjects were better with a communication robot than with a tablet. The pronunciation practiced with a communication robot had improved with a fewer trial. The subjects wrote in the questionnaire that they could practice pronunciation with a communication robot aiming to be able to talk with a person in the future. Thus, compared to a tablet, a communication robot can be an effective tool to practice English medical words.

2pSC34. Tongue position as an articulatory property of voicing in Brazilian Portuguese and Thai. Suzy Ahn (Dept. of Linguist., New York Univ., 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu)

Articulatory adjustments are often necessary to ensure that closure voicing will be present in stops (Rothenberg 1968). One common adjustment is to enlarge the supralaryngeal cavity volume via tongue root advancement (Westbury 1983). This study uses ultrasound to examine tongue positioning during Brazilian Portuguese and Thai stops. Portuguese has a two-way laryngeal contrast: voiced and voiceless (unaspirated), and Thai has a three-way contrast: voiced, voiceless unaspirated, and voiceless aspirated. Eight native speakers of each language recorded phrase-initial stops followed by /a/. Results show a clear distinction in tongue position between voiced and voiceless unaspirated stops in both languages. Tongue root is more advanced for voiced compared to voiceless unaspirated in alveolar and velar stops. For labial stops, Thai speakers lower the tongue front (which is another cavity enlargement maneuver) whereas Portuguese speakers advance their tongue root for voiced stops. The results suggest that speakers of both languages employ tongue positioning for voicing during closure. How the tongue position is operationalized may be language-specific as long as voiced stops have a larger cavity volume. On the other hand, the role of tongue position for aspiration is less clear since in Thai, alveolar stops and velar stops show the opposite pattern when voiceless unaspirated and aspirated stops are compared.

2pSC35. Nasalization as a correlate of word-final voicing in English. Suzy Ahn (Dept. of Linguist., New York Univ, 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu) and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

Maintenance of laryngeal voicing during closure of word-final stops is believed to be articulatory challenging (Westbury & Keating, 1986). Additional articulatory maneuvers may help sustain voicing during final closure, and one such maneuver is pre-nasalization (Rothenberg, 1968). It has been demonstrated that pre-nasalization is associated with voicing in utterance-initial stops in French, Spanish, and to a lesser extent, in English where voiced stops are often realized without voicing during closure (Solé, 2011). Laryngeal contrast is typically maintained between word-final voiced and voiceless stops in English, and voicing is often found during closure. Therefore, articulatory maneuvers supporting the phonetic realization of this contrast are particularly important in final position. The proposed study examines the degree of nasalization in the vowel immediately preceding word-final stops in monosyllabic real words of American English ($N = 18$, monolingual speakers of Mid-western dialect). Preliminary examination of word-final stops in English indicates that pre-nasalization occurs for word-final voiced stops as well. Moreover, the presence of nasalization is predicted to correlate with the amount/duration of voicing during the stop closure of final consonant. The results will contribute to our understanding of the phonetics of word-final laryngeal contrast and potentially expand the inventory of secondary perceptual cues to word-final voicing.

2pSC36. Onset f0 as a correlate of voicing in Marathi. Olga Dmitrieva (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu) and Indranil Dutta (The English and Foreign Lang. Univ., Hyderabad, India)

Onset f0 (the fundamental frequency at the onset of vowel immediately following the consonant) has been shown to correlate with voicing in a variety of languages with two-way voicing contrasts in stop consonants, such as the voice-type contrast (between prevoiced and voiceless unaspirated stops) or the aspiration contrast (between voiceless unaspirated and voiceless aspirated stops). Onset f0 is typically higher after phonologically voiceless than after voiced stops. With some notable exceptions, onset f0 data from languages with three-way and four-way voicing contrasts is relatively scarce. The present study explores onset f0 as a correlate of voicing in Marathi, an Indo-Aryan language with a four-way voicing contrast which includes both aspirated and unaspirated voiced and voiceless stops. Results demonstrate that onset f0 covaries with both laryngeal voicing and aspiration in Marathi: f0 is significantly lower for both sets of voiced than voiceless stops. Onset f0 is also significantly lower for both sets of aspirated than unaspirated stops but the effect of voicing is greater in magnitude and more consistent across individuals than the effect of aspiration. Combined with previous findings, the data suggest that onset f0 covariation with laryngeal voicing is more universal across languages than its covariation with aspiration.

2pSC37. Characteristics of saliva swallowing in the reading of prepared texts. Kuniko Kakita (Toyama Prefectural Univ., 5180 Kurokawa, Imizu, Toyama 939-0398, Japan, kakita@pu-toyama.ac.jp) and Shizuo Hiki (Waseda Univ., Tokorozawa, Japan)

This study examines the nature of saliva swallowing during speech production. Speakers read two contrasting types of texts, a "paragraph" and a "list." The "paragraph" was a Japanese version of "The North Wind and the Sun," consisting of four sub-paragraphs. The "list" was a simple repetition of quasi-identical short sentences, without any paragraph construction. Simultaneous recordings of speech sounds (via microphone), swallow sounds (via laryngeal contact microphone), and respiratory movements of the thorax and abdomen (via respiratory belt transducers) were obtained and analyzed, especially in relation to sentence-final pauses. Preliminary results indicate that (1) swallowing tends to occur roughly in the middle of a pause and is followed by inspiration for the upcoming utterance, (2) inspiration is generally deeper when preceded by swallowing, (3) swallowing adds approximately one second to a pause, and (4) in paragraph reading, swallowing is more likely to occur during a pause between two sub-paragraphs, i.e., at a topic transition, whereas in list reading, swallowing reflects speakers' physiological needs more directly.

2pSC38. Novel imaging tools for supporting the teaching of singing and spoken performance. Reed Blaylock and Shrikanth S. Narayanan (Univ. of Southern California, 1150 W 29th St. Apt. 4, Los Angeles, CA 90007, reed.blaylock@gmail.com)

Recently, singers and singing instructors have begun to use real-time magnetic resonance imaging (rtMRI) videos of speech and singing movements as a pedagogical tool. Singers use videos of vocal tract movements to learn about the movements of speech, and are then able to contrast those movements with their ideal singing productions. These validated images can now replace the previous "practice-of-speculation" regarding the articulatory postures desirable during singing. While rtMR videos are most often used by professional instructors and aspiring-professional singers, these videos may soon commonly be used to assist amateur choruses as well, which often struggle to produce unified vowel qualities. Beatboxing is a related and unexplored pedagogical domain, yet beatboxing was one of the earliest spoken performance styles acquired at USC with real-time MRI. These videos, together with USC's publicly available inventory of rtMR videos demonstrating the sounds of the world's languages, provide a rich resource for the extreme and varied percussive sounds used by beatboxers. In sum, students of singing and spoken language performance have a brand new asset available to them in the form of dynamic real-time MR videos of the moving vocal tract. [NIH R01DC007124.]

Session 2pSP

Signal Processing in Acoustics and Underwater Acoustics: Detection, Classification, Localization, and Tracking (DCLT) using Acoustics (and Perhaps Other Sensing Modalities) II

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Contributed Papers

1:00

2pSP1. Simulation of passive source localization in near-Arctic conditions using frequency-difference matched field processing. David J. Geroski (Appl. Phys., Univ. of Michigan – Ann Arbor, Randall Lab., 450 Church St., Ann Arbor, MI 48109, geroskdj@umich.edu) and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, Ann Arbor, MI)

Matched Field Processing (MFP) is a well-known technique for passive source localization in complex acoustic environments. It involves correlating array-recorded acoustic fields with replica fields calculated using a model of the acoustic environment, and works well is known environments. However, acoustic signals often depend on unknown environmental details which are not included in the replica-field calculations. The severity of this mismatch increases with frequency and source-array range, and it can cause MFP to fail in the signal band at relevant ranges in the arctic ocean. A proposed remedy to this problem is to utilize the frequency-difference autoproduct of the measured acoustic field instead of the acoustic field itself, and to perform the replica calculations at the difference frequency. The simulated performance of frequency difference MFP is shown for a generic near-Arctic environment at signal frequencies of 200 to 300 Hz, and ranges of 30 to 300 km. Here, mismatch between in the acoustic and replica field is modeled by imposing a range-dependent altimetry and surface sound speed profile to simulate surface ice, and by random time delays applied to each ray-path between the source and each receiver to simulate refractive index fluctuations in the ocean. [Sponsored by ONR.]

1:15

2pSP2. Passive acoustic localization and tracking using synthetic data for the Northern Gulf of Mexico. Britt J. Aguda, Kirk D. Bienvenu (Physics, Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, bjaguda1@uno.edu), Bradley J. Sciacca (Physics, Univ. of New Orleans, Harvey, LA), Joshua Veillon, SydniCherise O. Austin, and Juliette W. Ioup (Physics, Univ. of New Orleans, New Orleans, LA)

Passive acoustic localization and tracking of marine mammals shows potential for future acoustic monitoring efforts. The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015 using Environmental Acoustic Recording Systems (EARS) buoys, returning to sites previously surveyed by LADC. The localization and tracking method developed for the EARS hydrophones uses Monte-Carlo based simulations based on this data. Synthetic whale clicks at random times along a sinusoidal path were propagated to the EARS receivers and the location of the synthetic source was tracked. Several whale clicks in a series determine the most probable path of the animal. The localization/tracking method and relative localization are adaptable for variables such as site location, mooring locations, number of moorings, depth, etc. Localization of two synthetic moving whales in three-dimensional space will be demonstrated. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

1:30

2pSP3. Future deployment corrections for passive acoustic localization and tracking using real data for the Northern Gulf of Mexico. Kirk D. Bienvenu, Britt J. Aguda (Physics, Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, kdbienv1@uno.edu), Bradley J. Sciacca (Physics, Univ. of New Orleans, Harvey, LA), Joshua Veillon, SydniCherise O. Austin, and Juliette W. Ioup (Physics, Univ. of New Orleans, New Orleans, LA)

The Littoral Acoustic Demonstration Center–Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015 using Environmental Acoustic Recording Systems (EARS) buoys. A localization and tracking method developed for the EARS hydrophones uses Monte-Carlo based simulations based on these data but was only tested using synthetic whale clicks at random times along a sinusoidal path. The location of the synthetic source was tracked. Real data obtained in 2015 could not be used by the method developed due to excessive clock drift between moorings with no way to correct the phase differences. It will be shown, using the geometry of the site and acoustic theory, how a simple, low powered, periodic acoustic ping can aid in correcting the phase difference of the clocks after significant time has passed. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

1:45

2pSP4. Spatial-division-multiplexing in time-variant and multi-path channel based on vector adaptive time reversal technique. Dian Lu, Xueli Sheng, Yali Shi, Hanjun Yu, and Longxiang Guo (Underwater Acoust. Eng. College, Harbin Eng. Univ., No.145, Nantong Rd., Nangang District, Harbin, Heilongjiang Province 150001, China, bryce_era@sina.com)

In the field of multistatic remote detection for underwater target, as a time-variant and multi-path channel interfered by environmental noise, complex underwater acoustic environment limits the spatial share of channel and poses a great challenge against spatial-division-multiplexing (SDM) in multistatic detection. In this paper, a kind of method of spatial-division-multiplexing(SDM) based on vector adaptive time-reversal technique is

proposed. By use of time-reversal technique, multi-path structure of channel is efficiently restrained; Meanwhile, with adaptive filtering applied, time-variant characteristic of channel is effectively suppressed. Moreover, with single vector hydrophone, spatial focus of target echoes and noise interference suppression is accomplished by spatial filtering. Finally, excellent results are performed in SDM of multistatic detection, supported by the experimental simulation.

2:00

2pSP5. An improved target detection and azimuth angle estimation method using a single acoustic vector sensor. Lin Ma, Anbang Zhao (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, HLJ 150001, China, malin@hrbeu.edu.cn), Juan Hui, Caigao Zeng, and Xuejie Bi (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

In this paper, an improved underwater acoustic target detection and azimuth angle estimation method using a single acoustic vector sensor (AVS) is proposed based on time-reversal and convolution theory. The proposed method can be applied in the active and the passive sonar detection system. According to the conventional detection and estimation method based on complex acoustic intensity measurement, the mathematical and physical model of this proposed method is described in detail. Computer modelling and simulation is applied to demonstrate the proposed method's effectiveness. In order to further verify the practical application performance of the proposed method, the research group carried out the open lake experiments. The computer simulation and open lake experiments results indicate that this method can realize the azimuth angle estimation with high precision by using only a single AVS. Compared with these conventional methods, the proposed method achieves better detection and estimation performance.

2:15

2pSP6. Detection performance analysis of product processing of colinear arrays. Kaushallya Adhikari (Louisiana Tech Univ., 600 Dan Reneau Dr., Ruston, LA 71270, kaushallyaadhikari@gmail.com) and John R. Buck (Univ. of Massachusetts Dartmouth, North Dartmouth, MA)

Non-uniform linear arrays (NULAs) often achieve better resolution than standard uniform linear arrays (ULAs) with equal numbers of sensors. However, conventional beamforming (CBF) of an NULA received signal leads to the same detection statistic PDF as a ULA with the equal number of sensors, undercutting NULA's improved resolution. Nested and coprime arrays partition an NULA into two subarrays and multiply the subarrays' CBF outputs. This research compares this product processor's detection performance against a CBF detector with an equal number of sensors for a narrowband Gaussian signal in spatially white additive Gaussian noise. The product processor's detection PDF is a scaled product of the detection statistic with modified Bessel functions. Receiver operation characteristics (ROC) curves illustrate that the product processor's performance is inferior to the CBF detector with an equal number of sensors. The detection performance of a product processor matches the CBF detector only for high SNRs and large numbers of sensors. However, in the presence of interferers, the product processor for coprime arrays can outperform both the CBF detector and product processing nested arrays with an equal number of sensors. [Work supported by ONR.]

2:30

2pSP7. Double-difference tracking of bowhead whales using unsynchronized directional acoustic recorders in the Beaufort Sea. Ludovic Tenorio-Hallé (Scripps Inst. of Oceanogr., 1044 Loring St., San Diego, CA 92109, ludovictenorio@gmail.com), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), and Katherine H. Kim (Greeneridge Sci., Inc., Santa Barbara, CA)

Passive acoustic monitoring has become a standard method for detecting bowhead whale (*Balaena mysticetus*) activity in Arctic waters. Between 2007 and 2014, over 40 directional acoustic recorders, known as DASARs, were deployed in the Beaufort Sea during the bowhead whale migration season. Individual DASARs can estimate azimuth, allowing calls to be localized by triangulation using multiple DASARs. However, these bearings are subject to calibration biases, and individual recorders were not precisely time-synchronized, making relative time-of-arrival information unusable for standard localization purposes. Double-difference methods have previously been applied in seismology to obtain high-precision relative positions of earthquakes by measuring changes in relative travel-times between multiple events over widely distributed seismic sensors. Here, the double-difference method is applied to detected bowhead whale calls in order to improve the relative localization resolution. The approach uses changes in both relative call travel-times and bearings, detected at multiple DASARs, to determine high-precision relative locations of these calls despite the presence of systemic timing and bearing errors in the measurements. The resulting positions allow tracking of individual whales, which may provide insight into the function of these calls.

2:45

2pSP8. Research on passive detection parameter measurement technology based on single vector sensor. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn)

Passive detection as a major function of the sonar system, over the years has been the work of waterborne workers to solve the problem. This paper mainly talks about the measurement technology of passive detection parameters based on single vector sensor. The problem of low frequency line spectrum signal extracted from ship radiated noise is studied, and the azimuth sequence and frequency sequence are obtained by estimating and analyzing the low frequency line spectrum signal. Ship radiated noise contains a lot of target information, so the extraction of the line noise of the ship's radiated noise is of great significance in the measurement of passive detection parameters. The radiation noise of the ship is a broadband spectrum modulated by the spectral signal envelope. The average azimuth and complex intensifier are used to estimate and analyze the target azimuth, and the estimated angle is statistically processed by histogram and weighted histogram. For the frequency estimation technique, the adaptive frequency estimator based on the adaptive notch filter and the adaptive LMS (Least Mean Square) algorithm is chosen to study the target signal frequency. Through the 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.

3:00–3:15 Break

3:15

2pSP9. Acoustic localization of distributed coherent and incoherent sources using SEMWAN with subarray smoothing. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

Accurate estimation of the location and level of remote acoustic sources from recorded acoustic signals is attractive in many applications. In wind tunnel tests, where noise sources are commonly distributed and incoherent, high resolution array signal processing techniques like the spectral estimation method with additive noise (SEMWAN) have been useful for source localization when background noise measurements are available. However, for many continuously distributed systems, such as a simple vibrating plate, the assumption of incoherent sources is incorrect, and techniques like SEMWAN may yield spurious results. In this presentation, results are reported for the use of SEMWAN alongside a subarray smoothing technique to formulate the coherent source localization problem as an incoherent source localization problem. Simulations comparing localization performance for distributed coherent and incoherent sources are shown. Results from a proof-of-concept experiment using multiple sources and a 15-element linear receiver array are also evaluated against simulation. Performance comparisons are made between SEMWAN, MUSIC, and conventional beamforming techniques in addition to showing the effects of subarray smoothing. [Sponsored by NAVSEA through the NEEC and by the U.S. DoD through an NDSEG Fellowship.]

3:30

2pSP10. Experimental investigation of the unit circle minimum variance distortionless response adaptive beamformer. Matthew Tidwell and John R. Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, mtidwell@umassd.edu)

Source masking occurs when loud interferers produce high side lobes in the conventional beamformer (CBF) scanned response, obscuring the true direction-of-arrival of the desired signal. The minimum variance distortionless response (MVDR) adaptive beamformer (ABF) places deep beampattern notches near interferer directions to suppress their power in the ABF output while maintaining unity gain in the look direction. In practice, the sample matrix inversion (SMI) MVDR ABF replaces the ensemble covariance matrix with the sample covariance matrix (SCM) when computing the array weights. Estimation of the SCM in snapshot-limited and snapshot-deficient scenarios causes perturbations of the array polynomial zeros from the unit circle. The unit circle (UC) MVDR ABF projects the array polynomial zeros radially back to the unit circle, producing deeper notches and reduced sidelobe levels [Tuladhar and Buck, ICASSP 2015]. Preliminary experiments in an outdoor, free-field environment with a 21-microphone uniform linear array at a design frequency of 2.3 kHz found the UC MVDR ABF consistently outperformed both the conventional and SMI MVDR beamformers in interferer suppression when limited to 21 snapshots (one snapshot/sensor). Additionally, the UC MVDR beamformer averaged 18 dB better white noise gain than the SMI beamformer across six independent trials. [Research funded by ONR.]

3:45

2pSP11. Automated passive acoustic detection, classification, localization, and tracking methods applied to recorded data from the Pacific Missile Range Facility, Kauai, Hawaii. Stephen W. Martin (Environ. Program, National Marine Mammal Foundation, 1929 Cherrywood St., Vista, CA 92081, martin.steve.w@gmail.com), Brian Matsuyama (Environ. Program, National Marine Mammal Foundation, San Diego, CA), Tyler A. Helble (Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA), Cameron R. Martin (Environ. Program, National Marine Mammal Foundation, San Diego, CA), E. E. Henderson (Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA), and Gabriela C. Alongi (Environ. Program, National Marine Mammal Foundation, San Diego, CA)

Automated methods will be reviewed for performing passive acoustic detection, classification, localization, and tracking of some marine mammal species and man-made sources. The methods have been applied to recorded hydrophone data from a large aperture seafloor array at the Pacific Missile Range Facility (PMRF) at Kauai, Hawaii, with some of the methods currently implemented in a real-time system at PMRF. The process consists of custom software both in C++ and Matlab in a 3 or 4-step process. Automated detections of various sounds in specific frequency bands are first performed. In some cases, a classification stage is also performed. The third stage involves model-based localization of the detections or classifications. The fourth stage converts the localizations into individual source tracks. Source tracks are currently generated for fin, sei, Bryde's, humpback, and sperm whales and mid-frequency active sonar (MFAS) transmissions. Performing this process on marine mammals allows information regarding the movement patterns of the whales while calling, as well as information on the species' calls (e.g., call rate, frequencies, durations, estimated source levels). By performing similar processes on man-made sources, it is possible to determine some marine mammal responses from proximity of Navy vessels and mid-frequency active sonar sources.

4:00

2pSP12. Detection and localization of weak impulses and continuous sources within the urban acoustic environment. Richard Haskins, Christopher Simpson, and Mihan H. McKenna (usace, ERDC, 3909 Halls Ferry Rd., Vicksburg, MS 39180, richard.w.haskins@usace.army.mil)

Signal detection and localization with large aperture acoustic arrays in urban environments can be inherently difficult. This poster presents some preliminary approaches under investigation for; determining the direction of arrival of continuous sources that are changing in frequency, detecting and localizing low signal to noise ratio impulses in the presence of coherent background noise, and a fast mapping approach for converting array element delays to direction of arrival using a neural network fitting function. For uniform circular array geometries with a central reference array element, the direction of arrival of sources with changing frequency can be visualized by applying the Hilbert transform and analyzing the relative phase angle rate of change of the outer array elements. Next, various computational approaches for low level impulse detection in the presence of coherent acoustical noise are presented. Last, a neural network fitting function is discussed that performs array specific delay mapping to direction of arrival with high computational efficiency and trained noise immunity.

4:15

2pSP13. Small aperture acoustic methods to locate transients. Michael V. Scanlon (US Army Res. Lab., Acoust. & EM Sensing Branch, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Acoustic microphone arrays have been used in battlefield environments to detect and locate continuous wave targets like helicopters and ground vehicles, and for transient events like gunshots, mortars, rockets and explosions. Depending on the application, source bandwidth, propagation distance, and required localization accuracy, these arrays can vary their sensor separation greatly. Unattended ground sensors, manned- and unmanned-platforms, and Soldier-worn systems can all benefit from very small apertures if performance can be maintained. A commercially available acoustic particle velocity sensor, approximately one-half an inch in diameter, showed good transient localization at a field experiment. Data will also be presented from a small cluster-array of cardioid microphones.

4:30

2pSP14. Evaluating direction of arrival uncertainty in undersea canyons with internal tides. Timothy F. Duda (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Bruce Cornuelle (Scripps Inst. of Oceanogr., La Jolla, CA), Ying-Tsong Lin, Arthur Newhall, and Weifeng G. Zhang (Woods Hole Oceanographic Inst., Woods Hole, MA)

The slopes of undersea canyon regions impart complexity to underwater acoustic fields for two reasons: intricate patterns of reflection from the seabed, and sound-speed anomalies from internal tides generated at the slopes. Both may spread the horizontal directional spectrum of sound from "line of sight" to a source. The directional spectrum is tied to the covariance matrix of the field, a fundamental quantity that can be measured or modeled. Here, sound-field horizontal-lag spatial covariance matrices and other derived quantities are generated from time-stepped 3D parabolic equation acoustic simulations made using sound-speed fields from ocean models. The covariance matrices are then inserted into the direction-of-arrival (DOA) estimation problem. Analysis of DOA estimates and error bounds is done for a conventional beamformer and for a Gauss-Markov inverse-based beamformer for a variety of signal-to-noise ratios. The method treats non-line of sight acoustic energy as a form of noise. At low signal-to-noise ratio, the Gauss-Markov estimator can perform better that the other. This analysis of field variability allows performance degradation caused by evolving ocean structures to be directly compared to other detrimental influences such as excess noise and array deformation. Detection, localization, and tracking are affected by the processes examined here.

4:45

2pSP15. Extrapolated open spherical microphone arrays beamforming for acoustic source localization. Boquan Yang, Shengguo Shi, Ying Li, Lanyue Zhang, and Chao Wang (Harbin Eng. Univ., NanTong Str NanGang Dist., Harbin, HeiLongJiang 150001, China, ybq416751946@163.com)

Spherical microphone arrays have become a particular tool for analyzing the spatial sound field, especially in source localization and identification. Spherical Harmonic Beamforming (SHB) is a fundamental algorithm with the spherical microphone arrays in aeroacoustics. However it presents some intrinsic limitations, specific poor resolution and sidelobe suppression capability. This paper aims to overcoming this limitations by employing Functional Beamforming and Beamforming Regularization Matrix. Functional beamforming has a narrower the mainlobe width and lower sidelobes by

increasing the order of matrix functions. To improve the performance of algorithm at low frequency, this paper uses an extrapolated spherical arrays that is a larger radius of the spherical arrays through the Beamforming Regularization Matrix algorithm which can improve the spatial resolution. The results of numerical simulations and experiments show that the proposed method can remarkably suppress beamforming sidelobes, improve dynamic range of the system, and obtain a higher array gain, which leads to the accurate identification the location of sound sources.

TUESDAY AFTERNOON, 5 DECEMBER 2017 SALON F/G/H, 1:00 P.M. TO 4:45 P.M.

Session 2pUW

Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques II

David P. Knobles, Cochair
KSA LLC, PO Box 27200, Austin, TX 78755

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

Invited Papers

1:00

2pUW1. In situ measurements of sediment sound speed and attenuation at the Seabed Characterization Experiment site in the frequency band of 0.5–10 kHz. Jie Yang (Acoust. Dept., APL-UW, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu)

The Seabed Characterization Experiment, sponsored by the Office of Naval Research, was carried out 5 March–10 April, 2017 (SCE17) on the New England Mud Patch, approximately 100 km south of Martha's Vineyard. The main SCE17 experimental site covers an area of 15 km x 30 km with water depths of 75–80 m. The Sediment Acoustic-speed Measurement System (SAMS) is designed to measure sediment sound speed and attenuation simultaneously over the surficial 3 m of sediments. SAMS consists of ten fixed sources and one receiver which is driven into the seabed. The ten sources combined covers a frequency band of 0.5-10 kHz. During SCE17, SAMS was successfully deployed at 18 sites, which were chosen to measure sediment sound speed and attenuation in both the surficial mud layer and the sandy basement to support modeling effort. In addition, all 18 sites have co-located gravity and/or piston cores that were collected during the two survey cruises in 2015 and 2016. In this talk, a summary of SAMS operation during SCE17 is presented, as well as the preliminary results of sediment sound speed and attenuation and their spatial variation in the frequency band of 0.5–10 kHz. [Work supported by ONR.]

1:20

2pUW2. Geophone and tetrahedral array measurements in the New England Mud Patch and preliminary results. Gopu R. Potty, James H Miller, Poonam Aggarwal, Nipun Aggarwal (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Ying-Tsong Lin, and Arthur Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA)

Measurements of acoustic pressure and particle velocity were made during the Seabed Characterization Experiment (SCEx) in the New England Mud Patch south of Cape Cod in about 70 meters of water. The University of Rhode Island and Wood Hole Oceanographic Institution deployed the "geosled" with a four-element geophone array, a tetrahedral array of four hydrophones and several hydrophone receive units (SHRUs). In addition, a new low frequency source, interface Wave Sediment Profiler (iWaSP) was deployed to excite interface waves. The iWaSP system consists of a source to generate the interface wave and a four-element accelerometer receive array. The range between the iWASP and geosled was about 100 meters. Examples of data collected during the experiments will be presented. Preliminary results of the acoustic and particle velocity data will be discussed. Preliminary analysis of the data and approximate initial estimates of the seabed properties will be presented. [Work supported by Office of Naval Research.]

1:40

2pUW3. Preliminary analysis of geophone and tetrahedral array data from the Seabed Characterization Experiment. Poonam Aggarwal, Nipun Aggarwal, Gopu R. Potty, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882, poonam.aggarwal@gmail.com), Ying-Tsong Lin, and Arthur Newhall (WHOI, Woods Hole, MA)

As part of the Seabed Characterization Experiment (SCEx), the University of Rhode Island and Wood Hole Oceanographic Institution deployed the low frequency shear measurement system and the interface Wave Sediment Profiler (iWaSP) system in the New England Mud Patch south of Cape Cod in about 70 meters of water. Multiple sensors were utilized to collect data which include a geosled with a geophone array of four vertically gimbaled geophones and a tetrahedral array of four hydrophones. The iWasp was used for exciting the interface waves. The iWaSP system was deployed from the R/V Sharp and it transmitted 10 seconds chirp signals every one minute between 33 and 200 Hz. The geophone array was approximately 100 m from the iWaSP source. Detection of the chirps from geophone and tetrahedral array signals was challenging due to the presence of noise sources such as ship traffic and environmental noise. Data were explored for the presence of iWaSP chirp signals in the audio and further analyzed using techniques such as matched filtering. The preliminary results and initial hypothesis regarding the wave types corresponding to the multiple arrivals will be presented. [Work supported by Office of Naval Research.]

2:00

2pUW4. Geoacoustic inferences from seabed reflection measurements on the New England mud patch. Charles W. Holland, Chad M. Smith (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Josée Belcourt, and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of VIctoria, Victoria, BC, Canada)

The vast majority of sediment acoustics research has been aimed at understanding propagation in granular (sandy) sediments. The focus of the ONR Seabed Characterization Experiment was to improve understanding of fine-grained/cohesive sediments. A variety of measurement techniques by various researchers were conducted to that end. Here, we report on initial results from broadband wide-angle reflection measurements at two sites, one with a $\sim$2 m "mud" layer thickness and the other $\sim$11 m thick. The measured reflection coefficient exhibits features that permit estimation of geoacoustic properties including the critical angle (with a rather weak frequency dependence, 200-2500 Hz) and interference patterns in frequency-angle space (which provide information on properties in individual layers). Modeling permits insight into the information content of the data and some initial estimates of the geoacoustic properties. [This research was funded by the Office of Naval Research, Ocean Acoustics Program.]

2:20

2pUW5. Bayesian geoacoustic inversion of seabed reflection data at the New England mud patch. Josée Belcourt, Stan E. Dosso (Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Bob Wright Ctr. A405, Victoria, BC V8P 5C2, Canada, joseebelcourt@uvic.ca), Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA), and Jan Dettmer (GeoSci. Dept., Univ. of Calgary, Calgary, AB, Canada)

This paper presents nonlinear Bayesian inversion of wide-angle seabed reflection-coefficient data for fine-grained/cohesive sediments recorded in the ONR Seabed Characterization Experiment at the New England mud patch. In particular, the inversion is applied to high-resolution broadband reflectivity data from a site with a $\sim$11-m thick mud layer. Trans-dimensional inversion, sampling over an unknown number of seabed layers, and spherical-wave reflection modeling are employed. The inversion provides maximum a posteriori parameter estimates with uncertainties quantified in terms of marginal posterior probability profiles for sound speed, density, and attenuation as a function of depth in the sediment. [The research was funded by the Office of Naval Research, Ocean Acoustics Program, and the Canadian Department of National Defence.]

2:40–3:00 Break

3:00

2pUW6. Surface wave effects on bottom geoacoustic inversions. Ying-Tsong Lin, Timothy F. Duda, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), and Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA)

Underwater sound propagation in the ocean can suffer from the scattering, focusing and defocusing effects caused by surface gravity waves. These effects can in fact also influence bottom geoacoustic inversions, which use sound pressure field measurements in the water column to estimate/infer acoustic properties of the seafloor and/or seabed. In our study presented in the paper, analytical, numerical and experimental approaches are taken to investigate how surface gravity waves can affect bottom geoacoustic inversions (the dependencies on acoustic frequency, surface wave spectrum and directivity, and acoustic waveguide parameters.) The analytical approach is based on acoustic mode theory, and the numerical approach is utilizing both the parabolic-equation (PE) model and the ray model. Experimental data were collected from the Seabed Characterization 2017 experiment conducted on the New England shelf in March and April, 2017. During the experiment, several storms passed through the experimental area, and broadband sound transmissions (500 to 1000 Hz) from fixed sources to fixed receivers have shown significant acoustic effects of surface wind waves. These sound transmissions were at 160 dB source level and had two scheduling window types, one with rapidly repeated sampling and one with less-frequent sampling for a long duration. [Work supported by the Office of Naval Research.]

3:20

2pUW7. Measuring broadband wide angle bottom loss using a hydrophone equipped underwater glider. Yong-Min Jiang (Res. Dept., NATO-STO-Ctr. for Maritime Res. & Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, yong-min.jiang@cmre.nato.int)

Seabed bottom loss is an important factor for predicting sound transmission loss in water. The NATO–STO–CMRE has been investigating novel remote sensing solutions for transforming well established bottom characterization methodologies developed for scientific studies to autonomous platforms. During ONR sponsored Seabed Characterization Experiment 2017, an omni-directional hydrophone equipped Slocum glider was adopted to measure the wide angle bottom loss of a mud patch in New England, 60 miles south of Martha's Vineyard, Massachusetts, USA. An omni-directional acoustic source was deployed over the side of R/V Endeavor to a depth of 30 meters under the sea surface. Chirp pulses ranging from 2 to 20 kHz in frequency domain were transmitted by the acoustic source. The glider was programmed to glider from and then back to the source. Broadband, wide angle bottom loss as a function of grazing angle and frequency was obtained by registering the ratios of bottom reflected and direct arrivals measured at the hydrophone on the glider over an angular range and at different 1/3 octave bands. The measurement technique, signal processing procedure and preliminary results are presented in this paper. [*Work funded by NATO-Allied Command Transformation and Office of Naval Research—Global*.]

3:40

2pUW8. Characterization of mud sediments using the frequency dependence of phase velocity and attenuation of compressional waves. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

The present paper's theory predicts phase velocity and attenuation for mud sediments that contain silt particles, and is based on the model of a suspension consisting of solid particles dispersed in a viscous liquid; the attenuation expression dates back to Lamb's Hydrodynamics and to Urick (JASA, 1948), clarified and rederived by Pierce, Siegmann, and Brown (POMA, 2017). The application to mud is based on the premise that silt particles are held in suspension by the loosely connected matrix of clay particles and that their natural oscillation frequencies are significantly less than the frequencies used in underwater acoustics. The present paper extends that theory with a fresh derivation based on concepts of matched asymptotic expansions and results in an expression for the complex wave number k as a function of the angular frequency. The results for phase velocity disagree with results published in the past by Ahuja (JASA, 1972) and Temkin (JASA, 2000). The assertion is made that the use of the theoretical predictions will enable one to deduce mud sediment properties such as porosity, fraction of volume that is suspended particles, and representative grain sizes from measurements of frequency dependence of sound speed and attenuation. [Work supported by ONR.]

Contributed Papers

4:00

2pUW9. Bayesian inference and model selection to investigate seabed shear-wave velocity profiles via interface-wave dispersion inversion. Stan E. Dosso (School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca) and Hefeng Dong (Dept. of Electron. Systems, Norwegian Univ. of Sci. & Techonolgy, Trondheim, Norway)

This paper applies a general geoacoustic profile parameterization based on Bernstein polynomial basis functions to consider the form of seabed shear-wave velocity profiles via Bayesian inversion of interface (Scholte) wave dispersion. Theory and observations indicate that the shear-wave velocity profile of seabed sediments of uniform composition and density often corresponds (approximately) to a power law function of depth. Hence, past inversions have often been based on a power-law parameterization without considering independently if this form is actually required by the data. The Bernstein parameterization, based on a linear combination of Bernstein-polynomial basis functions with the polynomial order determined by the Bayesian information criterion, is general and allows the form of the profile to be determined by the data, rather than by a subjective prior choice. In this paper, measured-data inversions are compared for power-law, Bernstein-polynomial, and layered trans-dimensional parameterizations to investigate the shear-wave velocity profile form.

4:15

2pUW10. Applications of a suspension theory of silt particles in mud to frequency and other parameter dependencies of geoacoustic quantities. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Retired, East Sandwich, MA), Jason D. Chaytor (U.S. Geological Survey, Woods Hole, MA), Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

A recent suspension theory of marine mud [Pierce, *et al.*, POMA 29, accepted] hypothesizes that embedded silt particles are the dominant contributors to compressional wave attenuation. The approach predicts frequency intervals within which attenuation increases roughly linearly with frequency, as often assumed. These intervals depend on the measured (or assumed) mean silt particle size. This presentation investigates the influence of distributions of silt particle sizes on attenuation, including the distribution shapes obtained from data, and the intervals of linear frequency with multiple particle sizes. In addition to attenuation, the theory also provides compressional sound speed predictions. Their sensitivity to changes in measured physical parameters and frequency will be determined, and the results compared with archival data and recently analyzed SBCEXP core data along a range-dependent experimental track. Another consequence of the theory is how porosity is affected by the minimum separation distance between silt particles and including viscous boundary layers. An environmental-acoustic model of the experimental track will be constructed, based on ocean sound-speed profiles, bathymetry, analyzed core data, and attenuation from theory. This model will be used to calculate transmission loss for propagation along the track to compare with acoustic measurements from the experiment. [Work supported by ONR.]

4:30

2pUW11. Physical and acoustical properties of marine sediments with a wide particle size distribution. Justin T. Dubin, Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, justin.dubin@utexas.edu), Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Naturally occurring marine sediments can possess poorly sorted or multimodal particle size distributions. Furthermore, while well-sorted coarse grained sediments are typically characterized according to their grain size, for mixed or fine grained sediments the shape of the particles can have significant influence on the acoustical properties. In this work, particle size and morphology are investigated using scanning electron microscopy (SEM). Images of dry particles are analyzed to determine the sphericity and angularity of sand and silt grains and to estimate the shape (e.g. platelets or needles) and aspect ratio of clay particles. SEM images of wet samples are analyzed to understand the sediment microstructure, including the interaction between sand, silt, and clay particles. Additionally, the effect of the proportionally varying composition of sand, silt, and clay particles on the bulk properties of the sediment, including the porosity and compressional wave speed, will be examined using a multiphase effective medium model. These techniques are applied to sediments collected in the New England Mud Patch as part of the Seabed Characterization Experiment. [Work supported by ARL:UT IR&D and ONR.]

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday, 5 December

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

Committees meeting on Wednesday, 6 December

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

Committees meeting on Thursday, 7 December

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